AASTRA

Ascotel IntelliGate 2025 / 2045 / 2065
System Manual

Ascotel IntelliGate Telecommunication Systems

This document describes the Ascotel IntelliGate telecommunication system as of SW version 6.1:

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1 What's New

Overview of the Innovations in Ascotel IntelliGate with I6.1

New terminals

- Office 1600IP:
  The PC-based system terminal Office 1600IP is an OIP client application (see "Open Interfaces Platform", page 136). With its user-friendly interface it extends the boundaries of the Office system terminals, provides powerful group functions and integrates outstandingly well into standard PC programs.
- Office 135/135pro:
  The new DECT handset is the Office 130 / Office 130pro’s successor. It stands out by virtue of a new colour concept, an improved battery pack and a number of new functions.
- SB-8 radio unit:
  The new SB-8 radio unit allows the user to make up to eight calls simultaneously, and is connected to the PBX using two AD2 interfaces. It can also be used on I6.1 < systems, in which case it responds in the same way as an SB-4. The SB-8 supports all Office DECT handsets.
- SB-8ANT radio unit:
  This model differs from the SB-8 radio unit through two external antenna connections. Radio coverage can be adapted more effectively to local conditions by using external antennas.

New Features

- New terminal features:
  – Switching on / off with the C-key (Office 135/135pro only)
  – Redkey function, e.g. for triggering an alarm
  – New Hotkey mode on the DECT system terminals
- New system features:
  – New UG type "Large User Group" with up to 400 subscribers
  – Possibility of changing your personal password using a * procedure (throughout the PISN)
  – New configuration options for exchange-to-exchange connections
  – Possibility of uploading Courtesy texts in the form of wave files
  – Possibility of upgrading a 2025/2045 system from I5.x to I6.1 remotely
  – Possibility of making private calls throughout the PISN using a password

Higher System Limits

- The maximum number of corded terminals, abbreviated dialling numbers, PISN subscribers and CTI clients has been increased.

New Licences

- Office 1600IP licence:
  A licence is required to operate the IP-based PC phone Office 1600IP. One Office 1600IP client is enabled with each licence.
- Trial licence (Office 1600IP and CTI):
  This licence temporarily enables CTI clients and Office 1600IP clients. The number of clients is determined solely by the system limits. The licence is valid for 30 days once it has been generated on the licence server.

Documentation

- New Documents
  - User's guide Office 135/135pro
  - Online User's guide Office 1600IP
- New versions:
  - User's guides Office 10 / 25 / 35 / 35pro / 45 / 45pro / 155pro, Function Overview, System Assistant Office 45
  - Ascotel IntelliGate 2025 / 2045 / 2065 System Manual
  - Ascotel IntelliGate 2025 / 2045 / 2065 System Description
  - System Manual application interfaces
- Each system supplied comes complete with the CD containing the User’s Guide (excl. System Manuals) and the AIMS Client Management Set.

Overview of the Innovations in Ascotel IntelliGate with I6.0

Hardware

- The hardware of the PBX platforms remains essentially the same for I6. The only exception is the memory expansion requirement on the basic system 2065. The memory expansion consist of a Flash card and a RAM card and are included in the corresponding "I6 Update-Set". All you need to do to upgrade a 2025 / 2045 basic system to I6 is upload a new system software. Existing configuration data can be ported onto the new systems.

- New expansion cards are available for expanding the system with a larger number of corded subscribers and DECT subscribers:
  – Subscriber card SC-32AD2 with 32 AD2 interfaces
  – Subscriber card SC-16AB with 16 a/b interfaces
  – Special card DSP-04 with 36 DECT channels and 9 full-duplex hands-free channels

- The new hardware launched with I5.5 is still available and can be used with I6:
  – Expansion card AIP-6350
  – Office 35IP system terminal
  – Rack version Ascotel IntelliGate 2025R / 2045R and 2065R
  – ISDN-08ST Expansion Card

- All I5 expansion cards not specifically mentioned here can also be used on an I6 system.

New Features

- New terminal features:
  – Suppression of the call number display (CLIR) for each call
  – Forwarding or rejecting a call during the ringing phase
  – Improved menu prompting for function key configuration
  – Choice of more functions for function key configuration
  – Redirecting information during the ringing phase for outgoing and incoming calls
  – Distinction between unknown and suppressed CLIP in the case of incoming calls
  – Protect against call waiting/announcement configurable by menu
  – Simplified navigation in lists of callers and answered calls
  – In addition to standard texts possibility of storing 5 personal texts (Office 45)
  – Possibility of selecting favourite display in idle state (Office 45 VA)
  – 3 lines available for displaying messages, alarms, call numbers and names (Office 130, Office 130pro and Office 155pro only)
  – 4 lines available for navigating through lists Office 130, Office 130pro and Office 155pro)
  – Possibility of configuring new "LED only" ringing setting (Office 130, Office 130pro and Office 155pro)
  – New GAP features for convenient handset operation on third-party systems (Office 130 und Office 130pro)
  – Possibility of unlocking the keypad for 20 seconds (Office 130, Office 130pro und Office 155pro)

- New system features:
  - New "virtual subscriber" type available for implementing various applications
  - Function/team keys on system terminals individual lockable using AIMS
  - Possibility of configuring 50 time-controlled functions (*/# procedures)
  - Default forwarding of unanswered calls configurable for each subscriber
  - Voice Mail destination configurable for CFU chains
  - One analogue user-network interface per system configurable for connecting a general bell
  - Supports public ISDN network notifications
  - Operation of features on third-party PINXs in private QSIG network with visual confirmation
  - Recall time configurable for each subscriber
  - Private calls with password for automatic billing to the private charge counter

Higher System Limits

Now that the system limits on the 2065 basic system have been massively increased, it is possible to configure and operate considerably larger systems:

- The maximum number of network and user-network interfaces has been doubled
- The maximum number of user groups has been more than tripled

New Licences

- Advanced Messaging licence:
  Enables the SMPP protocol to be used for integrating an SMS server and 9d handsets to be logged on as system terminals (Ascom Wireless Solution products). User-friendly messaging systems can then be implemented with Ascotel IntelliGate.
- AMI licence:
  For the implementation of very large DECT systems Ascotel IntelliGate 2065 can be connected to the DECT radio system DCT 1800 (Ascom Wireless Solutions product). The Ascotel Mobility Interface (AMI) is available for this purpose. Activation of the AMI functionality is subject to a licence.

Information and Management System AIMS

- Backwards compatible with I5 systems
- One version for all channels, regardless of country
- New version of the Project Manager
- Hardware generation and version displayed as well as the card type
- Consistency adaptations for the different Managers
- Right mouse button displays a context-sensitive menu
- Redesigned Information Manager with expanded documentation

Documentation

- New version of the Office User’s Guide and Quick User's Guide, the "Function Overview" and the "System Assistant" for Office 45
- New version of the System Manual Ascotel IntelliGate 2025 / 2045 / 2065
- New version of the System Description Ascotel IntelliGate 2025 / 2045 / 2065
- The User’s Guide (excl. System Manuals) can be found on a CD together with the AIMS Client Management Set, and is available to order.
- The System Documentation (incl. System Manuals) can be found on a CD together with the AIMS Configuration Set, and is available to order.
- The entire documentation is available on the Internet/extranet:
  – User-related documents on the Internet:
    http://www1.aastra.com/docfinder

Overview of the Innovations in Ascotel IntelliGate with I5.5

New Hardware

- The new AIP-6350 expansion card and the corresponding Office 35IP terminal enable the use of the IP infrastructure for telephony and expand the Ascotel IntelliGate platform to the IP data network. The system terminal Office 35IP is a fully fledged IP terminal with the complete range of features of an Office 35.

- Rack version Ascotel IntelliGate 2025R / 2045R and 2065R:
  With the greater IP integration of Ascotel IntelliGate systems the existing PBX range is being expanded with a rack version. The scope of performance and functionality do not differ from those of systems with conventional design. This

means that all the information contained in the Ascotel IntelliGate documentation is also valid for the rack versions.

- The range of expansion cards is complemented by the new ISDN-08ST expansion card.
- Following the redesign of the 2025 / 2045 mainboard (hardware version with the designation MBS-2), an external Auxiliary Terminal Power Supply (ATPS) can now be used, as on the 2065 mainboard (MBL-2). The auxiliary power supply is required for system expansions involving a large number of terminals with high power requirements.

New Features

- "DECT Follow Me" improves the reachability of DECT subscribers in a private network of up to 4 systems.
- More configuration possibilities on the ETSI S bus (format and exchange access prefix)
- More configuration possibilities for Courtesy Service (pause duration)
- The software of the Office 155pro handset can be updated via the DECT interface (AIR upload).

Greater Expandability

- The restriction to 2 of the 5 expansion slots on the Ascotel IntelliGate 2025 has been lifted. This means the 2025 and 2045 systems differ only in their system limits.

New Licences

- The new licence "CTI Basic" is available to make use of the CTI basic functions on the third-party CTI interface of Ascotel IntelliGate.
- ATAS licence, for connecting external alarm and messaging sources to the PBX via Pocket Adapter, V.24 or Ethernet. The licence is granted with certified special applications.

New Documents

- AIP 6350 / Office 35IP System Manual
- Office 35IP User’s guide and Quick User's Guide
- Application Interfaces System Manual (replaces CTI Interface System Manual)
- Addendum to System Manual Ascotel 2025 / 2045 / 2065 Version 5.2

Overview of the Innovations in Ascotel with I5

PBX Hardware Platforms

- The PBX hardware has been completely revised. The three modular systems are based on two basic systems. One shared system for Ascotel 2025 and Ascotel 2045, and a separate system for Ascotel 2065.
  Older systems cannot be upgraded. Existing configuration data can be ported onto the new systems.
- Compared with I4Net systems the new hardware platforms process up to three times more traffic capacity (load).
- 4x, 8x, 16x and 24x cards are available for expanding the system with AD2 user-network interfaces.
- The systems' basic equipment includes two serial interfaces and an Ethernet interface.
- The IPI-100BT card replaces the MIPR module; it is used for the application IP Gateway AIP 6400.
- The USP-12V module ensures that the system is able to operate without interruption in the event of a 230 V mains failure. This means that emergency circuits are now redundant.
- A system can be operated with 48 VDC using the DC-48V module.
- All systems can either be wall-mounted or rack-mounted inside a 19" cabinet.
- An external Auxiliary Terminal Power Supply (ATPS) is available for system expansions with large numbers of terminals with high power requirements.

System Terminals

- The new family of system terminals with 5 corded terminals Office 10, Office 25, Office 35, Office 45 and Office 45pro stands out by virtue of its design and proved Foxkey operation. Office 45/45pro can be used as an ordinary phone, a key telephone or as an Operator Console, and replaces the Crystal. The system terminals Office 20, Office 30 and Office 40 are supported as before.
- An expansion keypad and an alpha keyboard are available as options.
- For Office 45/45pro the system offers centralised full-duplex handsfree operation.
- The CTO family of system terminals (Crystal, Topaz, Opal) is no longer supported.

- The new release of the PC Operator Office 1550 features a number of new functions. It is backwards compatible to I3 and is operated on the S bus, as before. The Terminal Adapters (ABSC-TA) have to be upgraded with new firmware.

Cordless System

- The new Office 130 / Office 130pro is a light, practical DECT terminal ideally suited for an office environment, hospitals and retirement homes.
- The new splashwater proof and shock-proof DECT terminal Office 155pro is particularly well suited for the workshop and heavy industry sector.
- The system still supports the DECT terminals Office 100 and Office 150. The bcs cordless terminal is no longer supported.
- There has been a significant increase in the maximum number of radio units and handsets that can be operated.

Voice Mail System

- The fully integrated Voice Mail System AVS replaces ACCS and offers a number of new functions for up to 128 mailboxes. There are 2 AVS cards that can be used in all three PBX systems.
- With the Voice Mail Manager integrated in AIMS the application can be configured locally or from a distance. A complete backup and restore of the configuration data, welcome texts and messages is also possible.

New Features

- Terminal features:
  – Unlocking a phone for each call
  – Making calls on a third-party terminal using one's own authorisations
  – Logging into and out of specific individual user groups (instead of the previous collective logon/logoff)
  – Enhanced Voice Mail support on system terminals
  – Different ringing on two-company systems
- System features:
  – External remote control of subscriber and system features
  – Alternative routing of incoming calls if busy or if no answer, for example for creatign a UG overflow
  – Additional ISDN supplementary services on the S bus
  – Supports the ISDN service MCID

- Supports ACD applications with an internal ACD queue, and emergency routing in the event of a failure of the ACD server
- Possibility with P-MP connections of automatically call forwarding unconditional to the ISDN exchange in the case of exchange-to-exchange connections (viz. PARE with P-P connections)
- LCR fallback to an alternative network provider (alternative routing), automatic or manual
- Exchange-to-exchange connections also permitted via analogue network interfaces
- Enhanced functionality of the Courtesy service (announcement prior to answering)
- Internal music for callers on hold available on all systems (Ascotel 2025 / 2045 and 2065)

Information and Management System AIMS

- The Ascotel Information and Management System AIMS is the Installer's tool for PBX configuration; Crystal is no longer available. As an innovation AIMS can now be connected to the central, high-speed Ethernet interface. The Find and Sort functions have been expanded.
- Data from older AIMS versions can now be imported. The application managers for Voice Mail and IP Gateway can be started directly from AIMS.
- With the upgraded Upload Manager it is now possible to load not only the PBX system software but also the software for Office 45/45pro , the radio units and Office 130 / Office 130pro either locally or remotely. It is no longer necessary to replace memory modules on site.
- Frequently occurring configuration changes can now be carried out by the customer himself on the Office 45 using the System Assistant application.

Licensing

- In addition to the licensing of the QSIG B channels, licences can now be obtained for the following products and functions:
  - System upgrade Ascotel 2025 to 2045
  - Activation of the PBX-internal ACD features
  - Number of simultaneous third-party CTI interface users

Documentation

- Manual:
  - One manual for all systems
  - Clearly structured
  - Prefaced with a complete table of contents
  - Separate list of abbreviations and glossary
- Operating Instructions for Office x5 terminals:
  - One set of Operating Instructions for each language
  - New "Function Overview" document
  - New "System Assistant" Operating Instructions for Office 45
- Most of the documents are available on the Internet:
  - The address and scope of the documentation provided depends on the distribution channel

2 Safety and Commissioning

Please observe the following instructions on proper use and procedure:

- The Ascotel IntelliGate 2025, Ascotel IntelliGate 2045 , and Ascotel IntelliGate 2065 systems are designed exclusively to be used as telecommunication systems for voice and data transmission.
- The Ascotel IntelliGate 2025, Ascotel IntelliGate 2045, and Ascotel IntelliGate 2065 systems can be operated in conjunction with Office family terminals, standard ISDN terminals, analogue terminals, and with supplementary equipment and applications distributed or certified by Aastra.
- Observe the provisions of the System Manual when installing, commissioning, and operating the system. These provisions of the System Manual are essential for safety-relevant operations, such as earthing, installation, connection to the 230 V mains, etc.

Safety Considerations

Special hazard alert messages with pictograms are used to signal areas of particular risk to people or equipment.

Hazard:
Failure to observe information identified in this way can put people and hardware at risk through electrical shock or short-circuits respectively.

Warning:
Failure to observe information identified in this way can cause a defect to a module.

Warning:
Failure to observe information identified in this way can lead to damage caused by electrostatic discharge.

Note:
Failure to observe information identified in this way can lead to equipment faults or malfunctions or affect the performance of the system.

Before Commissioning

Check for completeness when unpacking the delivered system. Complaints are to be settled within an appropriate period.

Damaged Components

Check the components for damage. Damaged systems or components must not be put into operation.

Foreign Matter in the Hardware

If objects or liquids have penetrated the equipment and/or modules, said equipment and/or modules must not be put into operation.

Hazard:
There is a risk of electrical shock or short-circuits if the hardware is damaged in any way or if liquids or objects get inside the equipment. Immediately disconnect the system from the power supply (230 V mains as well as UPS and DC power supply if present).

Electrostatic Discharge

Warning:
The system's reliability can be adversely affected by electrostatic discharges caused by touching electronic components and elements, and subsequent damage can result. Always observe the ESD guidelines.

Mounting Instructions

Observe the following stipulations when installing the system:

- Do not expose the system to direct sunlight or other sources of heat (radiators). Observe the guaranteed temperature range.
- Observe the minimum distances specified in the assembly guidelines.
- To ensure the necessary circulation of air, do not obstruct the system's cooling ducts.
- The premises in which the system is installed must meet the following requirements:
  - Protected against water, condensation and humidity
  - Well ventilated
  - Observance of the permitted humidity range

Hazard:
Explosion hazard through operation in areas subject to explosion hazards.
Do not install the systems in areas subject to explosion hazards.

Installation Instructions

Observe the installation instructions precisely with regard to the components installed, the earthing concept and the connection of the system.

Observe the electrical limiting values specified in the System Manual when connecting third-party equipment such as switchgears, music sources, etc., to the system.

During Operation

During operation, the system is under 230 V mains voltage. Disconnect the system from the power supply when work requires opening the inner housing cover.

Maintenance Work

Any servicing, expansion or repair work is to be carried out only by technical personnel with the appropriate qualifications.

Replacing Components

Cards should be fitted or removed only once they have been disconnected from the power supply. Always use original parts and components exclusively from the Ascotel range.

PCBs are to be stored and shipped only in the specially designed antistatic packaging.

Cleaning

Clean the terminals with a damp cloth. Never use aggressive detergents.

Data Protection

During operation the system records and stores personal customer data (call logging). Take the following precautionary measures:

- During configuration, always keep the configuration / planning data contained on the relevant data carriers under supervision.
- Ensure that only authorized persons have access to the data.

Regulating Access to the System Configuration

To ensure that only authorized persons have access to the system data, consistently implement the following protective measures:

- Change the initialization passwords.
- Change the passwords at regular intervals and keep them under lock and key.
- Regulate remote maintenance access.

3 Target Audience and Structure of the System Manual

The System Manual is intended for planners, installers and system managers of telephone installations. A basic knowledge of telephony, in particular of ISDN technology, is required to understand the content of the System Manual.

The System Manual is available only in electronic form as a document in Acrobat Reader format, and can be printed out. Navigation in PDF format is based on the bookmarks, table of contents, cross references and index. All these navigation aids are linked, i.e. a mouse click takes you directly to the corresponding places in the Manual. We have also ensured that the page numbering in the PDF navigation corresponds to the page numbering of the Manual, making it much easier to jump to a particular page.

Parts of the System Manual

The System Manual is divided into a preface and 9 parts. The following chapters provide an overview of the contents of each part and indicate which parts are to be used for specific tasks. Chapter 1 of each part of the manual describes the contents of that part in detail.

"Part 1 System Overview"

Part 1 contains information on the expansion of the systems, interfaces, networking types, terminals, supplementary equipment and AIMS (Ascotel Information and Management System).

Always refer to Part 1 whenever you need basic information, for instance the table of interface types or the definition of a networking technology term.

"Part 2 System Functions and Features"

Part 2 provides an overview of the system's functions and features. Knowing all about these functions and features is an essential requirement for providing customer-oriented planning and implementation of systems.

System Functions

This chapter is based on the principles of the networking philosophy discussed in Part 1 and features the different types of internal and external numbering plans for private and public networks.

Other system functions:

- Subscriber identification and display using CLIP (Calling Line Identification Presentation) and CNIP (Calling Name Identification Presentation)
- Routing elements and the way they interact in distributing incoming and outgoing calls to network and user-network interfaces
- Routing data services to destination tables or individual destinations
- Logging and output of call data and call charge data

Numerous examples are used to illustrate this complex subject and to make it easier to understand.

Features

Each subscriber has the possibility of activating a multitude of features in day to day telephony operations. Depending on the terminal type connected, this is done using function keys and menu-driven key sequences. Most of these features can be activated using */# procedures even on terminals with the lowest level of features. The table in "Features Overview", page 563 provides an overview of the features available with the different terminal types.

"Part 3 Planning"

Part 3 begins with a system overview of the basic systems and their expansion stages using expansion cards and modules. It also looks at technical and licence-related system and expansion limits. The customer's requirements are the prime concern when expanding a system. The objectives, demands and requirements of the customer must be reconciled with the technical circumstances and possibilities. Part 3 supports you in accomplishing the following tasks:

- Determining customer needs
- Coordinating objectives, requirements, wishes and possibilities
- Drawing up a rough concept
- Planning the system

AIMS Managers provide assistance when planning the system

The planning of DECT systems is examined in particular detail. Cordless systems require not only that you determine the customer's requirements, wishes and current communication situation, but also that you carry out meticulous measurements to ascertain the site conditions. The last chapter looks at the different aspects involved in planning a private network. An example network is used to explain the planning procedure step by step.

"Part 4 Installation"

In Part 4 you will find information on the installation and wiring of the system and on the choice of the right installation material. This is followed by a description of how to wire up the different types of network interfaces, user-network interfaces and special interfaces, and how to connect the terminals.

The sequence of chapters in Part 4 corresponds to the procedure in installation practice:

- Mounting, preparation and connection of a system
- Installation of the system and expansion cards
- Connection of the terminals

At the end of Part 4 you will find a checklist for verifying the installation.

"Part 5 Configuration"

Part 5 explains how to configure a system efficiently and implement the characteristics specific to the customer. The practical description of the configuration processes takes account of the different requirements of the system variants.

The AIMS Managers provide support with the configuration. With their help you can configure the system and customer data, and specify the access concept for authorization. At the end you will find instructions for selecting the initialization values.

"Part 6 Commissioning"

Before commissioning and handing over the system to the customer, a check is carried out to see whether the system complies with the project specifications, and whether it is operating correctly. Part 6 lists the general tests and function tests for the technical commissioning.

The check-list at the end provides valuable assistance when handing over the system to the customer.

"Part 7 Operation and Maintenance"

Part 7 deals with the maintenance, expansion and smooth operation of the system. It discusses the following topics:

- The updating and servicing of configuration data and system software
- Hardware updates and system expansion
  - Licence types and how to adapt them
  - Replacing, dismantling and reducing hardware and terminals
- Monitoring operations and troubleshooting

This part also discusses how the System Event Manager (SEM) has been integrated to log event messages issued by the system and the Fault and Maintenance Manager for remote maintenance. Numerous error tables and troubleshooting measures are provided to help with fault diagnosis and recovery.

"Part 8 Annex"

The Annex lists the generation change, the functions and products no longer supported, the systematic PCB designation model, compatibility information, technical data, and the PC Dial commands. You'll also find an overview of the control elements and digit key assignment for the system terminals, and a summary of additional documents that can be accessed via the Internet.

"Part 9 Abbreviations, Glossary and Index"

The list of abbreviations in Part 9 provides an explanation of all the abbreviations used. The Glossary explains the main terms used in the Manual. Abbreviations are listed there in their full form. The index can be used to find a particular topic or subject quickly.

4 Documents and Help Systems with Further Information

Product	Document
Ascotel IntelliGate	System Description Application Notes
AIMS	Installation instructions (readme file) Information Manager Operating Instructions for setting up a PC-PBX connection for Windows NT 4.0 Operating Instructions for setting up a PC-PBX connection for Windows 95 / 98 / ME Operating Instructions for setting up a PC-PBX connection for Windows 2000 / XP Help system Application Notes
Upload Manager	Help system
System Event Manager	Help system
Project Manager	Operating Instructions
AVS Voice Mail System	System Manual Operating Instructions Quick User's Guide Help system
Application interfaces	System Manual Help system
Networking	System Manual AIP 6400
IP network	System Manual AIP 6350
IP system handsets	User's guide Office 35IP / Office 1600IP Quick User's Guide Office 35IP
Office System terminals	Quick User's Guides for Office 10 / Office 25 / Office 35 / Office 45/45pro / Office 135/135pro / Office 155pro User's guides for Office 10 / Office 25 / Office 35 / Office 45/45pro / Office 135/ 135pro / Office 155pro Operating Instructions for Office 20 / Office 30 / Office 40 Function overview System Assistant Operating Instructions Office 45
PC Operator Console	Help system Installation and configuration instructions Quick User's Guide Office 1550 a/b Adapter Operating and Installation Instructions

User-related documents on the Internet: http://www1.aastra.com/docfinder

Part 1 System Overview

1 Overview of Chapters

Telecommunication system Ascotel IntelliGate

The overview of basic systems, supplementary equipment and applications in Chapter 2 illustrates the wide array of possibilities available with the Ascotel IntelliGate system to businesses in all industries. It provides basic information on the PBX, interfaces, expansion stages and system limits.

System Interfaces

Chapter 3 features the system's interfaces and describes the different connection concepts for digital and analogue network and user-network interfaces. Special interfaces include the Ethernet interface, V.24 interface, door intercom systems and the general bell.

Networking the Systems

Chapter 4 explains the networking philosophy for private networks based on the ISDN standard (PISN). It deals with such topics as permanent and virtual networking, communication protocols, connecting systems within the network, numbering plans and network services.

Terminals

The system terminals of the Office family featured in Chapter 5 allow full use of the system's entire performance spectrum. The PC-based system terminal Office 1600IP and the PC Operator Office 1550 are two other high-performance terminals. Terminals by other manufacturers can also be connected, albeit with a restricted scope of performance.

AIP 6350 and Office 35IP

The joint use of the two IP components, namely the IP interface card AIP-6350 and the system terminal Office 35IP, expands the Ascotel IntelliGate platform to the IP data network, and makes the IP infrastructure available for telephony, too. Chapter 6 lists the advantages of such IP integration.

Ascotel IP Gateway AIP 6400

Chapter 7 describes the use of LANs for the system via an IPI (IP Interface expansion card), a gateway for voice transmission in data networks, and possible applications.

Ascotel Voice Mail System AVS 5150

Chapter 8 explains how availability in telecommunications can be enhanced using the integrated Voice Mail System

Supplementary Equipment and Additional Applications

Chapter 9 tells you which supplementary equipment can be connected to the system. CTI (Computer Telephony Integration) and TAPI (Telephony Applications Program Interface) provide powerful programs for PC-aided telephony and data services. The OIP middleware provides the application interfaces as an homogeneous interface for the applications. Paging systems, large DECT radio systems, messaging and alarm systems are examples of useful supplementary equipment.

Ascotel Information Management System (AIMS)

The Ascotel Information Management System (AIMS) assists you with the planning, configuration and monitoring of the system. Chapter 10 describes the AIMS Managers used for different tasks, and the data exchange between the PBX and the PC's hard disk.

2 Telecommunication system Ascotel IntelliGate

Ascotel IntelliGate is a comprehensive digital communications system for companies operating in all industries, with subscriber numbers ranging between 7 and 400 subscribers. Thanks to its networking possibilities with systems by other manufacturers Ascotel IntelliGate is also used in branches of large companies. The core of the Ascotel IntelliGate telecommunications system is made up of PBX platforms with different expansion capacities. The system concept is modular, which means it can be expanded step by step to accommodate a greater number of subscribers or additional functions.

The PBX for Digital, Analogue, Public and Private Networks

Ascotel IntelliGate can be operated on digital (ISDN) and analogue public networks. Several PBX nodes can be networked in different locations via fixed connections or, virtually, via the public ISDN network. Ascotel IntelliGate is ideally compatible with systems by other manufacturers as an international language (the QSIG protocol) is used between the network nodes.

High computing and switching capacity

Ascotel IntelliGate systems are equipped with extremely powerful computers. The 2065 system is capable of processing up to 16,000 calls an hour, and up to 120 calls can be made simultaneously.

A Multitude of Interfaces to Terminal Equipment

Besides the array of digital user-network interfaces used for connecting Office terminals, standard interfaces are also used for connecting the wide range of commercially available voice and data terminals. This ensures Ascotel IntelliGate compatibility with digital and analogue telephones, with modems, PCs, fax machines and paging systems. Also door intercom systems and music sources can be hooked up to Ascotel IntelliGate.

Open and Receptive

Ascotel IntelliGate systems are open communication platforms that use and support internationally defined standards. Complying with standards is a requirement ensuring that a switching system is compatible with terminals and equipment from third parties so that it is possible to seamlessly integrate into existing customer communication infrastructure.

Important standards supported by Ascotel IntelliGate:

- DECT (Digital Enhanced Cordless Telephony), the standard for cordless telephony within companies
- GAP (Generic Access Profile), standardized interface for integrating DECT terminals by other manufacturers.
- QSIG, the protocol for networking switching systems of different manufacturers
- Microsoft TAPI (Telephony Application Programming Interface), the interface for computer-supported telephony

Installation

Thanks to its compact assembly Ascotel IntelliGate can be wall mounted or installed in a 19-inch cupboard with other communication and network components. The rack versions of the Ascotel IntelliGate systems are designed for the latter variant. All the interfaces of the basic systems and expansion cards are accessible on the front panels.

Power supply

No business, no organisation can afford to be without a phone system for any length of time. The integrated UPS electronics (UPS: Uninterruptible Power Supply) ensures that in the event of a mains power failure the system goes on operating without interruption using an external battery (12 VDC). Power can also be supplied to the system by a 48 VDC central battery.

Networking Systems on Multiple Locations

There are several differentiated options for networking Ascotel systems among themselves or with systems by other suppliers. Networking can be via ISDN leased lines, with only the fixed rental costs incurred for the leased lines. Alternatively, networking is set up using an existing, voice-compatible IP data network (interconnected Ascotel systems only), in which case there are no call charges due to the network provider or rental costs to pay for leased lines. A third variant available with Ascotel is virtual networking, which runs neither via permanent leased lines nor via the private data network but via the public ISDN dial-up network.

Office Corded System Terminals – the Right Terminal for Everyone

The Office system terminals fully meet all requirements when it comes to functionality and user-friendliness. Four corded terminal types are available, from entry-level to added-feature model. There are two alternatives for the Operator's workstation: Office 45 as Operator Console and the PC Operator Console Office 1550.

The PC as Operator – Transparency for the Attendant

Reception is a company's business card, also telephone reception. This is where the Office 1550 system terminal provides new options for maintaining business relationships. All key information is easily available at a glance on the screen for all external and internal calls. A mouse click or keystroke is all it takes to answer, process and forward calls. The PC Operator Console reduces attendant workload. This frees up the attendant for other tasks and ensures your reception is friendly, relaxed and in control.

Cordless Telephony – When Mobility is a Key Issue

Mobility is the key. Handsets featuring the full range of system functionality and based on the DECT standard are available for all systems. Compact and attractively styled handsets for the business sector and a sturdy handset for the industrial sector are the preferred means of communication for people on the move.

Aastra Paging System – Seek and You Shall Find

Mobility with the teleCOURIER 900 paging system: Seamlessly integrated into the system via the S interface, it is frequently used in hospitals and retirement homes.

Office 35IP: Easy, Convenient Phone Calls via the Data Network

An Ascotel IntelliGate system does not have to be installed at each company location to ensure networking throughout the company. The joint use of the two IP components, namely the IP interface card AIP-6350 and the system terminal Office 35IP, makes the IP infrastructure available for telephony, too. In this way networked, remote workstations can be integrated at low cost into the internal telephone system, without compromising the ease with which phone calls are made. All that is required to make and receive phone calls via the existing data network is an existing voice-compatible IP connection. The Office 35IP from Ascotel IntelliGate has been designed specifically for this purpose and features the same high-performance profile as the classic Office terminals. While neither users nor Administrator notice any differences in quality, the advantages are clear for all to see.

Office 1600IP: The PC-based system terminal

The PC-based system terminal Office 1600IP is an OIP client application. With its user-friendly interface it extends the boundaries of the Office system terminals, provides powerful group functions and integrates outstandingly well into standard PC programs. Although designed primarily with small to medium-sized workgroups in mind, the Office 1600IP is also an invaluable companion for individual users with high mobility requirements.

Intuitive Use – Uniformity across the Range of Terminal Devices

The operator interface of all Office terminals is consistently uniform and designed for intuitive use. All the conventional telephony and switching functions are simple to operate, without the need for operating instructions.

Computer Telephony Integration – The Efficient Way to Telephone

Ascotel IntelliGate offers open computer telephony interfaces based on the Microsoft standard TAPI 2.1. For single-station solutions, first-party CTI interfaces implemented using the Pocket Adapter are available. For multi-station solutions, Ascotel IntelliGate offers third-party CTI interfaces which are implemented via the Ethernet system interface. Based on Microsoft Windows, external applications use the Microsoft standard TAPI and the Ascotel CTI interfaces to control and supervise all Office PBX terminals. In addition to simple applications such as phone diallers and pop-up windows, other more sophisticated applications such as a server-based call center can also be implemented.

ACD – Automatic Call Distribution

ACD (Automatic Call Distribution) is a basic component of a call center system. To this end Ascotel IntelliGate provides an internal ACD queue for storing incoming calls. The distribution of stored calls to agents is performed by an external ACD server connected to the third-party CTI interface.

Ascotel Voice- Mail- System – and No Call is Ever Lost

The Ascotel Voice Mail System AVS 5150 makes sure you are always well informed. You will never miss a call and are always reachable -- without being bound to your desk. Also when you are on the move, you are notified of new messages and can access and listen to them remotely.

Operation with external messaging, monitoring and alarm systems

Alarms, faults and messages from building energy management systems, nurse paging systems, security systems, etc., can be signalled as text messages to Office terminals via the V.24 interface of the Pocket Adapter or via the Ethernet interface. The use of cordless terminals to receive text messages is the best way to achieve maximum user reachability.

Call Redirection, Team Work, Charge Acquisition: The basics of Ascotel IntelliGate

Ascotel IntelliGate supports all the standard telephony and switching functions, with one aim in mind: to make calls with outside call parties and colleagues within the company as efficient and as user friendly as possible. Ascotel IntelliGate takes full account of the fact that people are increasingly working in teams. Team circuits, key telephones, user groups and several variants of call forwarding functions support this modern form of organisation.

Another important group of features are call data functions. Call charges are displayed to the subscriber; they can be logged and assigned on a user-pays basis, and also printed out. Morever, charges can also be reduced by using telephone locking functions, targeted exchange access restrictions and integrated least-cost-routing functions (LCR). These are ways to prevent misuse.

AIMS, the Tool Kit for All Management Tasks

Communication systems need to adapt to the customer's organization and requirements. AIMS, the PC-based Information and Management System, is designed for planning, configuring and supervising the system. Changing configuration data during operation and automatic PC updating, directory management, printing terminal labels – these are just some of the diverse management functions easily performed with AIMS. AIMS is a tool designed specifically for customizing a system.

Frequently occurring settings on the system can be made by the customer himself. For this purpose the Office terminals Office 45 and Office 45pro feature the "System Assistant" function.

2.1 Ascotel IntelliGate Basic systems

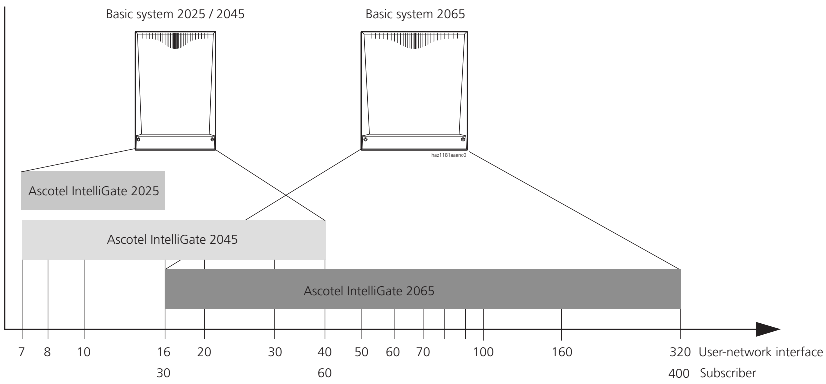


Fig. 1.1: Ascotel IntelliGate Basic systems

Ascotel IntelliGate's three systems are based on two basic systems. One shared system for Ascotel IntelliGate 2025 and Ascotel IntelliGate 2045, and a separate system for Ascotel IntelliGate 2065.
The 2025 / 2045 basic system is already equipped with 7 user-network interfaces and 3 switchable ISDN interfaces, with an additional 5 slots available for expansion cards.

Ascotel IntelliGate 2025 and Ascotel IntelliGate 2045 differ through their system limits. An Ascotel IntelliGate 2025 can be expanded to an Ascotel IntelliGate 2045 by purchasing an appropriate licence. This does not require changing the basis system, and existing expansion cards can still be used.

Ascotel IntelliGate 2065 is the family's big brother, with a total of 14 expansion slots, but without user-network interfaces and network interfaces in its basic configuration.

Both basic systems feature an Ethernet interface for connecting AIMS or a CTI server. There are also two V.24 interfaces, e.g. for connecting a printer or AIMS, in the event the Ethernet interface is occupied, as well as a socket for an external music source and a potential-free relay contact for hooking up a general bell. The audio input can accommodate a music source that plays music for queued calls.

2.2 Expansion cards, modules and auxiliary power supply

A full complement of expansion cards consisting of trunk cards, subscriber cards and special cards connects the system with its environment, or enables special functions.

The trunk cards contain interfaces for connection to the analogue or digital public network (ISDN), or for networking PBX systems to create a private telephony network.

Subscriber cards are used for connecting digital and analogue voice and data terminals such as:

- Office System terminals
- DECT radio units
- Pocket Adapter
- Standard ISDN equipment
- Analogue terminals
- Fax machines
- Answering machines
- External Voice Mail interfaces

Special cards are used in the following cases:

- Networking Ascotel systems via an existing private IP data network
- the Ascotel Voice Mail System, mobile telephony (DECT) and full-duplex handsfree mode
- for connecting door intercoms

The UPS-12V module ensures that the system is able to operate without interruption in the event of a 230 V mains failure. Ascotel IntelliGate can be powered by a 12V backup battery via the module. The DC-48V module is used to power the PBX from a 48V central battery. Both modules are optional and do not take up an expansion slot.

The full spectrum of expansion cards and modules can be used for all Ascotel IntelliGate systems (including rack versions), within the system limits (exception, see "Expansion cards", page 583). As a result logistics at all levels are very simple.

The external Auxiliary Terminal Power Supply (ATPS) is used to increase the supply power available for the terminals. It is required only for system expansions with a large number of terminals with high power requirements (e.g. a large numbe of DECT radio units without their own power supplies.)

2.3 Interface Overview

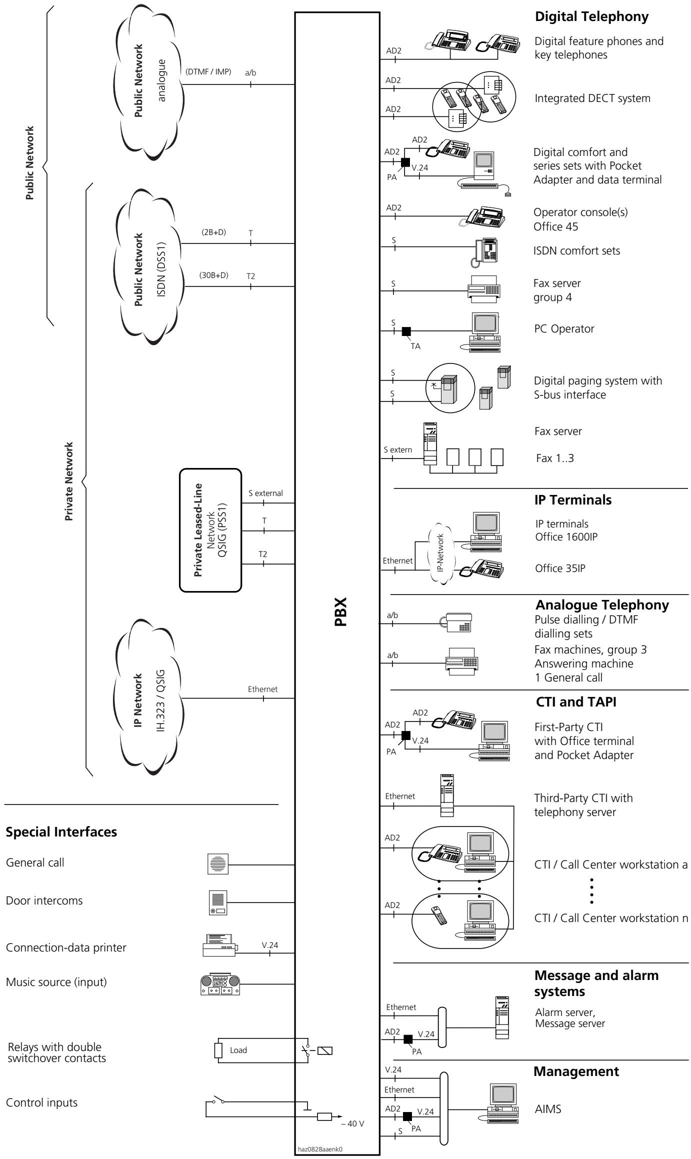


Fig. 1.2: Overview of interfaces with possible terminal equipment

2.4 Expansion Stages and System Limits

Tab. 1.1: Expansion stages of the system family with a few important system limits

	Ascotel IntelliGate 2025	Ascotel IntelliGate 2045	Ascotel IntelliGate 2065
Expansion slots	5	5	14
Network interfaces, total (a/b, T, T2, S external)	8	19	64
Total user-network interfaces (AD2, a/b, S)	16	40	320
AD2 user-network interfaces	12	36	320
S user-network interfaces	7	11	64
Analogue user-network interfaces (a/b) DTMF/PD (incl. voice mail ports)	12	23	168/56[1)]
Subscriber with own number[2)] (incl. DECT subscriber)	30	60	400
Radio units	4	32[3)]	64
DECT subscribers	20	40	150

1) For performance reasons only 56 terminals with pulse dialling are permitted.

2) Incl. pager, DECT subscribers, virtual subscribers, etc.

3) Maximum of 18 SB-8 radio units for operation on two AD2 interfaces in each case

See also:
For more details see "Expansion Stages", page 579 and "System and Expansion Limits", page 592.

3 System Interfaces

The following table provides an overview of the system's different interfaces and their design:

Tab. 1.2: System interfaces and channels

Term	Explanation
B channel D channel	User information channel: Each connection occupies one user information channel, e.g. 2 user information channels (connections) can be occupied simultaneously using one basic access. Control and signalling channel: Channel for control and signalling as well as for packet data transfer.
2B+D / 30B+D	2 2 B channels and 1 D channel / 30 B channels and 1 D channel
Ports	Physical connection points on the PBX for network interfaces and user-network interfaces
Network interfaces • Analogue network interface (a/b network interface) • Basic Access T • Primary rate access T2 • Basic access S external	Network-side connection possibilities for the PBX An analogue network connection has 1 user information channel. Digital network interface 2B+D Digital network interface 30B+D Digital network interface 2B+D: A user-network interface S configured as "EXTERNALS".
User-network interfaces • Analogue user-network interfaces (a/b user-network interface) • ISDN user-network interface (S user-network interface) • AD2 user-network interfaces (AD2 user-network interface)	User-side connection options for the PBX An analogue user-network interface has 1 user information channel. Digital user-network interface 2B+D: Connection for Euro ISDN terminals, PC Operator or paging system. Digital user-network interface 2B+D: A maximum of two AD2 system terminals, one system terminal and a Pocket Adapter, or one DECT radio unit can be operated on a proprietary AD2 bus.
Special interfaces • Ethernet interface on the basic system • Ethernet interface on the expansion card AIP 6400 • Ethernet interfaces on the expansion card AIP 6350 • V.24 interfaces • Door Intercom Systems • General Bell	Other PBX connection possibilities central interface for connecting AIMS or a CTI server, directly or via LAN Interface to an IP network for networking Ascotel systems Interface to an IP network for linking up IP system terminals to an Ascotel IntelliGate system Serial data interface Special interface for connecting door intercoms Special interface for general bell

3.1 Network Interfaces

The system supports the following types of network interfaces:

- Basic access T for connection to
  - the public ISDN network
  - the private leased-line network
- Basic access S external for connection to
  - the private leased-line network
  - a terminal with its own direct dialling plan (DDO)
- Primary rate access T2 for connection to
  - the public ISDN network
  - the private leased-line network
- Analogue network interface for connection to the public analogue network

3.1.1 Basic Access Variants

A basic access is a digital network interface for connection to the public network or to the private leased-line network. It can be set for the protocols DSS1 (public ISDN network) and QSIG / PSS1 (private leased-line network).

A basic access has two 64 kbit/s user information channels and one 16 kbit/s control and signalling channel (2B+D).
One call or data connection can be set up on each bearer channel (B channel) simultaneously and independently of the other B channel.

A basic access can be barred for outgoing calls ("Outgoing Calls Barred" setting).

Basic accesses for connecting the PBX to the public network can be operated as point-to-point and, with some network providers, also as point-to-multipoint (multiple subscriber number) access.

There are two types of basic access:

- Basic Access T
- Basic access S external

3.1.1.1 Basic Access T

Basic access T is suitable for connection to both the public ISDN network and the private-leased-line network.

3.1.1.2 Basic Access S External

The basic access S external is an S interface configured as external ("S-bus protocol = EXTERNALS" setting in the interface configuration).

The basic access S external is designed for the following purposes:

- For connection to the private leased-line network or
- For connecting DSS1 terminal equipment, which evaluates the DDI number sent by the PBX and routes the call accordingly
(e.g. a fax server, see "Call to a DSS1 Terminal equipment on the S Bus (DDO)", page 302)

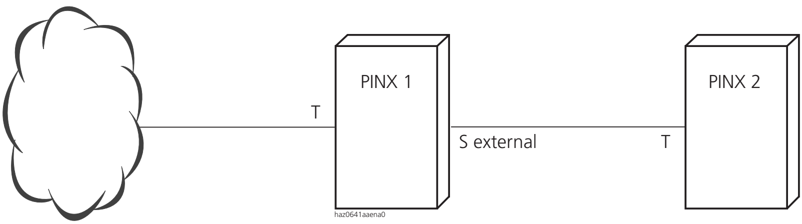


Fig. 1.3: S external in a private leased-line network: PINX-PINX connection

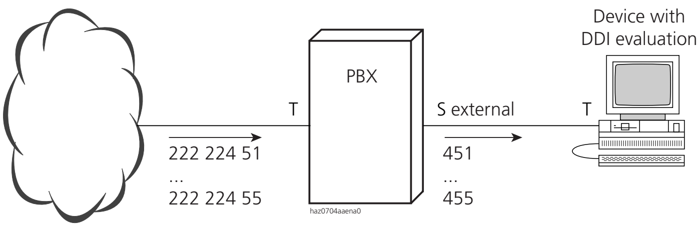


Fig. 1.4: S external in a DDI configuration

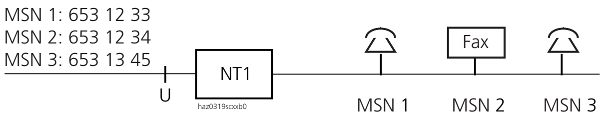
Note:
An S interface configured as external is a fully-fledged network interface and is no longer available as a user-network interface. A basic access S external cannot be used as a connection to the public ISDN network.

3.1.1.3 Point-to-Point and Point-to-Multipoint Connections

Basic accesses can be configured as point-to-point or as point-to-multipoint ("TEI Management" setting in the configuration of the network interfaces).

Point-to-Multipoint Connection without a PBX

The basic access in point-to-multipoint configuration allows a selective dial-up of the terminals connected in parallel using MSN, the Multiple Subscriber Number. Here the network itself provides a kind of direct dialling, so to speak.



NT1: Network Termination
MSN: Multiple Subscriber Number

Fig. 1.5: Single basic access in point-to-multipoint configuration

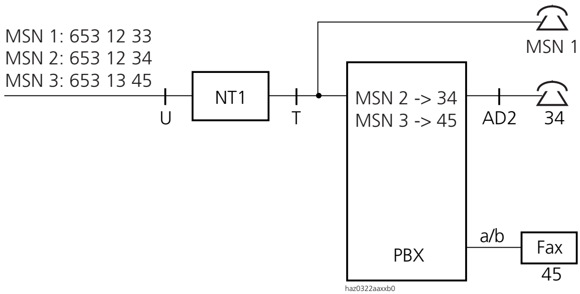
Note:
The fax with ISDN connection is implemented as a fax card in a PC.

Initialization setting

Digital network interfaces are set on point-to-point configuration.

Point-to-Multipoint Connection with PBX

If a PBX is connected using point-to-multipoint, a direct dial number must be created for each MSN number, with all the digits of the MSN number.



NT1: Network Terminal
MSN: Multiple Subscriber Number
U/T: ISDN reference point
AD2: Digital user-network interface AD2
a/b: Analogue user-network interface a/b

Fig. 1.6: Basic access in point-to-multipoint configuration, with single-digit direct dial and parallel terminal

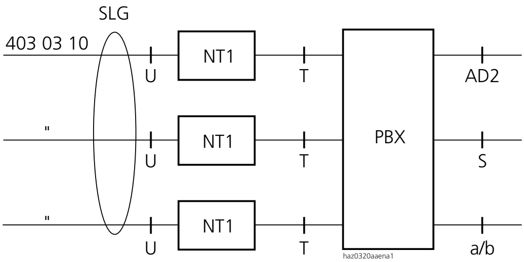
Combinations are also possible in the case of several lines, e.g. one line in point-to-multipoint configuration and the remaining in point-to-point configuration.

Note:
If terminals (e.g. MSN1) are connected in parallel on the T interface, Collision Detection has to be activated as the PBX and the terminal influence each another. This also applies in cases where a/b connections are used on NT1.

Point-to-Point Connection without Direct Dial

Without direct dialling in, only one call number is available. The individual PBX subscribers can only be reached indirectly via the number.

This variant is suitable above all for systems with primarily outgoing traffic.



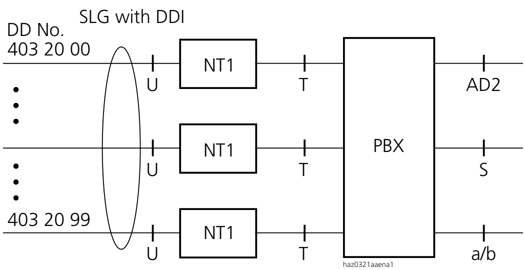
NT1: Network Terminal
SLG: Subscriber Line Group
U/T: ISDN reference point
AD2: Digital user-network interface AD2
S: User-network interface S
a/b: Analogue user-network interface a/b

Fig. 1.7: Several basic accesses in point-to-point configuration,without direct dial number

Note:
Do not connect any terminals between the NT1 and the PBX.

Point-to-Point Connection with Direct Dial

With direct dial the individual PBX subscribers can be reached directly via their direct dial number.



NT1: Network Terminal
SLG: Subscriber Line Group
DDI: Direct dialling
U/T: ISDN reference point
AD2: Digital user-network interface AD2
S: User-network interface S
a/b: Analogue user-network interface

Fig. 1.8: Several basic accesses in point-to-point configuration, with direct dial number

Note:
Do not connect any terminals between the NT1 and the PBX.

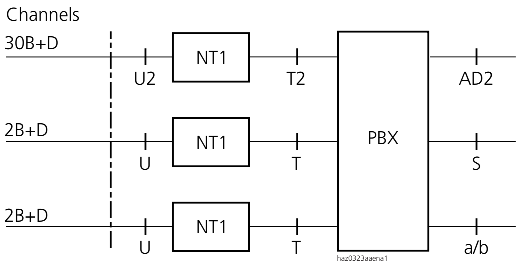
See also:
"Periodic Reactivation of Layer 2 on the T-Interface", page 604

3.1.2 Primary Rate Access T2

A primary rate access is a digital network interface for connection to the public network or the private leased-line network. It can be set for the protocols DSS1 (public ISDN network) and QSIG / PSS1 (private leased-line network).

A primary rate access has thirty 64 kbit/s user information channels and one 64 kbit/s control and signalling channel (30B+D). One call or data connection can be set up on each bearer channel (B channel) simultaneously and independently of the other B channel.

Note:
Primary rate accesses can only be used as point-to-point connections.



NT1: Network Terminal
U2/U/T2/T: ISDN reference points
30B+D: Primary rate access channels
2B+D: Basic access channels
AD2: Digital user-network interface AD2
S: User-network interface S
a/b: Analogue user-network interface a/b

Fig. 1.9: System with basic and primary rate accesses

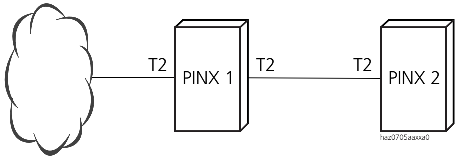


Fig. 1.10: Primary rate access in a private leased-line network: PINX-PINX connection

3.1.3 Analogue Network Interfaces

The analogue network interfaces support DTMF and pulse dialling. A range of parameters in the System Configurations allows country-specific adaptations to the public network as well as other settings. The table below shows the configuration options available:

System configuration

Tab. 1.3: Analogue network interfaces: System configuration

Parameter	Parameter value	Remarks
Behind PBX	[Yes / No]	See "Analogue down-circuit connection", page 605
Line Attenuation	[short / long / short D long D]	See "Attenuation on analogue network interfaces", page 604
Dialling mode	[PULSE / DTMF]	DTMF dialling should be used in preference whenever both dialling types are supported.
Ringing cycle	[5..60 seconds]	With incoming calls the internal ringing signal is discontinued if the time between each ringing signal on the exchange line is longer than the configured ringing cycle. This is the case for example when the external caller hangs up.
Dialling tone detection	[Yes / No]	If "Yes", the PBX waits for the dial tone from the exchange before starting to dial.
Dialling tone time	[0..1200 seconds]	Maximum waiting time for the exchange dial tone if exchange dial tone detection is activated. After that, the PBX switches over to the next free trunk line. If dial tone detection is deactivated, dialling begins after the set time, even without an exchange dial tone

Parameter	Parameter value	Remarks
International dialling tone	[Yes / No]	If "Yes", the dialling process is interrupted after one of 10 predefined digit sequences to wait for the international dialling tone.
Exchange digit barring	[No international dialling tone or 1…10]	If the trunk line is configured to "Behind PBX" (analogue down-circuit connection), the exchange access prefix of the up-circuit PBX has to be entered in the exchange digit barring.
Release signal	[Yes / No]	In most cases the public network sends the PBX a release signal whenever the external subscriber ends the call. If the parameter is configured to "Yes", the connection will be subsequently cleared down by the PBX (see also "Clearing down Exchange-to-Exchange Connections", page 323). Note: On analogue network interfaces the only release criterion the PBX recognises is a loop interruption, not a polarity reversal or busy tone, etc.

3.2 User-Network Interfaces

The PBX supports digital and analogue user-network interfaces.

3.2.1 Digital user-network interfaces

On each of these digital interfaces several appropriate terminals can be hooked up and powered simultaneously.

Each of these interfaces has two 64 kbit/s user information channels and one 16 kbit/s control and signalling channel (2B+D). This makes it possible to establish two independent call or data connections simultaneously.

3.2.1.1 S user-network interface

The S user-network interface is a digital 4-wire interface used for connecting system terminals, ISDN terminals, Terminal Adapters and ISDN PC cards. The PC Operator is connected via a special Terminal Adapter.

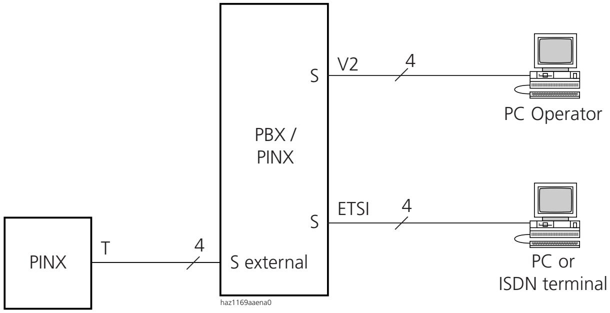


Fig. 1.11: S user-network interface

Up to 8 terminals can be operated on an S user-network interface. They are addressed with the single-digit terminal selection digit (TSD).

There are three available modes ("S bus protocol" setting in the interface configuration) for operating the interface S:

- The "V2" mode is used to operate the PC Operator Office 1550 and the paging system.
- The "ETSI" mode is used to operate ISDN terminals, Terminal Adapters and ISDN PC cards.
  For the ISDN services supported, see "ISDN services supported by the system", page 424.
- With the "EXTERNS" mode an S interface can be used as a basic access S for private networking with QSIG / PSS1 or DSS1. It is then no longer available as a user-network interface (see "Basic Access S External", page 61).

Format of the ETSI S-bus

The format on the ETSI S-bus can be configured in the interface configuration for each S interface.

Tab. 1.4: System configuration: Format of the ETSI S-bus

Parameter	Parameter value	Remarks
MSN format on the S-bus	• TSD	• Single-digit terminal selection digit (TSD) as per interface configuration
	• Subscriber No.	• Default setting • Mode of operation as customary in the public ISDN network
	• DDI No.	• For special applications, e.g. Unified Messaging Systems • If the DDI number is missing, the system attempts to transmit one of the following numbers, in the sequence shown below: – Number of the CDE – UG number – Subscriber number • Also functions internally

Note:
If the parameter "S-bus Protocol" is configured to "V2" in the system configuration, the format is not diallable on the S-bus. In this case the TSD is transmitted

Exchange Access Prefix for Terminals on the ETSI S Bus

For terminals on the ETSI S-bus the interface configuration can be used to select whether or not the exchange access prefix of the CLIP should be truncated for incoming calls (setting "Delete exchange access prefix" = "Yes" / "No", Default = No). This setting is effective only in the S-bus mode ("S bus protocol = ETSI").

See also:
"Voice and data terminals on the S interface", page 605

3.2.1.2 AD2 user-network interfaces

The AD2 digital user-network interface is a proprietary, system-specific 2-wire interface used for connecting system terminals:

- Corded system terminals
- V.24 interface Pocket Adapter
- Ascotel DECT Radio units

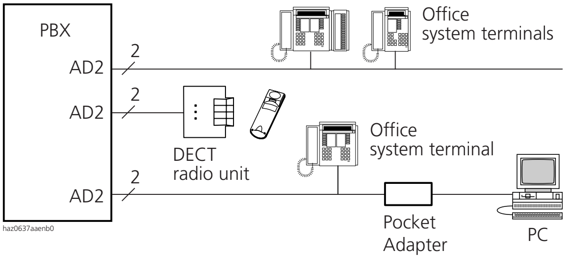


Fig. 1.12: AD2 user-network interfaces

Two system terminals can be connected to an AD2 user-network interface. Address allocation is done by means of a switch on the terminal.

3.2.2 Analogue User-Network Interfaces

This 2-wire interface supports the following off-the-shelf analogue terminals:

- Terminal with DTMF or pulse dialling (corded and cordless)
- Group 3 fax machines
- Answering machines
- Modem

Neither call charges nor CLIP information is transmitted to the connected terminals via analogue user-network interfaces.

One analogue user-network interface per system can be configured for connecting a general bell (see "General Bell on an Analogue user-network interface a/b", page 522).

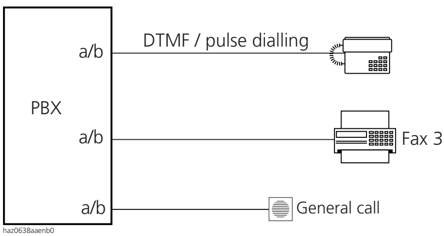


Fig. 1.13: User-network interface a/b

3.3 Special Interfaces

The system supports a range of special interfaces.

3.3.1 Ethernet Interface

The system provides two types of Ethernet interfaces:

- Ethernet interface on the basic system for data exchange with AIMS and for connecting a CTI server
- Ethernet interface on the AIP 6400 expansion card for networking Ascotel systems and on the AIP-6350 expansion card for linking up IP system terminals to an Ascotel IntelliGate system.

3.3.2 V.24 Interface

The V.24 interface is a serial interface. Data transfer is asymmetrical, bipolar with a minimum of +/- 4 V over a minimum of 3 wires (SGND, TXD, RXD). Control signals are transmitted by code (XON, XOFF) or with additional lines for hardware flow control.

The system provides several V.24 interfaces:

- Two V.24 interfaces directly on the mainboard
- V.24 interfaces on user-network interfaces via the Pocket Adapter

Not all V.24 interfaces on the system have the same functionality. The following table shows how the individual interfaces are used:

Tab. 1.5: Functionality and application of the V.24 interfaces on the system

Function / Application	V.24 on the Mainboard	V.24 on the Pocket Adapter
Call data output (OCL / ICL) via printer or PC	✓	✓
Individual charge data output (ICC) via printer or PC	✓	✓
Dialling from PC with AT commands	–	✓
Dialling from PC with Unimodem on Windows	–	✓
TAPI TSPI PC Windows	–	✓
Ascotel Messaging System	✓	✓
Data service with PC	–	–
Interface to hotel management systems	✓	–
PBX configuration with PC (AIMS)	✓	✓ [1)]

1) PA Version ≥ V2.4

For more information on the V.24 interfaces on the system see "V.24 Interface", page 788.

3.3.3 Interface for Door Intercom System

Door intercom systems can be connected to the system via the OI-2DOOR expansion card. A bell key is backed by an internal destination. The door intercom system can be addressed via an internal number.

A loudspeaker system can also be operated via the interface for door intercom systems.

3.3.4 Interface for General Bell

Calls can also be routed to the general bell. Bells or lamps connected to the general bell interface signal calls which can be answered by anyone from any subscriber's phone.

With "Coded ringing" can be used to assign different ringing patterns to different destination persons or groups and, in this way, create a simple type of paging system.

Tip:
One analogue user-network interface per system can be reconfigured in such a way that it is also used for connecting a general bell. This eliminates the need for an external ringing voltage source (see "General Bell on an Analogue user-network interface a/b", page 522).

4 Networking the Systems

Several systems can be combined into a private network (Private Integrated Services Network: PISN) There are different ways of networking systems into a PISN:

- QSIG networking over IP network (see page 80):
  Customers who already operate a voice-compatible IP data network between different locations can use the network via IP Gateway to transmit voice traffic, also (see "Ascotel IP Gateway AIP 6400", page 124) .
- Networking via leased lines (leased-line networking, see page 81):
  If the systems are networked via ISDN leased lines, the only costs incurred as for the permanent leased lines, not for call charges.
- Virtual networking (see page 81):
  In virtual networking, traffic between the different locations is handled via the public ISDN network. This type of networking is well suited, for example, for a low call volume or for integrating field staff into the corporate network.

The system supports all three networking types as well as hybrid forms. The networking type best suited depends on the individual circumstances (see "Choosing the Appropriate Networking Type", page 84).

Networking technology terminology

Before the basics of private networking are explained, the table below defines the key terminology for networking technology:

Tab. 1.6: Networking technology terminology

Abbr.	Term	Explanation
–	Exchange	Short for → public network
–	Break-in	An external incoming → DDI connection for a PISN subscriber is routed into the → PISN at the → PINX that is closest to the caller.
–	Break-out	An outgoing external connection is routed into the public ISDN only at the PINX that is closest to the call destination.
CTX	Centrex	The designation Centrex, for Central Office Exchange Service, is a product name which some network providers use for the services provided by the → virtual PBX.
DSS1	Digital Subscriber Signalling 1	Signalling protocol for ISDN networks (also called Euro-ISDN)

Abbr.	Term	Explanation
DDI	Direct Dialling In	DDI numbers enable internal subscribers to be reached directly from a public network. In the direct dialling plan the end portion of a call number is allocated to the number of an internal subscribers or → PISN subscriber. Several direct dialling plans can be drawn up for each PBX.
DDO	Direct Dialling Out	The PBX can forward direct dial numbers to the private leased-line network via the interface S external.
E.164	–	• Numbering plan identifier of the public network as per ITU-T • Parameter value of parameter → NPI
–	Gateway-PINX	A PINX is a gateway PINX for the duration of a connection if it routes that connection from the PISN to the public network or from the public network to the PISN.
–	Node	Branch point or end point in a communication network
LCR	Least Cost Routing	Routing function used to determine the network operators via which a call is to be routed. Usually the most cost-effective route (least cost) is selected. LCR is also needed for the overflow from the private to the public network.
–	Dialling by name	If a name is stored under a call number, the name can be used for dialling on the terminal instead of the call number.
–	QSIG networking in the IP network	Networking via an existing voice-compatible IP data network
NPI	Numbering plan identifier (Numbering Plan Identifier)	• In the public network, the numbering plan identifier used is →E.164. In the private area, the numbering plan identifier used is → PNP • Configuration parameter used for specifying the numbering plan identifier. Parameter values: E.164 / PNP / unknown
–	Overflow	If the chosen line of a→ PINX is not available due to overloading or due to a defect, the connection pending is set up via an alternative path determined by the configuration.
PISN	Private Integrated Services Network	Private network based on the ISDN standard in which all the connected subscribers can communicate with one another as internal subscribers. This applies to both voice traffic and to ISDN-based data traffic.
–	PISN Subscriber	• Subscriber in a different → node of a private network • Category in the internal numbering plan used to replicate the subscribers in the private network
PINX	Private Integrated Services Network eXchange	→ node of a → PISN, usually a PINX is an ISDN PBX.
PNP	Private Numbering Plan	• Internal numbering plan of a PBX or PINX • Service offered by the network provider. Essentially a virtual counterpart to a PBX numbering plan. Most important component of a → virtual PBX. • Parameter value of parameter → NPI
–	Private leased-line network	Private network implemented using dedicated lines. In the system configuration we need to differentiate between the private leased-line network and the public network.

Abbr.	Term	Explanation
QSIG/ PSS1	QSIG / PSS1 protocol	ECMA-standardized signalling protocol used for networking several → PINXs. Now standardized worldwide (ISO / IEC) under the name PSS1 (Private Signalling System 1) • Parameter value of the "Protocol" trunk group parameter. The system supports two versions of the QSIG / PSS1 protocol: QSIG (ETSI, 2nd edition) and QSIG/PSS1 ISO.
TON	Type of number	Parameter used for classifying a call number: Parameter values, if the call number corresponds to a NPI = E.164: unknown / subscriber / national / international. Parameter values, if the call number corresponds to a NPI = PNP: unknown / level 0 / level 1 / level 2
–	Transit PINX	A PINX acts as a transit PINX for the duration of a connection if it routes that connection from one PINX to another PINX.
–	Source PINX	A PINX acts as a source PINX for the duration of a connection if the connection was set up by one of its subscribers.
–	Virtual PBX	Network provider offer which comprises a → PNP and various ISDN supplementary services. Also known under the name → Centrex. With a virtual PBX the network provider is able to offer his customers the functionality of a PBX.
–	Destination PINX	A PINX acts as a destination PINX for the duration of a connection if the connection's destination subscriber is one of its subscribers.

4.1 Private ISDN-Based Network (PISN)

A private network based on the ISDN standard is referred to as a PISN (Private Integrated Services Network). Its characteristic feature is that all connected subscribers can communicate with one another in the same way as PBX-internal subscribers. This applies to both voice traffic and to ISDN-based data traffic. A PISN is the keystone of any corporate communication structure optimized in terms of flexibility, user convenience and cost.

A network is generally defined by its nodes and the connections between those nodes. In a PISN they are formed by ISDN PBXs. A PISN consists of at least two nodes and can be structured both as a meshed network and as a star-shaped network. Hybrid forms are also admissible.

Usually a PISN is connected to the public network in at least 1 place.

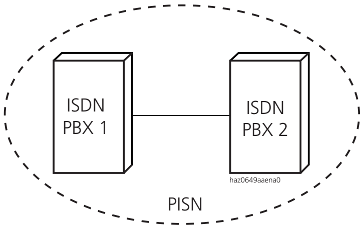


Fig. 1.14: PISN with at least two nodes

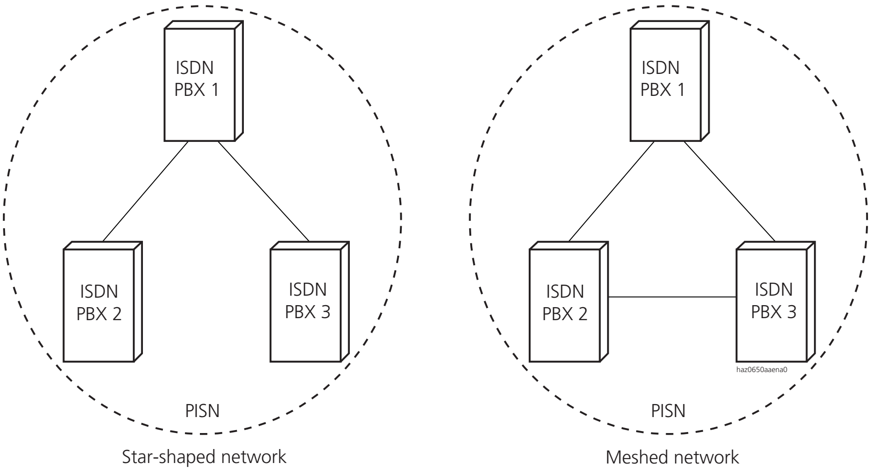


Fig. 1.15: Star-shaped and meshed PISN

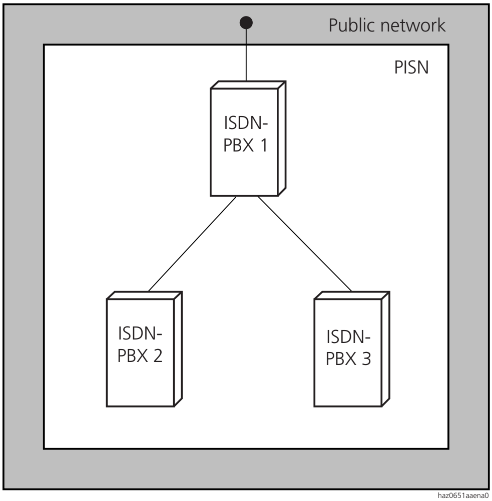


Fig. 1.16: A PISN usually has at least 1 connection to the public network

Networking Philosophy

Owing to the system's sophisticated networking options, it is possible to set up a whole range of private networks (PISN). Homogeneous networks exclusively with PBXs of the Ascotel system family are as easily implemented as heterogeneous networks with a mix of systems and PBXs by other manufacturers.

The system's networking philosophy is based on the assumption that the user does not need any prior knowledge of the network topology. He can use the available features always in the same way, regardless of whether and how he is networked.

Homogeneous and Heterogeneous PISN

Each PBX forms a node which is networked in a PISN. In network terminology, networked PBXs are referred to as a PINX (for Private Integrated services Network eXchange). A PINX can be set up using the following PBX types:

- ISDN PBX from the Ascotel system family
- ISDN PBX by other manufacturers
- Virtual PBX (private numbering plan from a public network provider)

In a homogeneous network all the PINXs are from the Ascotel system family. If PBXs by other manufacturers are also used in a network, we talk of a heterogeneous network.

4.2 Networking Types (Topologies)

Taking technical, organizational and tariff conditions into account calls for a flexible networking concept so that optimum customer solutions can be implemented. Other factors that influence the choice of networking type are the density and nature of the communication relations. The following chapters provide explanations of the different networking types.

4.2.1 QSIG networking over IP network

With QSIG networking over IP network voice data and QSIG signalling are transmitted in the Intranet as IP data packets. The intranet with QSIG / PSS1 supports the same scope of features as networking via leased lines (this is also referred to as "QSIG tunnelling").

If an existing intranet can be utized for voice connections (Quality of Service: QoS must be supported), ISDN network call charges do not apply (tollbypass). If all the voice channels of the IP Gateway are busy, the calls can automatically be routed via the ISDN network (see "Testing overflow routing in the PISN", page 343). An IP Gateway is required for each PBX.

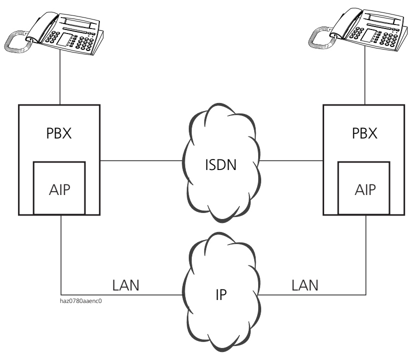
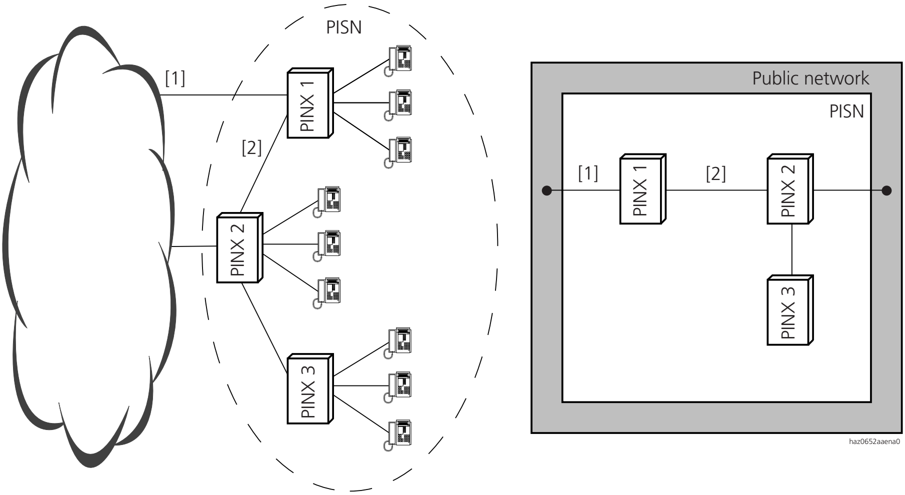


Fig. 1.17: QSIG over IP network (AIP LAN route)

4.2.2 Leased-line Networking (Leased-line Network)

With this type of networking, the PINXs are connected with specific or leased dedicated lines. The characteristics of this type of networking are:

- fixed line resources
- fixed costs



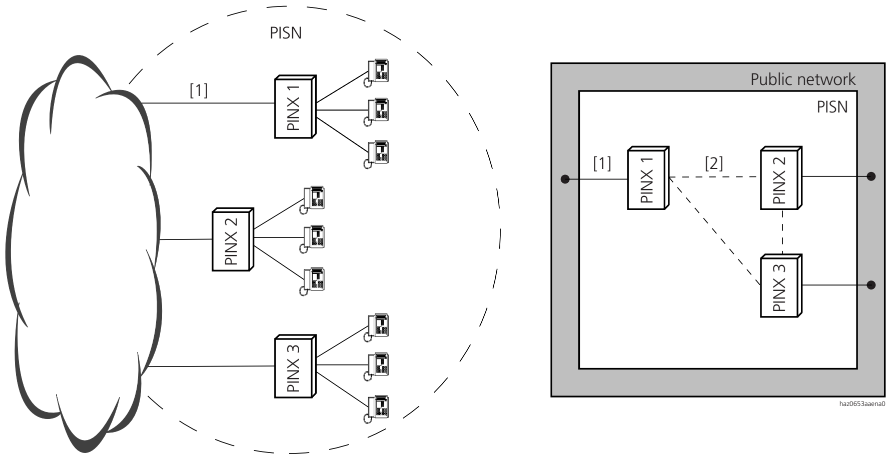
[1] Connection to the public network
[2] Physical connection between two PINXs

Fig. 1.18: Example of a leased-line network, represented in two different ways

4.2.3 Virtual Networking (Virtual Network)

With this type of networking all the PINXs are connected to the public ISDN network. The connections between the PINX are switched lines, not direct physical connections. The characteristics of this type of networking are:

- Line resources are required for the current connections only.
- Voice and data traffic via the public network is charged according to duration and distance.
- The necessary networking-specific equipment is minimal.
- The range of services available in a virtual network depends on the range of services offered by the network provider.



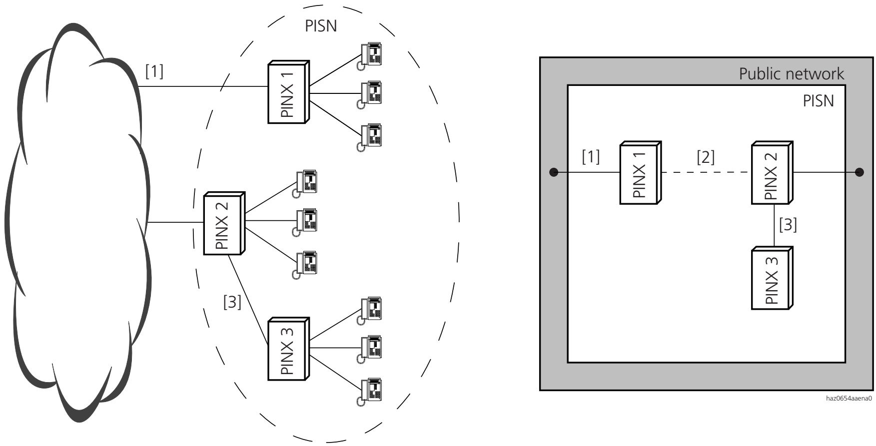
[1] Connection to the public network
[2] Virtual connection between two PINXs

Fig. 1.19: Example of a virtual network

In this example (Fig. 1.19) all three PINXs are virtually interconnected via the public virtual network. This depends on the configuration. If for example there is no need for a virtual connection between PINX 1 and PINX 3, the configuration can be implemented accordingly.

4.2.4 Combining Leased-line and Virtual Networking

With the system, even combinations of leased-line and virtual networking are possible within one and the same PISN.



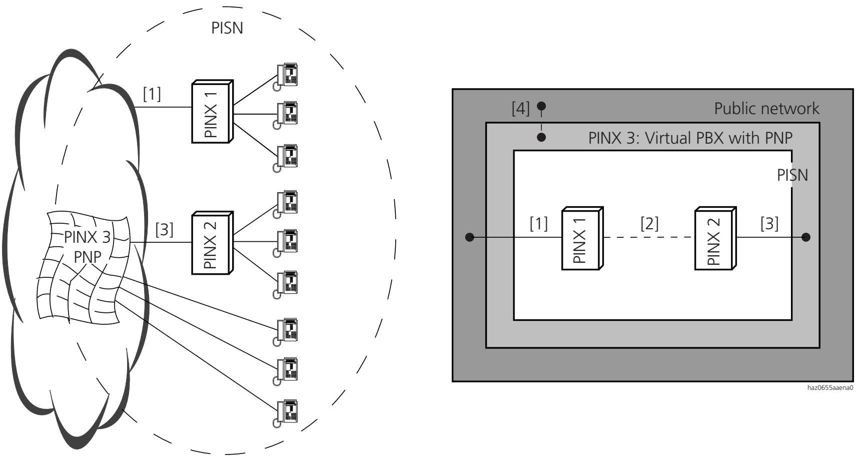
[1] Connection to the public network
[2] Virtual connection between two PINXs
[3] Physical connection between two PINXs

Fig. 1.20: Example of a combined network

4.2.5 Virtual Networking with a Virtual PBX (Centrex)

Some network providers offer a private numbering plan (PNP) as a service. Together with ISDN supplementary services, e. g. rerouting services, this means that the customer has a virtual PBX at his disposal. Some network providers market this facility under the name Centrex while others use the general term VPN (Virtual Private Network).

A virtual PBX can easily be integrated as a PINX in a PISN. The system supports private numbering plans and incorporates them seamlessly into its own network functionality.



[1] Connection to the public network
[2] Virtual connection between two PINXs
[3] Physical connection to a virtual PBX in the public network
[4] Virtual connection between a virtual PINX and the public network

Fig. 1.21: Example of a virtual network with a virtual PBX

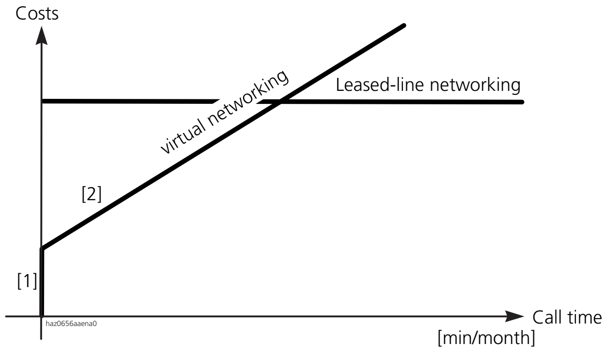
4.3 Choosing the Appropriate Networking Type

The choice of the appropriate topology depends on the following factors, among others:

- Number of locations
- Distance between the locations
- Calling rate between the locations
- Networking equipment required
- Existing data network (see System Manual AIP 6400
- Scope of performance required

With virtual networking, the costs incurred consist of the basic charge for connections and DDI numbers and the charges per call, while a dedicated line incurs only fixed costs (for the rental of the leased line for example). The following comparison (Fig. 1.22) shows that virtual networking is better suited to systems with lower calling rates while networking with leased lines can be more advantageous for

higher calling rates.
As Fig. 1.20 shows, combinations of both networking types are also possible.



[1] Basic charge
[2] Charges per call and time unit

Fig. 1.22: Cost structure between leased-line and virtual networking

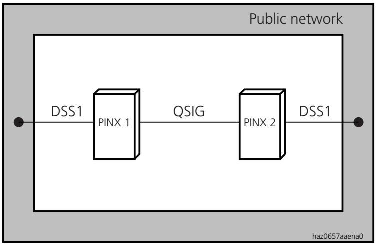
4.4 Communication Protocols

The system supports the two main protocols for setting up a PISN:

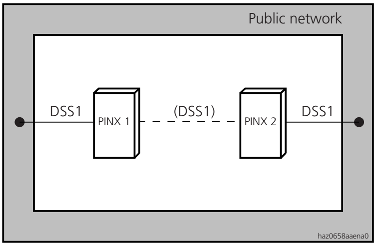
- The ISDN protocol DSS1 is used mainly in virtual networks.
- The QSIG / PSS1 protocol is based on an international standard that is supported by all leading PBX providers. QSIG / PSS1 is used to set up private networks with very high performance and capacity.

4.4.1 Protocol and Topology Type

The system supports any combination of protocol and topology type. Normally, however, the QSIG / PSS1 protocol is used for leased-line networking and the DSS1 protocol for virtual networking.



[1]

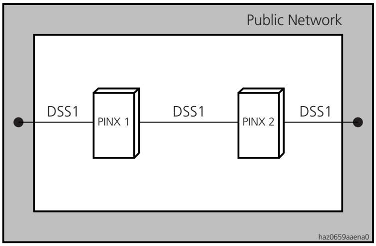


[2]

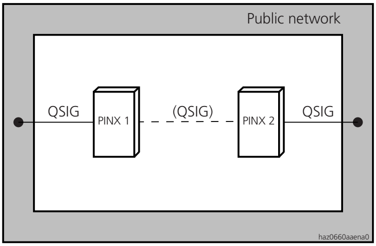
[1] QSIG / PSS1 protocol in a leased-line network
[2] DSS1 protocol in a virtual network

Fig. 1.23: Normally used protocols

In special cases the DSS1 protocol can also be used for leased-line networking and the QSIG / PSS1 protocol for virtual networking.



[1]



[2]

[1] DSS1 protocol in a leased-line network
[2] QSIG / PSS1 protocol in a virtual network

Fig. 1.24: Protocols used in special cases

If different topologies are combined within a PISN, then different protocols are normally also used.

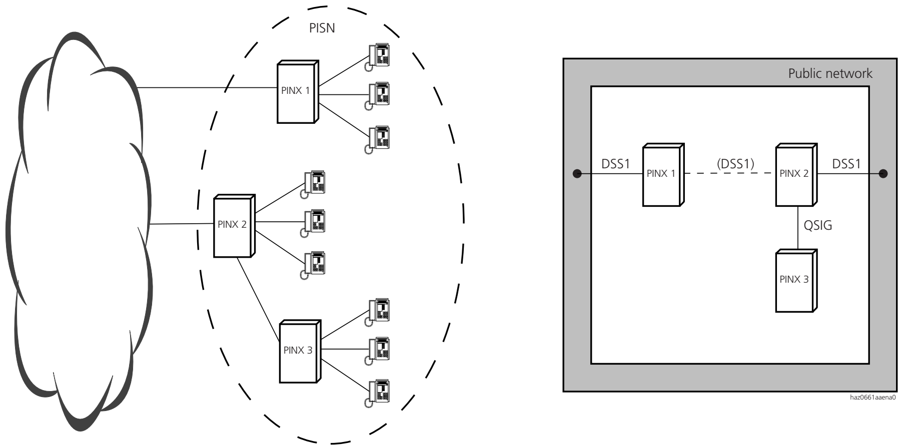


Fig. 1.25: **Combined network: virtual connections with DSS1, fixed connections with QSIG / PSS1**

4.4.2 Connection between Protocol and Range of Services Available

The range of services available in a PISN is determined by the protocol used and the local features of the system. The services offered under QSIG / PSS1 and DSS1 differ only marginally. With virtual networking the range of services available also still depends on the public network provider. The system supports a multitude of the ISDN services on offer and combines them effectively with its own features.

Digital networking is based on the ISDN standard and therefore supports both voice and data traffic.

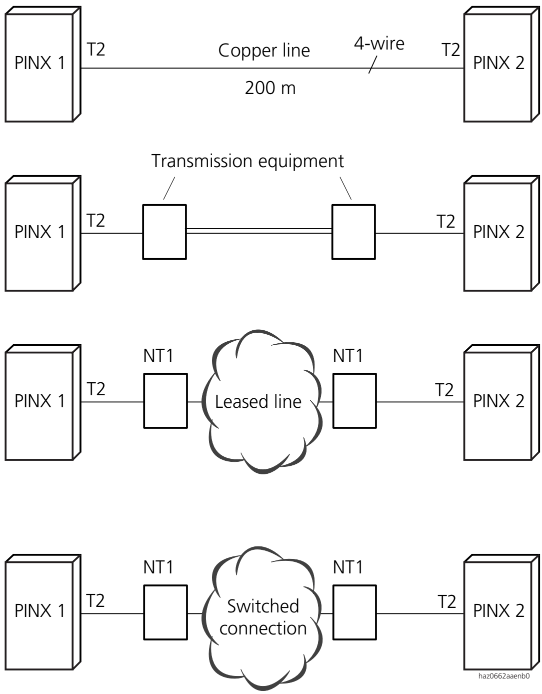
4.5 Connecting PISN Nodes

The nodes of a PISN (PINX) can be connected via basic, primary-rate or Ethernet accesses. For short distances the connection can consist of one copper cable without any ancillary equipment. For longer distances transmission equipment or leased lines of the public network need to be used.

4.5.1 Connection via Basic and Primary Rate Access

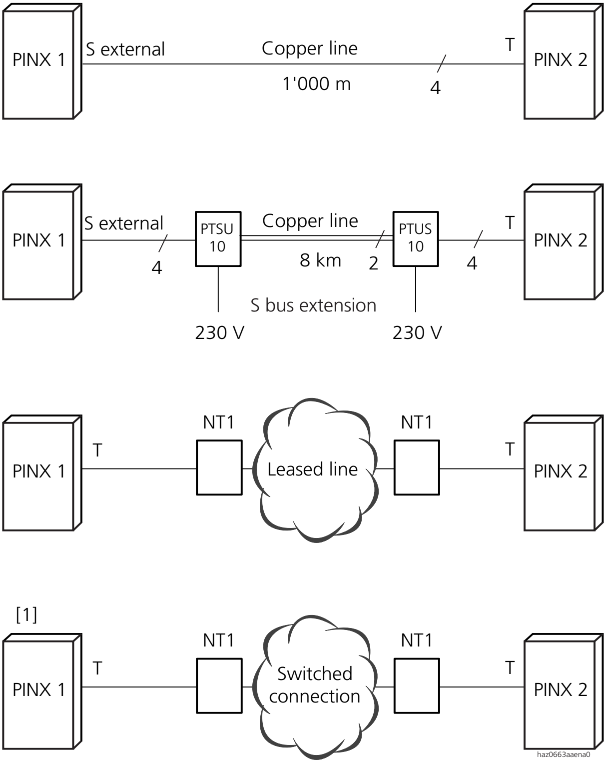
PINX connections are connected to an S external, T or T2 interface. Each of these interfaces can be configured as an interface with QSIG / PSS1 or DSS1 protocol.

With the DSS1 protocol, only Base Call is supported on the S external interface.



T2: T2 primary access on the PBX
NT1: Network termination

Fig. 1.26: Primary access connections between two PINXs



T: T basic access on the PBX
S: "S external" interface on the PBX
[1] Not available in all countries

Fig. 1.27: Basic access connections between two PINXs

4.5.2 Connection via Ethernet Interface

PINX connections are connected to the Ethernet interface of the AIP-6400 special card. Thereby enabling QSIG networking in the IP network. For more details on this connection type, see System Manual AIP 6400.

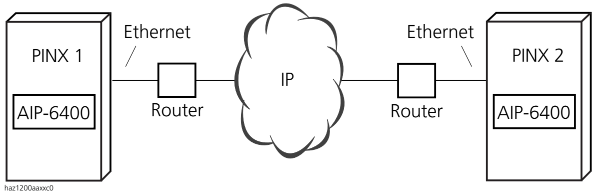


Fig. 1.28: Ethernet connection between 2 PINXs

4.6 Numbering Plan and Regions

Subscribers connected directly to a PINX are always internal subscribers. Subscribers of a different PINX in the same private network are PISN subscribers. Several PINXs can be grouped together into regions. All the regions together form the PISN.

The relationships between PINX and PISN subscribers are specified in the internal numbering plans of the individual PINXs.

4.6.1 Shared Numbering Plan

If two or more PINXs are structured in such a way that they split the range of subscriber numbers among themselves, we talk of a shared numbering plan. Each number can occur in the PISN only once. Together the PINXs form a region, in which all the subscribers can be reached under internal call numbers.

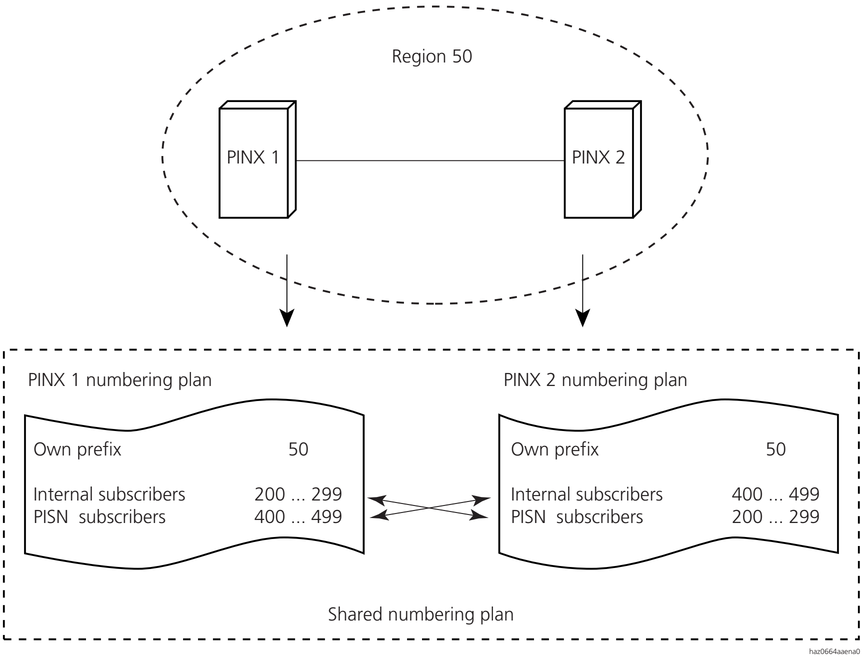


Fig. 1.29: Shared numbering plan: two PINXs share the numbers of a numbering plan.

4.6.2 PISN with Regions

If a PISN is subdivided into several regions, each own regional prefix is determined in the internal numbering plan of each PINX.
Subscribers who call a subscriber in a different region first dial the regional prefix of the destination region, followed by the internal number of the subscriber they want.

The numbering plan is organized independently of the PISN topology.

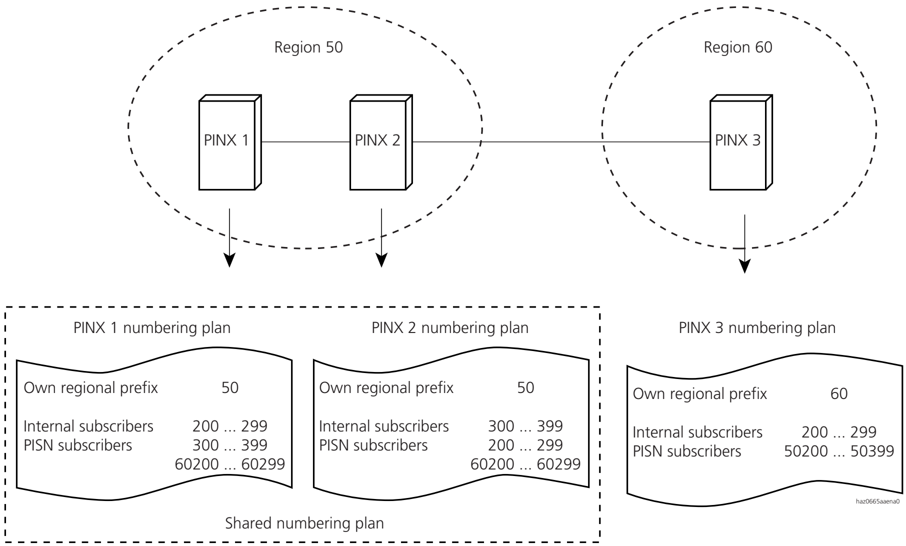


Fig. 1.30: PISN with two regions and shared numbering plan for Region 50

The purpose dividing into regions is that then the existing DDI numbers can continue to be used.

4.6.2.1 Numbering Plan for Two Regions

Even a virtual PBX can easily be integrated into a PISN.

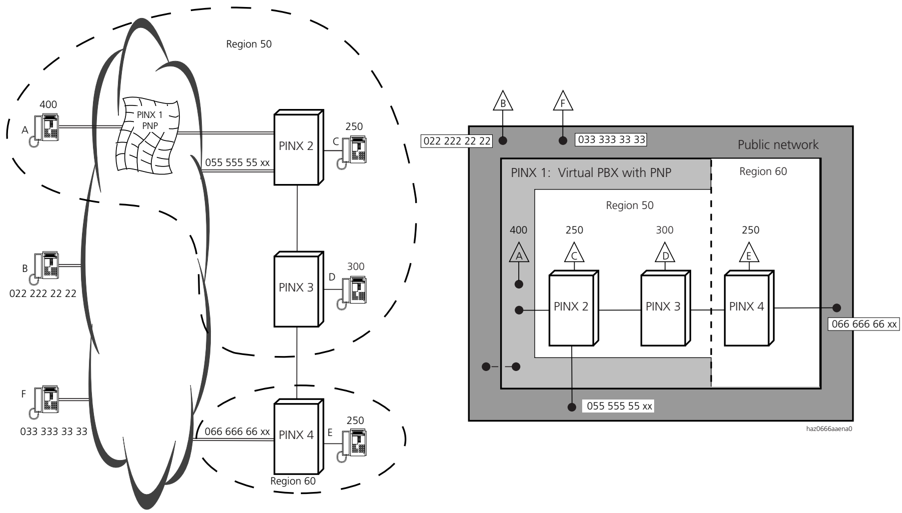


Fig. 1.31: **Example: PISN with two regions and a shared numbering plan for Region 50**

Tab. 1.7: **Entries in the numbering plans for the above example**

Numbering plan of	Specific regional prefix	Internal (local) subscriber	Numbers in the PISN	
			PISN Subscriber	in region
PINX 1	50	400 ... 499	2xx, 3xx 602xx	50 (specific) 60
PINX 2	50	200 ... 299	3xx, 4xx 602xx	50 (specific) 60
PINX 3	50	300 ... 399	2xx, 4xx 602xx	50 (specific) 60
PINX 4	60	200 ... 299	– 502xx to 504xx	60 (specific) 50

PINX 1, 2 and 3 share a numbering plan. PINX 4 has its own, specific numbering plan.

4.6.2.2 Call Routing in Regions

The following illustrates how calls are routed in a PISN with regions.

Call within the Same Region

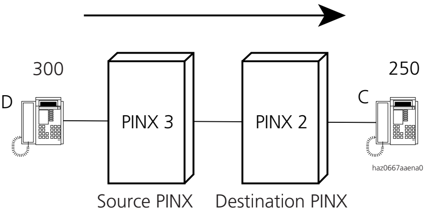


Fig. 1.32: Subscriber D dials 250 (subscriber C)

The call is routed as follows:

1. In PINX 3's internal numbering plan the number 250 is entered as a PISN subscriber. The call is routed to PINX 2 via the allocated route.
2. Under the number 250 PINX 2 finds the internal subscriber C. The connection is set up.

Call to a Different Region

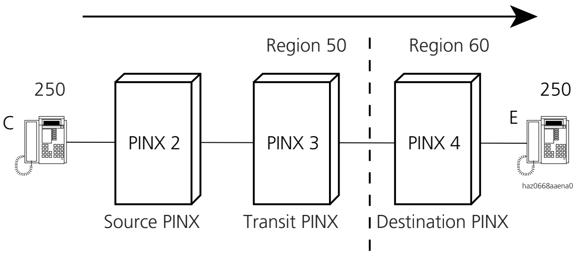


Fig. 1.33: Subscriber C dials 60 250 (subscriber E)

The call is routed as follows:

1. In PINX 2's internal numbering plan the number 60250 is known for a PISN subscriber in a different network region. The call is routed to PINX 3 via the allocated route.
2. In PINX 3's internal numbering plan the number 60250 is also known. The call is routed to PINX 4 via the allocated route.

3. PINX 4 recognizes the number 60 as a specific prefix and truncates the digits. Under the number 250 it finds the internal subscriber E. The connection is set up.

Call to a Virtual PBX (Centrex)

This is a call within the PISN Region 50.

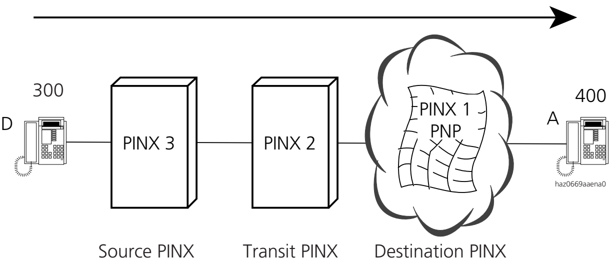


Fig. 1.34: Subscriber D dials 400 (subscriber A on the PBX)

The call is routed as follows:

1. In PINX 3’s internal numbering plan the number 400 is entered as a PISN subscriber. The call is routed to PINX 2 via the allocated route.
2. In PINX 2’s internal numbering plan the number 400 is also entered as a PISN subscriber. The call is routed to PINX 1 via the allocated route.
3. Under the number 400 PINX 1 finds the internal subscriber A. The connection is set up.

Call from the Public Network

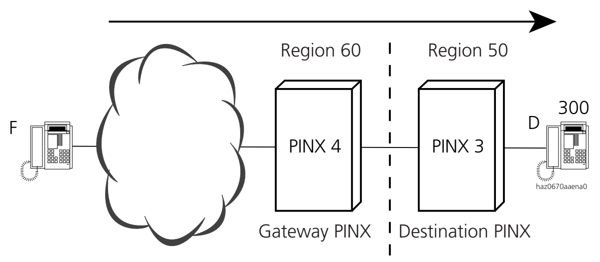


Fig. 1.35: Subscriber F dials 066 666 66 30 (subscriber D)

The call is routed as follows:

1. In PINX 4's direct dialling plan the number 630 is linked with the PISN subscriber number 50 300. The call is routed to PINX 3 via the allocated route. The configuration of the routes and DDI numbers is done via AIMS.
2. PINX 3 recognizes the first 2 digits (50) as a specific prefix and truncates them. Under the number 300 PINX 3 finds the internal subscriber D. The connection is set up.

4.7 Network-Related Services and Functions

Provided the networks and terminals have the technical prerequisites, the system supports a whole series of conventional services and functions.

4.7.1 Services Supported in a Networked Environment

The system supports the services listed in the following table:

Tab. 1.8: Network-related services supported by the system

Abbr.	Designation		Remarks
	International	System	
	Base Call	Basic service	
CCBS	Call Completion to Busy subscriber	Callback if busy	Possible everywhere if supported by the public network
CFB	Call Forwarding Busy	Call Forwarding Busy	Executed locally in the PINX concerned, with display on the terminal
CFNR	Call Forwarding No Reply	Call Forwarding on No Reply CFNR	Executed locally in the PINX concerned, with display on the terminal
CFU	Call Forwarding Unconditional	Call Forwarding Unconditional CFU	Executed locally in the PINX concerned, with display on the terminal
3pty	Three-party services		In a heterogeneous network these features also depend on the third-party PINX.
	• Call Transfer by join	Call transfer	Always possible under QSIG / PSS1, with correct display. The transferring PINX becomes a transit PINX.

Abbr.	Designation		Remarks
	International	System	
	• Call Enquiry	Enquiry call	Executed locally in the PINX concerned, with display on the terminal
	• Brokering	Brokering	Executed locally in the PINX concerned, with display on the terminal
	• Conference	Conference	Under QSIG / PSS1 with correct display
	• Recall	Recall	Only under QSIG / PSS1
CLIP	Calling Line Identification Presentation	Caller Identification (Call number)	
CLIR	Calling / Connected Line Identification Restriction	Suppress CLIP	
CNIP	Calling Name Identification Presentation	Caller identification (name)	Defined only in QSIG / PSS1
CNIR	Calling / Connected Name Identification Restriction	Suppress CNIP	Together with CLIR
COLP	Connected Line Identification Presentation	Identification (call number) of the called party	
CONP	Connected Name Identification Presentation	Identification (name) of the called party	Defined only in QSIG / PSS1
DDI	Direct Dialling In	Direct dialling plan	
HOLD	Hold	Hold	
PARE	Partial Rerouting	Partial Rerouting	Not supported in QSIG / PSS1
CD	Call Deflection	Call Deflection	Implemented locally as a subscriber-related feature in the relevant PINX
PNP	Private Numbering Plan	Private numbering plan	Centrex is supported and integrated into a region.
UUS	User-to-User Signalling	User-to-user signalling	Not supported in QSIG / PSS1
SUB	Subaddressing	Subaddress	Not supported in QSIG / PSS1

Other services

Dialling by name

A PISN subscriber can dial any PISN subscribers by name, irrespective of region, topology and protocol, if the PISN subscriber is explicitly listed by name in the numbering plan of the source PINX.

Least Cost Routing (LCR)

The Least Cost Routing function (LCR) is used for special network functions such as break-out or overflow (see "LCR Function", page 306).

Overflow Routing

When a connection is setup the system checks the availability of the selected path. If that path is not available due to overloading or due to a defect, the connection is set up via an alternative path determined by the configuration (see "Testing overflow routing in the PISN", page 343).

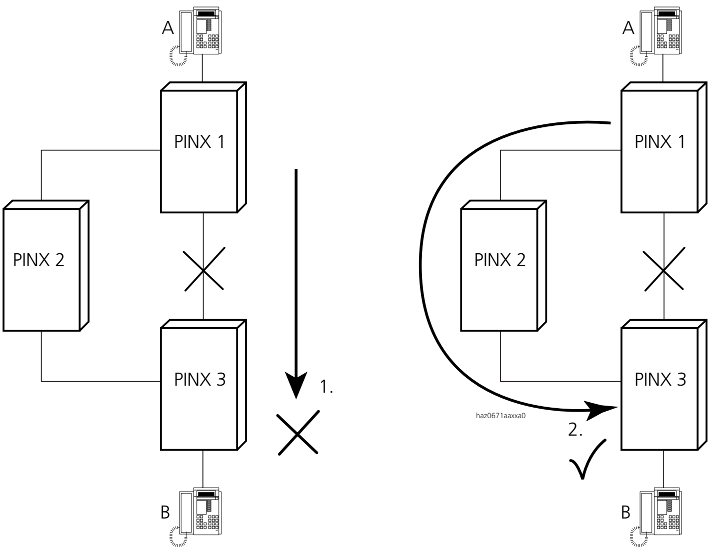


Fig. 1.36: Overflow via a dedicated line

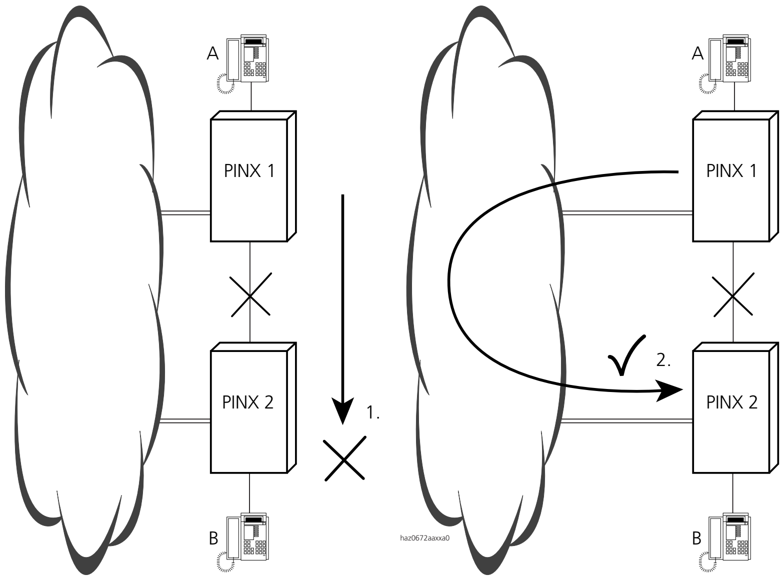


Fig. 1.37: Overflow via the public network -- the LCR function is used for this purpose

4.7.2 Break-Out

An outgoing external connection is routed into the public ISDN only at the PINX that is closest to the call destination. As the path in the public network is shorter, call charges can be saved in this way.
For more on this subject, see "Break-Out", page 348.

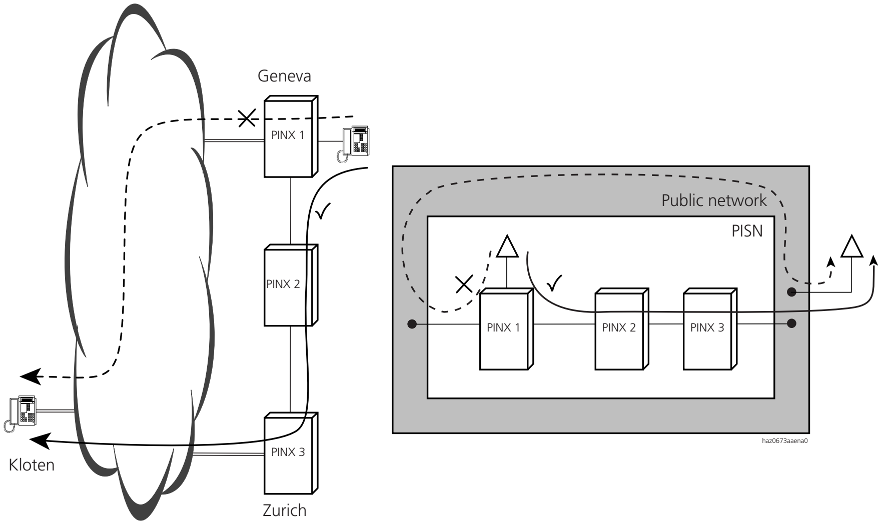


Fig. 1.38: Break-out: the shortest connection to a subscriber in the public ISDN

4.7.3 Break-In (Direct Dialling in for Virtual PISN Subscribers)

An incoming connection from the outside is routed into the PISN at the PINX closest to the calling party. For this the calling party must know the correct dial-in number for the PISN. This number can be displayed to him as a CLIP number, for example when calling from the PISN. Typical break-in applications:

- A company with several different locations wishes to present itself to the outside through one location only.
- Traffic from a virtually networked system to other systems in the leased-line network is always to be routed via the nearest system in the private network.

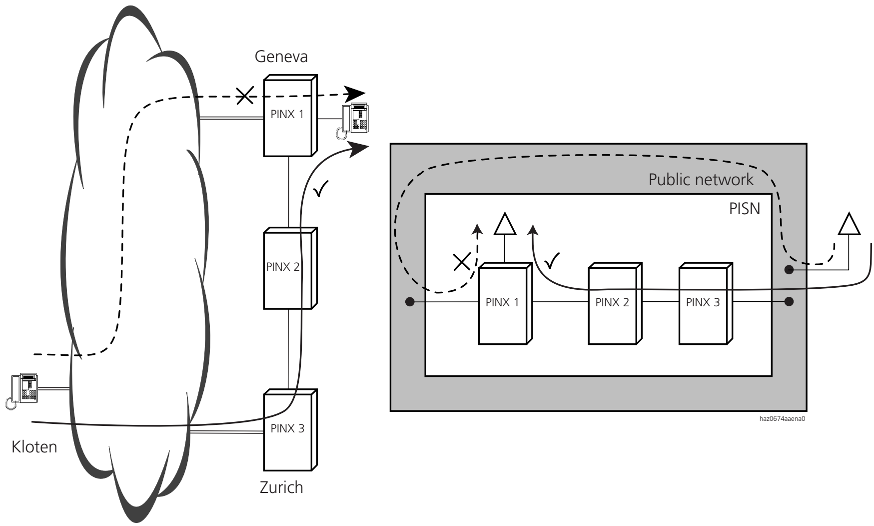


Fig. 1.39: Break-in: shortest connection from the public network to the PISN

5 Terminals

With terminals a distinction is made between terminals of the Office family (system terminals) and other terminals. At a secondary level another distinction is made between the type of terminal: corded or cordless. Added to this are the Pocket Adapter and the PC Operator.

5.1 System Terminals of the Office Family

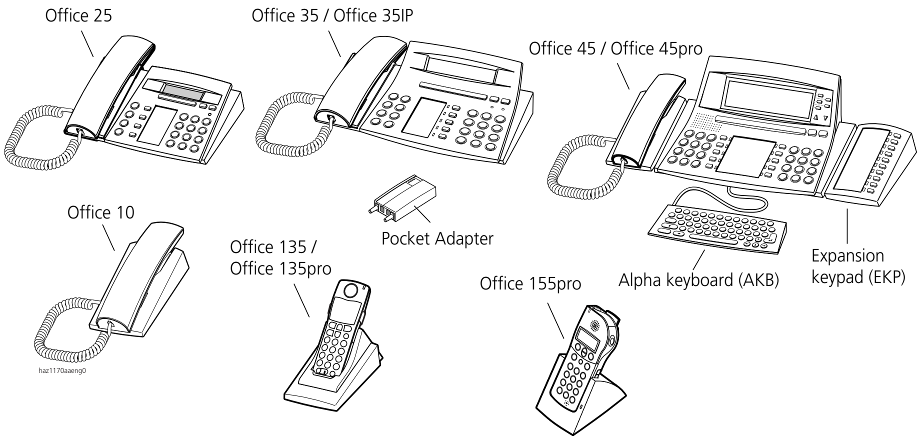


Fig. 1.40: The current generation of the Office family. (Office 20, 30, 40, 100, 150EEx, 130 and 130pro terminals will continue to be supported)

System terminals (including DECT handsets) are digital system terminals. Alphanumeric displays, menu prompting and an intelligent Foxkey ensure a user-friendly, intuitive handling of the complete spectrum of performance. Letters are assigned to the digit keys of the dial pad in accordance with the Vanity table and are keyed in via the dial pad. Freely configurable keys such as the hotkey are used to store frequently used call numbers, functions and operating procedures in the form of macros. A simple keystroke is enough to dial a call number or to listen to the Voice Mail System.

Foxkey

All the system terminals have a Foxkey, i.e. a variable function key that intelligently adapts to provide the right functions for each situation so that all the terminals can be operated intuitively. This means that depending on the system terminal's status you can easily access important commands with a simple keystroke. Scrolling at length through menus becomes superfluous.

Configurable keys

Freely configurable keys can be defined as number keys, function keys or team keys. Team keys provide a simple means, for example, of answering calls intended for absent colleagues on your own system terminal.

Expansion Keypad and Alpha Keyboard

Office 35/35IP and Office 45/45pro can be equipped with expansion keypads (EKP) and / or an alpha keyboard (AKB). Each expansion keypad provides 10 additional configurable keys. The alpha keyboard provides a convenient way of keying names or messages, with shortcuts available for certain features.

Convenient Dialling Options

Besides number dialling, Office system terminals also offer many other ways of setting up a connection. With dialling by name, for example, you can simply enter a person’s name (or initials), and the system terminal searches for the matching telephone number and dials it per keystroke. The only requirement is that the system knows the name in question.

Another convenient way of dialling call numbers is to dial from lists. A simple keystroke under the appropriate entry dials the number. The following lists are available with system terminals:

- Last-number redial list: The list of subscribers last called from that particular system terminal
- Unanswered call list: The list of subscribers who have tried to reach the system terminal
- Answered call list: The list of subscribers whose calls have been answered

Data Integrity

All the terminal data is stored in a non-volatile memory on the system and can be saved using a backup with AIMS.

AD2 interface

Two system terminals or a system terminal and a Pocket Adapter can be connected in parallel to the two-wire AD2 digital user-network interface. One DECT radio unit can be connected for each AD2 interface.

Ethernet interface

The Office 35IP and Office 1600IP terminals are connected directly to an Ethernet interface on the PBX, making the IP infrastructure available for telephony uses. The terminal Office 35IP has an integrated switch (100BaseT) for connecting the workstation computer. This means it is not necessary to install a separate network connection point to operate the Office 35IP.

Applications

Some system terminals are ideally suited to be used under special conditions. Tab. 1.9 provides an overview of such applications.

Tab. 1.9: Area of application for system terminals

Area of application	Office 1600IP	Office 45/45pro	Office 35IP	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office 135/135pro	Office 100	Office 150	Office 155pro
System terminal on AD2	–	✓	–	✓	✓	✓	✓	✓	✓	–	–	–	–
System terminal on the IP network	✓	–	✓	–	–	–	–	–	–	–	–	–	–
Handset (cordless)	–	–	–	–	–	–	–	–	–	✓	✓	✓	✓
Wall mounting possible	–	–	–	–	–	–	–	–	✓	–	–	–	–
Key telephone	–	✓	✓	✓	–	✓	✓	–	–	–	–	–	–
Operator Console	–	✓	–	–	–	–	–	–	–	–	–	–	–
PBX settings possible	–	✓	–	–	–	–	–	–	–	–	–	–	–
Splashwater resistant (IP54)	–	–	–	–	–	–	–	–	–	–	–	✓	✓
Explosion-protected type (EEx model only)	–	–	–	–	–	–	–	–	–	–	–	✓	–

5.2 Corded Terminals

Overview

Tab. 1.10 contains a summary of selected features of corded system terminals, which are described in more detail below.

Tab. 1.10: Corded system terminals

	Office 45 / Office 45pro	Office 35IP	Office 35	Office 25	Office 10
Interface	AD2	LAN	AD2	AD2	AD2
Function keys	12	8	8	4	3
Freely configurable keys	10	5	5	4	3
Foxkey	✓	✓	✓	✓	✓
Menu / Information keys	✓	✓	✓	✓	–
Display (lines x characters)	8 x 40	2 x 24	2 x 24	1 x 14	–
Dialling by name	✓	✓	✓	✓	–
Private phone book entries	350	40	40	10	10
Entries, unanswered call list	10	10	10	4	–
Entries, answered call list	10	10	10	4	–
Loudspeaker	✓	✓	✓	✓	–
Hands-free mode	✓	✓	✓	–	–
Alpha keyboard AKB (optional)	✓	✓	✓	–	–
Expansion keypad EKP (optional)	✓	✓	✓	–	–
Pocket Adapter (optional)	✓	✓	✓	✓	✓
Use as Operator Console	✓	–	–	–	–

5.2.1 Office 45 / Office 45pro

The Office 45 is the top model in the Office family. It offers the complete performance spectrum available with the Office family. All the features can be used intuitively through menu prompting and Foxkey. The large, clearly structured display and the display keys make operation all the easier and offer full operator functionality. On the Office 45pro the display is illuminated. The operator menus are available in many European languages. The Office 45 can be used as a featurephone, key telephone or Operator Console. The System Assistant function can also be used to define settings in the system that occur frequently.

Characteristics of Office 45/45pro:

- Can be used as a featurephone, key telephone and Operator Console
- Allows special settings to be made on the system
- Intelligent Foxkey
- 10 freely configurable keys with dual-coloured LEDs
- Message LED
- Graphics-compatible display with 8 lines of 40 characters (backlit on the Office 45pro)
- Dialling by name and by list
- Operator menus in many languages
- Loudspeaker and microphone for hands-free mode; possibility of full-duplex hands-free (hardware required)
- Headset (optional)
- Replaceable keypad cover (optional)
- Socket for expansion keypads with additional configurable keys (optional)
- Alpha keyboard for simplified character input and command shortcuts (optional)
- Absence key, e. g. for call forwarding to an ACD server

Office 45/45pro is recommended for the following applications:

- Premises with a very high calling rate
- Management
- Secretary
- Attendant
- System Assistant (PBX configuration)
- Telemarketing
- Telesales

5.2.2 Office 35

The Office 35 already offers the complete performance spectrum of the Office family. All the features can be used intuitively through menu prompting and Foxkey. The operator menus are available in many European languages. The Office 35 is the ideal terminal for users with higher standards.

Characteristics of the Office 35:

- Intelligent Foxkey
- 5 freely configurable keys with LEDs
- Message LED
- Graphics-compatible display with 2 lines of 24 characters
- Dialling by name and by list
- Operator menus in many languages
- Loudspeaker and microphone for hands-free mode
- Headset mode (optional)
- Can be used as a key telephone
- Interchangeable keyboard covers in different designs (optional)
- Socket for expansion keypads with additional configurable keys (optional)
- Alpha keyboard for simplified character input and command shortcuts (optional)
- Absence key, e. g. for call forwarding to an ACD server

Recommended for the following areas of application:

- Premises with a high calling rate
- Frequent phone users
- Distribution staff
- Sales staff

5.2.3 Office 35IP

The Office 35IP system terminal is an IP terminal with the same range of features and the same user prompting as an Office 35. The Office 35IP is connected to the PBX via LAN instead of via AD2. An additional LAN socket on the Office 35IP can be used to connect a workstation computer.

The Office 35IP is supplied with a plug-in power supply unit. This is required only if the terminal is not powered via the LAN (Power over IP).

Recommended for the following areas of application:

- Networked, remote workstations
- Smaller branch offices

See also "AIP 6350 and Office 35IP", page 122

5.2.4 Office 25

The Office 25 has a Foxkey and four configurable keys as well as an alphanumerical display for displaying the number and name of the caller (CLIP / CNIP), operator menus in many European languages, Foxkey assignment and more. With */# procedures you can take advantage of nearly the entire performance range of the system. With the Office 25 you can send and receive messages to and from equipment with an alphanumeric display.

Characteristics of the Office 25:

- Intelligent Foxkey
- 4 configurable keys
- Message LED
- Alphanumeric display with 14 characters
- Dialling by name and by list
- Operator menus in many languages
- Loudspeaker
- Minimum space requirements

Recommended for the following areas of application:

- Premises with a normal calling rate
- Office workers
- Administration
- Office

5.2.5 Office 10

The Office 10 is a cost-effective alternative to analogue terminals. It features the Office family's intuitive operator prompting: Pressing the Foxkey intelligently selects the function best suited to the situation. The terminal itself is small, compact and can also be wall-mounted.

Characteristics of the Office 10:

- Digital telephony at an affordable price
- Intelligent Foxkey
- 3 configurable keys
- Message LED
- Minimum space requirements
- Wall mounting (optional)
- The ringing can be set to very loud

Recommended for the following areas of application:

- Premises with a low calling rate
- Hospitals
- Hotel rooms
- Noisy working environments
- Workshops

5.2.6 Office 20, Office 30 and Office 40

The system terminals Office 20, Office 30 and Office 40 are supported as before. With only a few exceptions all the features of the Ascotel IntelliGate system are supported.

The table in "Features Overview", page 563 shows the complete spectrum of features and allows a comparison of the system terminals.

5.2.7 Other digital and Analogue Terminals

On the analogue and S user-network interfaces, appropriate terminal types from Aastra other manufacturers can be used. ISDN terminals must comply with the

Euro ISDN standard (ETSI). For these terminals on the S bus Ascotel IntelliGate also offers a number of ISDN features.

All terminals approved by the network operator can be used on the analogue user-network interfaces. The system supports pulse and frequency dialling modes. Ascotel IntelliGate features can only be operated with */# procedures.

The table in "Features Overview", page 563 shows the complete spectrum of features and allows a comparison with the system terminals.

5.3 Cordless Terminals

Overview

Cordless DECT terminals of the Office family have virtually the same features as the corded terminals. Tab. 1.11 contains a summary of selected features of cordless system terminals, which are described in more detail below.

Tab. 1.11: Cordless Office Terminals

	Office 135	Office 135pro	Office 155pro
Standby time / Talk time	120 Hours / 12 hours	120 Hours / 12 hours	100 Hours / 10 hours
Display	39 x 90 dots	39 x 90 dots	40 x 121 dots
Housing colour	anthracite	anthracite / titanium-silver	anthracite / grey
Backlighting (display)	✔	✔	✔
Backlighting (keypad)	–	✔	–
Menu / Information key	✔	✔	✔
Foxkey	✔	✔	✔
Hotkey / functions	1 / 6	1 / 6	1 / 6
Redkey function (Hotkey alarm trigger)	✔	✔	✔
Loudspeaker key	✔	✔	✔
Dialling by name	✔	✔	✔
Private phone book	40	40	40
Loudspeaker	✔	✔	✔
Hands-free mode	✔	✔	✔
Volume control	✔	✔	✔
Vibra call	–	✔	✔
Headset socket	–	✔	✔
Charging bay	✔	✔	✔

	Office 135	Office 135pro	Office 155pro
Socket for headset charger	–	✔	–
Leather case	✔	✔	✔
Carry clip	✔	✔	✔
Splashwater-proof	–	–	✔
GAP compatible	✔	✔	✔

Common DECT handset features

A handset is not allocated to any particular radio unit. It can set up and clear down incoming and outgoing calls in all radio units.

Even during a call, the subscriber is able to move around freely with the handset within the coverage area.

A handset can be logged on simultaneously to a maximum of 4 different Ascotel DECT systems.

The PBX recognizes only handsets that are logged on. This prevents unauthorized users from making use of the system.

An Ascotel DECT subscriber can be logged on to a system temporarily as a visitor.

Ascotel DECT subscribers can be integrated into user groups like other subscribers.

Problem-free operation is no longer guaranteed at the limit of and outside the radio area.

5.3.1 Office 135 / Office 135pro

The Office 135/135pro is ideal for office environments as well as for hospitals and retirement homes. As it is compact, light and handy, it can be taken along to any meeting. It is used mainly by subscribers who are on the move yet need to be reachable.

Its operator convenience is similar to that of an Office 35. It also features a number of functions that are very useful for mobile subscribers in particular, for example discreet ringing, key lock and a hotkey that can be used to store 6 numbers or functions. This system terminal's hands-free mode is of a very high quality.

Vibra call (Office 135pro) and a clearly visible LED round off the system terminal's range of equipment. Macro commands can also be used.

The Office 135/135pro can be updated at any time with the current operating software via the cellular network.

Features Office 135/135pro:

- Small, practical, lightweight system terminal
- Easy to operate with one hand
- Sturdy, high-quality keypad
- Intelligent Foxkey
- Hotkey (2 configurable operating modes)
- Message LED
- Graphics-compatible, illuminated display
- Dialling by name and by list
- Hands-free mode with volume control
- Can operate in combination with a corded system terminal (Twin Mode / Twin Comfort)
- Easy and convenient to use, even when in the charging bay
- Software update with radio network
- Leather pouch (optional)

Additional features of the Office 135pro:

- Vibra call
- Socket for optional headset
- Socket for optional plug-in power supply for direct charging
- Illuminated keypad

Recommended for the following areas of application:

- Mobile use outside and inside buildings (office, retirement homes, hospitals)
- In connection with a corded system terminal (Twin Mode / Twin Comfort)

5.3.2 Office 155pro

The splashwater proof and shock-proof Office 155pro is particularly well suited for the workshop and heavy industry sector.

Characteristics of the Office 155pro:

- Sturdy design
- Splashwater resistance (IP54)
- Vibra call
- Headset socket in the battery (separate type)
- Intelligent Foxkey
- Hotkey (2 configurable operating modes)
- Message LED
- Graphics-compatible, illuminated display
- Dialling by name and by list
- Hands-free mode with volume control
- SIM card (contains registration data and personal settings)
- Can operate in combination with a corded system terminal (Twin Mode / Twin Comfort)
- Can also be operated from the charging bay
- Leather pouch (optional)
- Headset (optional)

Recommended for the following areas of application:

- Mobile use outside and inside buildings (workshops, building sites, heavy industry)
- In connection with a corded system terminal (Twin Mode / Twin Comfort)

5.3.3 Office 100, Office 130/130pro, Office 150/150EEx

These system terminals are supported as before

5.3.4 9d handsets

The rugged 9d handsets from the Ascom Wireless Solutions product portfolio can also be registered as system terminals on Ascotel IntelliGate under the GAP standard or with the "Advanced Messaging" licence. User-friendly messaging and alarm systems can be implemented in combination with the IMS (Integrated Message Server) (see "Expanded messaging system with 9d handsets", page 143).

The following overview shows the features available on the handsets, with and without licence:

Tab. 1.12: Features on 9d handsets in GAP mode (without licence)

Features	Remarks
Local handset features	Local phone book, dialling by name, abbreviated dialling, configuration settings, etc.
Display caller's number to the called party (CLIP)	The name is displayed if a name is stored along with the phone number in the local phone book. The calls also appear in the call lists
Subscriber and system-specific configurations and activating/deactivating functions using */# procedures.[1)]	e. g. Call Forwarding Unconditional, telephone barring, Follow me, Do not disturb, remote control, control relays, operate switch group, etc.
Suffix dialling functions that can be activated with the Flash key during a call or while a connection is being set up.[1)]	e. g. enquiry call, brokering, call transfer, conference, call waiting, intrusion, parking, callback, etc.
Handover	Automatic call handover between radio cells

[1)] Same operation as on analogue terminals

Tab. 1.13: Additional features on 9d handsets with licence

Features	Remarks
Display caller's name to the called party (CNIP)	The name is provided by the PBX and so does not have to be stored in the local phone book
Display the number / name to the calling party (COLP / CONP)	The number / name is provided by the PBX
Display call charges	Where available, the call charges are shown on the display at the end of an outgoing call.
Use of Voice Mail functions	Notification on the display of Voice Mail messages received
Date/time synchronisation	Automatic date and time synchronisation after every registration with a system
Group calls	DECT group calls are possible with ordinary user groups. Up to 16 handsets are possible in one user group.

For more information on Ascom Wireless Solutions products see http://www.ascom.com/ws

5.3.5 Cordless Terminals by Other Manufacturers

Ascotel DECT supports the features of basic telephony defined as "compulsory" in the Generic Access Profile (GAP) standard (EN 300444, issue 1.2.2 of 19.8.1997).

These restricted functions can only be fully utilized if they are implemented in accordance with the GAP standard in both the handsets and the system supplied by the outside manufacturer.

Note:
Restrictions are likely also with regard to the quality of the radio links since mobility management with handover/roaming cannot be influenced where non-system handsets are concerned. In other words, the quality of these functions depends to a large extent on the software of non-system handsets.

5.4 DECT radio units

One or more radio units are required to be able to operate DECT handsets on the Ascotel system. They are connected to one or two AD2 interfaces in the same way as corded Office terminals. They are powered either via the AD2 or locally using separate plug-in power supply units. Office handsets as well as GAP-mode DECT handsets by outside manufacturers can be logged on to a radio unit.

Fig. 1.41: DECT radio unit and Office handsets

There are three types of radio units, and although they differ very little in appearance they do offer different features.

Tab. 1.14: Ascotel DECT radio units

Features	SB-4	SB-8	SB-8ANT
Number of simultaneous call connections	4	8[1)]	8[1)]
Number of AD2 interfaces required	1	1[2)] or 2	1[2)] or 2
Status display with LED	✔	✔	✔
Disconnectable LED display (normal mode)	✔	✔	✔
Automatic upload of new system software	✔	✔	✔
Plug-in power supply unit for local power supply [3)]	✔	✔	✔
Power requirements from PBX with local power supply to RU	620 mA	< 10 mA	< 10 mA
Max. length of AD2 bus[4)] for operation without local plug-in power supply unit	660 m	1200 m	1200 m
Alarm following the failure of a local plug-in power supply unit	–	✔	✔
External antenna connections	–	–	2
GAP compatible	✔	✔	✔

1) Only in Zero-Blind-Slot mode, which must also be supported by the handsets (Office 135/135pro, Office 155pro, includes any GAP terminals)

2) Only 4 call connections are possible simultaneously in the event of a connection via 1 AD2 interface

3) SB-4 and SB-8 / SB-8ANT require different plug-in power supply units

4) For wire diameter 0.5 mm

A SB-8 radio unit can also be operated on < I6.1 systems, in which case it acts like an SB-4 (see also "DECT Compatibility", page 970).

For additional information on planning and configuring a DECT system, see "Planning DECT systems", page 611.

5.5 Office 1600IP

The PC-based system terminal Office 1600IP is an OIP client application. (see "Open Interfaces Platform", page 136).

With its user-friendly interface it extends the boundaries of the Office system terminals, provides powerful group functions and integrates outstandingly well into standard PC programs.

Although designed primarily with small to medium-sized workgroups in mind, the Office 1600IP is also an invaluable companion for individual users with high mobility requirements.

With the central data management system Office 1600IP users have direct access to call lists, phone books, messages, etc., from their home, office and anywhere with a link-up to the company LAN.

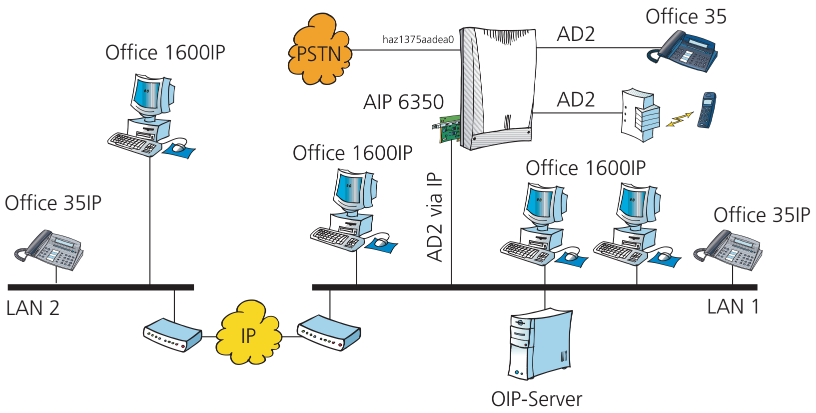


Fig. 42: System with Office 1600IP terminals and OIP server

Planning

From the PBX's viewpoint the Office 1600IP is as much an IP system terminal as the Office 35IP:

- Voice transmission is effected via the AIP 6350 card
- All the voice connections go via the PBX
- The subscriber data is stored in the PBX

But unlike the Office 35IP the OIP server signals and controls all the Office Office 1600IP's in the network. IP addressing is also effected via the OIP server and does not have to be done manually.

PC Requirements

An Office 1600IP can be operated on PCs with the following system requirements:

- Windows XP, Windows 2000, Windows NT, Windows ME, Windows 98
- 500 MHz CPU (recommended)

- At least 50 MB hard-disk space available
- 256 MB RAM (recommended)
- Ethernet connection

The PC has to be equipped with a headset or handset. Devices with USB and analogue interfaces are available. Please contact your specialist retailer for more information.

PBX Requirements

Voice transmission is effected via the AIP-6350 card. Up to 32 Office 1600IP can be operated on each card.

- The system and expansion limits can be found in the AIP 6350 / Office 35IP System Manual
- Each Office 1600IP is registered in the PBX as a subscriber.
- Telephony settings are made in the PBX
- A separate licence has to be obtained for each Office 1600IP.

IP-network Requirements

To achieve a high quality of speech, it is important to dimension and plan the IP network carefully, with the same meticulousness as when planning systems with the IP system terminals Office 35IP. All the relevant information can be found in the AIP 6350 / Office 35IP System Manual.

5.6 Pocket Adapter

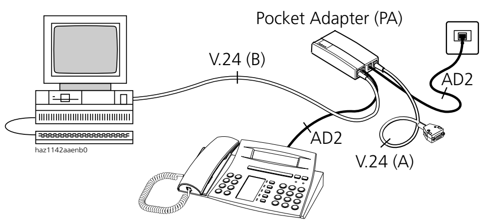


Fig. 1.43: Connecting a terminal and PC to the Pocket Adapter

The Pocket Adapter (PA)[1] is used to connect a PC to an AD2 user-network interface. At the same time an Office terminal can be connected to the same user-network interface. The Office terminal and the PA must have different terminal selection digits (TSD). TSD 1 is factory set on Office terminals, and TSD 2 on the PA. The PA does not have an internal subscriber number.

The PC is connected via a serial interface on the V.24 cable of the Pocket Adapter. Software drivers have to be installed on the PC to run the various applications.

Possible applications include:

- First-party CTI (see "Ascotel CTI", page 133)
- AIMS (see "Ascotel Information Management System (AIMS)", page 144)
- Connection of an external alarm server

5.7 PC Operator Console Office 1550

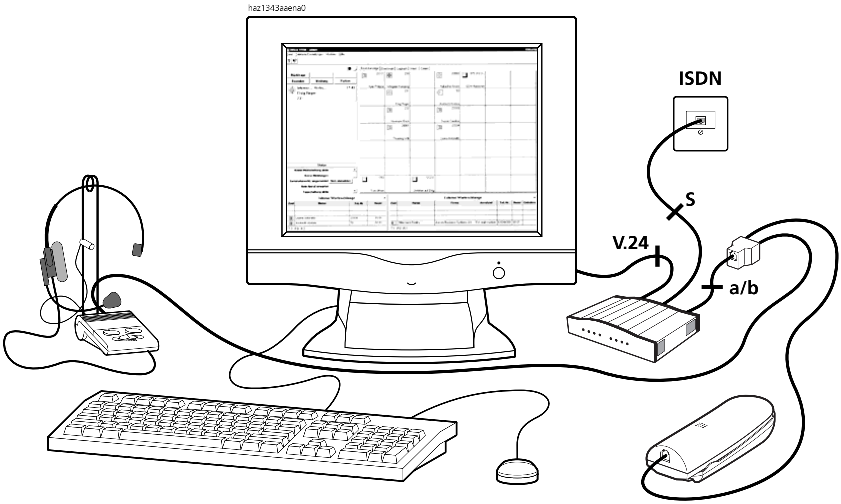


Fig. 1.44: PC Operator Console workstation

[1] PA Version ≥ V2.4

The PC Operator is a modern CTI application that allows Operator and Attendant functions to be carried out conveniently with a mouse click or keystroke on the PC.

By integrating operator and information centre functions the PC Operator Console assists telephone attendants in dealing with their many different tasks:

- Switching calls
- Maintaining a clear overview of telephone traffic
- Answering enquiries
- Administering calls

Other characteristics:

- Graphic interface
- Mouse and keyboard operation
- Foxkeys
- Busy lamp field, phone book and logbook (modules)
- Internal and external queues
- Feature Wizard (for activating Ascotel IntelliGate features)
- Background and foreground mode
- Database concept with import / export and DDE interface to other databases
- Call routing in event of a failure of the PC Operator Console
- Possibility of running several PC Operator Consoles on one PBX

Recommended for the following areas of application:

- Operator Console
- Information centre
- Frequent phone users
- Department secretary

6 AIP 6350 and Office 35IP

The joint use of the two IP components, namely the IP interface card AIP-6350 and the system terminal Office 35IP, expands the Ascotel IntelliGate platform to the IP data network, and makes the IP infrastructure available for telephony, too.

The system terminal Office 35IP is a fully fledged IP terminal with the complete range of features of an Office 35. It can be operated anywhere in the IP data network as long as the connection to the PBX complies with the quality criteria required for VoIP (Voice Over IP).

Like any Office system terminal, Office 35IP communicates with the PBX via the AD2 protocol. The features and user prompting are identical to those of the Office 35.

The AIP 6350 is the PBX's Ethernet interface to the Office 35IP system terminals. The expansion card from the AIP family has been designed specially for this application.

Both the Office 35IP and the AIP 6350 are configured and updated using the Ascotel AIMS management software. All the settings can be made both offline and online with the usual operating interface.

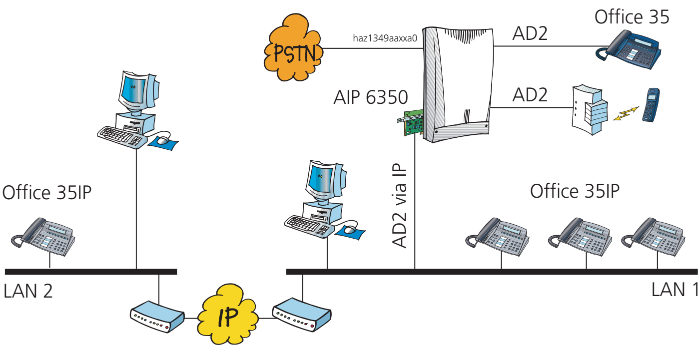


Fig. 1.45: Office 35IP and AIP 6350 expand the Ascotel IntelliGate platform to the IP network

This key expansion option translates into a whole range of advantages for the user:

- Networked, remote workstations can be integrated at low cost into the internal telephone system, without compromising the ease with which phone calls are made. Unlike a connection via the public telephone network, no call charges are incurred, and users can be reached as internal subscribers.
- Many features that are restricted when a remote subscriber is integrated as a virtual subscriber can be fully utilised, i.e. Call Forwarding Unconditional, Voice Mail, Courtesy, Text Messages, Announcements.
- In the case of smaller branch offices the customer can dispense with using an additional PBX in the branch.
- When expanding an existing infrastructure with new connections for PC and phones there is no need to expand the phone lines.

Note:
As a result of the expansion of the Ascotel IntelliGate platform to the IP data network the network used becomes part of the Ascotel IntelliGate system. The communication quality therefore depends directly on the network quality (in the same way as the road network influences the delivery quality of a haulage company).

More detailed documentation:
The following documents are available for AIP 6350 and Office 35IP:
– AIP 6350 / Office 35IP System Manual
– User's guide Office 35IP
– Office 35IP Quick User's Guide

7 Ascotel IP Gateway AIP 6400

By using the expansion card AIP 6400 the customer has the possibility of capitalizing on the resources of his existing intranet (LAN / WAN) to implement a cost-saving extension of his internal phone network.

The AIP 6400 expansion card is used for networking the PINX via the LAN / WAN (see AIP 6400 System Manual).

Data services such as remote maintenance, fax group 3 / group 4 and modem operation are not possible with the Ethernet interface on the expansion card.

Configuring the IP Gateway AIP 6400

The IP Gateway AIP 6400 can be configured from any PC with network access. The only requirement is that the IP address for the Ethernet interface is known on the expansion card. The main page of the configuration appears on the PC once the IP address has been entered in a conventional commercial web browser and after authentication by means of user name and password.

Technical data

The expansion card AIP 6400 is fitted into an expansion slot.

Tab. 1.15: IP-Gateway AIP 6400 technical data

Number of VoIP channels per Expansion card AIP-6400[1)]	max. 12
Ascotel systems networkable via IP	max. 50
LAN connection	Ethernet 10Base-T / 100Base-T with autodetect
Voice data conversion	ISDN-PCM (PBX) ↔ H.323 (LAN) with G.711 uncompressed or G.723 compressed
Voice compression	Approx. 3.5:1 (with G.723)
Signalling conversion	Q.931 (PBX) ↔ H.323 (LAN)
Number of expansion cards of the type AIP-6400 per PBX	2025: Max. 1 2045: Max. 2 2065: Max. 4
Quality of Service	IEEE 802.1 p/Q

1) 1 primary rate access is taken up on each Gateway interface. This primary rate access is therefore no longer available as a network interface.

8 Ascotel Voice Mail System AVS 5150

Availability is the be-all and end-all of telecommunications. Voice Mail Systems are a cost-effective means of achieving that availability. Callers who have to make several call attempts because a particular subscriber is unavailable are now a thing of the past. Voice Mail Systems make for far more efficient telephone communications between staff members, customers and suppliers, which in turn means far better operating results.

Even today, the most common communications aid, the answering machine, is still an essential piece of equipment in the small office and household office sectors (SOHO). In small and medium-sized companies, however, it quickly reaches the limits of its performance capabilities. A second or third answering machine is certainly not an ideal solution. By this stage at the latest, it is high time to consider expanding the PBX with a Voice Mail System that provides every employee with personal answering machine functions. That way you remain competitive and never lose any customers.

The Ascotel Voice Mail System AVS has been specially developed with small and medium-sized companies in mind. It is the further development of the telephone answering machine, providing a centralized answering system for the company as a whole. In addition to traditional answering functions Voice Mail Systems offer callers and operators a wide range of new possibilities such as voice and time-controlled recorded messages, retrievable additional information and call routing based on the evaluation of caller number (CLIP), etc. Tab. 1.16 shows all the features of the AVS at a glance. Further information can be found in the documentation on the Voice Mail System.

Thanks to the networking capability of Ascotel systems mailboxes can also be made available to subscribers of other PBX systems.

8.1 Features of the Voice Mail System AVS 5150

Tab. 1.16: Features of the Ascotel Voice Mail System AVS

Features	Ascotel Voice Mail System AVS
Number of Voice Mail channels	2 and 4 [1)]
System configuration	✓ [2)]
local with PC	V.24, Ethernet, ISDN or AD2(PA)
remote with PC	ISDN
local or remote with telephone	with DTMF
Ascotel integration	internal bus

Features	Ascotel Voice Mail System AVS
Signalling on system terminals	✓ [3]
Multilingual capability Number of languages activated simultaneously	✓ 3
Auto-Attendant Tree structure levels	✓ 18
Routing table for expanded routing (DDI, CLIP) Number of entries	✓ 20
Voice mail Number of mailboxes Number of messages per mailbox Recording capacity	✓ 128 99 4hrs and. 8hrs [1]
Mailbox groups, distribution lists Number of distribution lists per system	✓ 4
External notification (outcalling) Number of numbers per mailbox	✓ 1
Audio text Number of preconfigured audio texts Total number of possible audio texts	✓ 18 39
Additional information on messages received	Time and sender (CLIP) of call
Backup and Restore of voice and configuration data	✓
Fax tone recognition	✓
Time synchronization with PBX	✓

1) Voice Mail cards VM-02P and VM-04P

2) See also Fig. 1.46

3) The signalling can be set individually for each mailbox.

8.2 Connection Concept and Configuration Possibilities

Two different expansion cards can be used on Ascotel IntelliGate 2025, 2045 and 2065 systems: VM-02P and VM-04P. They differ in recording capacity and in the number of voice channels (see Tab. 1.16). Communications with the PBX are effected via the internal bus.

The configuration of the Ascotel Voice Mail System AVS is effected via AIMS using the same interfaces as the PBX configuration. Fig. 1.46 illustrates the possibilities for internal and external configuration. The configuration program for the Voice Mail System is called Voice Mail Manager (VMM) and is an autonomous application that can be started via a menu item in AIMS.

In addition to configuration with the Voice Mail Manager the AVS can also be configured from an internal or external connection using a phone with DTMF dial signals.

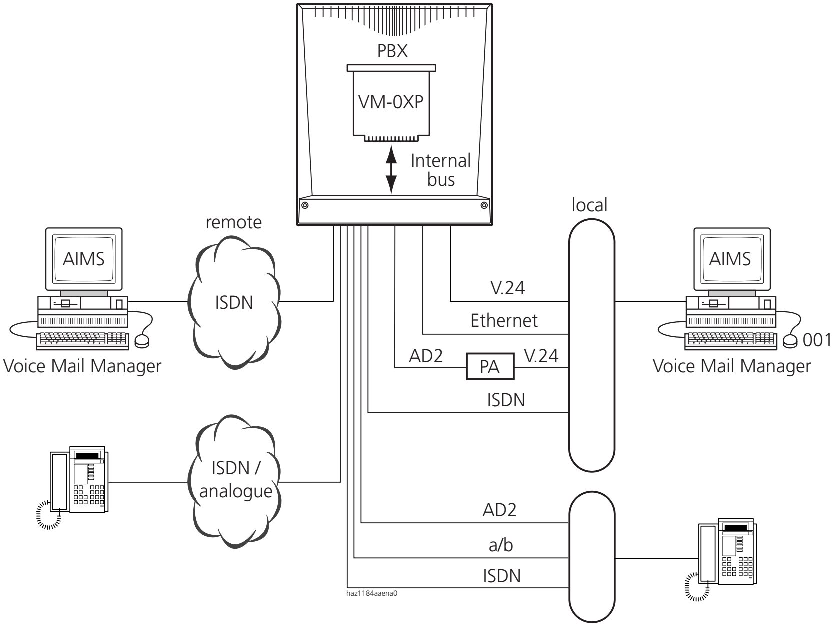


Fig. 1.46: Connection concept and configuration possibilities

9 Supplementary Equipment and Additional Applications

Supplementary equipment and additional applications extend the performance range of the system.

9.1 teleCOURIER 900 Paging System

The digital paging system is theoretically capable of controlling up to 200 (System 2065) pagers. The following pager types are available:

- Pager types without display
- Pager types with small display
- Pager types with large display

teleCOURIER 900 complies with the requirements of the "Functional Standard for the Connection of Radio Paging Equipment (RPE) to a Private Telecommunications Network (PTN)" (ESPA / ECMA).

Functions and Features

The description of the functions and features of the paging system deals with the basic functions, the interplay with the system and the special characteristics of the teleCOURIER 900.

Paging a Subscriber

The search can be activated specifically from any internal terminal. In the following cases external calls activate the paging:

- The destination is a pager number.
- The destination is diverted to the pager (CFU).
- The destination is forwarded to the pager (CFNR).
- The destination is forwarded to the pager (CD).

The caller obtains the normal ring-back tone. If the call is parked, the caller obtains the ring-back tone of the paging system.

Pager Discreet Ringing

As each pager has its own subscriber number, it can be called discreetly without having to activate the paging function beforehand.

Internal/External Ringing Distinction

On the teleCOURIER 900 system different ringing signals can be set for internal and external calls so the called party can identify the origin of the call.

Answering a Paging

The call is parked until the PS search time has expired and can be answered from any free internal terminal in the system. The paged subscriber identifies himself on the terminal by entering his terminal's or pager's subscriber number.

Office terminals support the function "Answer search" in the menu.

Paged Subscriber Does Not Answer

A ringing time is configured in the paging system which specifies the amount of time for which the pager rings.

The PS search time begins as soon as a call is parked. It specifies how much time the paged subscriber has to answer the ringing. The search time is configured in the system, within a range of 10 and 360 seconds.

If the paged subscriber does not answer before the PS search time expires and if the caller has not hung up, the paging system offers the following options:

- Triggering an internal call
- Routing an external call to user group 16 if the Operator Console or general bell are configured accordingly.
- A transferred call goes back in the form of a PS recall to the subscriber who transferred the call. The recall time can be configured between 10 and 240 seconds.

Call Forwarding Unconditional and Call Forwarding on No Reply to the Pager

Subscribers can divert calls to their personal pager, a non-personal pager or a third-party pager.

Personal and non-personal pagers are allocated to subscribers during configuration (for the procedures for the various forwarding types, see the descriptions in the Chapter "Call Forwarding Unconditional functions", page 438). Office terminals support the control of call forwarding unconditional in the menu.

Send Text Messages to a Pager

Pagers with a display can receive text messages. Incoming text messages are signalled both visually (LED) and acoustically.
If the pager is on the charging bay or switched to "Absent", the message is buffered in the paging system and then sent when the user is present.

The pager number is entered as the message destination.

Displaying CLIP / CNIP

If the display functions are available, the PBX will transmit the information to the pager.

Pagers and User Groups

Pagers cannot be entered in user groups. As soon as a user group subscriber diverts to a pager, he is switched out of the user group. The last subscriber in a user group cannot activate a call forwarding to a pager.

With call forwarding on no reply the subscriber remains in the user group.
(This applies to ordinary user groups, not to large user groups, see "User Group", page 236).

Pager Absent

Pagers are absent if they are on the charging bay or switched as "Absent" by the paging system.

On system terminals with display the caller can be shown that the pager is "Absent". The message is edited via the paging system. On terminals without display the call is diverted to a predefined destination.

Other Features

The following entries can also be made for pagers:

- Abbreviated dialling destination
- Emergency number
- Destination for Do not disturb
- Substitution destination
- Hotline

Connection

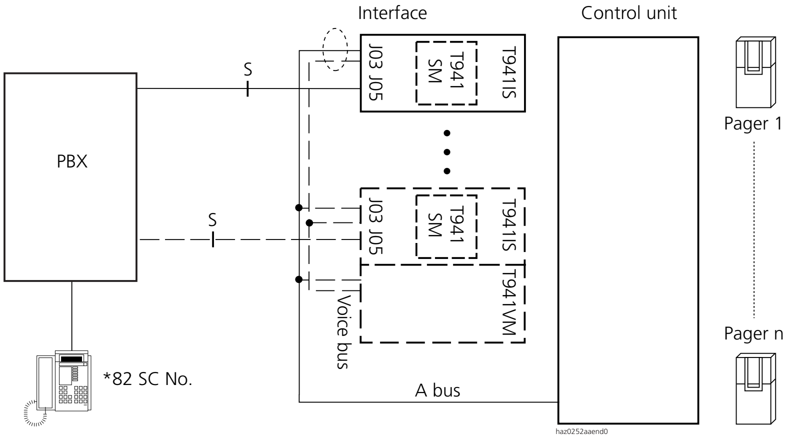


Fig. 1.47: teleCOURIER 900 connection (paging system with S interface)

The paging system is connected to the system via one or more S interfaces and operated in V2 mode. It communicates with the system via the S bus paging interface (PSI).

A D channel is used for signalling the pager. This means that several parallel paging operations are possible, with one S interface usually being sufficient.

User information channels (B) are used only for transmitting voice messages, provided the feature is supported in the first place. In this case more S interfaces are required accordingly. Transmitting voice texts presupposes that the following modules are fitted:

- for each paging system: 1 T941VM module
- for each S interface: the T941SM add-on module (in addition to the T941IS interface module)

Configuration

For each PSI module for S interfaces you need to enter the connected user-network interface. This allocation is made automatically by the system when the PSI module is connected and can be barred manually via the configuration.

Note:
With PSI modules for S interfaces the terminal selection digit (TSD) is permanently set on 8 and cannot be modified. No other terminal should be operated on the same S interface in parallel with a PSI module for S interfaces.
The maximum number of search paths depends on the number of S interface PSI modules since the search process is signalled via the D channel. This means that one S-interface PSI module is sufficient under normal circumstances.

Each pager is allocated a pager subscriber number – pager number for short – in the system's numbering plan; the pager number is entered during the subscriber configuration. The following allocations are possible:

- The same pager number for several subscribers
- The same pager number for several pagers (PSA feature)
- Several pager numbers on one pager (the maximum number is predefined by the PSA)

Internal subscribers without their own terminal can also be equipped with a pager.

9.2 Ascotel CTI

Computer Telephony Integration (CTI) designates the convergence of telephony and computer systems. CTI applications can be used to control telephony functions via the PBX's CTI interface and to monitor telephone states.

The TAPI driver provides the software link between the PBX and Microsoft TAPI. Ascotel CTI interfaces support applications to Microsoft TAPI 2.1 standard.

With CTI a distinction is made between first-party CTI (singler-user solution) and third-party CTI (multi-user solution).

9.2.1 First-Party CTI

A first-party CTI is the direct physical connection between a phone terminal and a telephony Client (workstation PC). Telephony functions and telephone states are controlled and monitored on the telephony Client.

A first-party CTI solution is ideal for a small number of CTI workstations and is easily implemented.

Connection to Ascotel IntelliGate

Ascotel IntelliGate only supports first-party CTI on corded Office terminals if the Pocket Adapter is used (Fig. 1.48). First-party CTI with Office 35IP terminals is not supported.

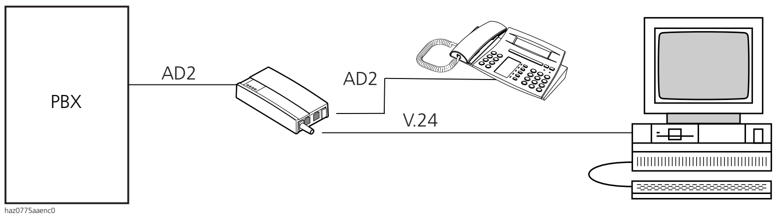


Fig. 1.48: First-party CTI on the AD2 interface

First-party CTI requires the Ascotel TAPI driver. For the scope of features supported with first-party CTI see System Manual Application Interfaces.

The first-party CTI interface on the Ascotel IntelliGate PBX does not require a licence.

Application example

Depending on the CTI application installed, the following applications for example can be implemented:

- Dialling from a database (phone book CD, ...)
- Caller identification (CLIP)
- Creating a call journal

CTI with PC Operator Console Office 1550

For information on the PC Operator Console and its features, see "PC Operator Console Office 1550", page 120.

9.2.2 Third-Party CTI

A third-party CTI is the connection of a PBX with a central telephony server. Here the telephone terminal and telephony Client (workstation PC) are allocated in the telephony server. The telephony Clients are connected with the telephony server via the network. Telephony functions and telephone states are controlled and monitored on the telephony server.

Connection to Ascotel IntelliGate

The PBX third-party CTI connection is implemented via the Ethernet interface. Terminals on ISDN and analogue interfaces can be monitored, Office terminals can be controlled additionally.

Third-party CTI requires the Ascotel TAPI driver, which is installed on the telephony server. For the scope of features supported with third-party CTI see System Manual Application Interfaces.

A CTI licence is required to use the third-party CTI interface. This activates a certain number of users (Clients), who can also use the third-party CTI interface on the PBX (see also "Licence-related System and Expansion Limits", page 593).

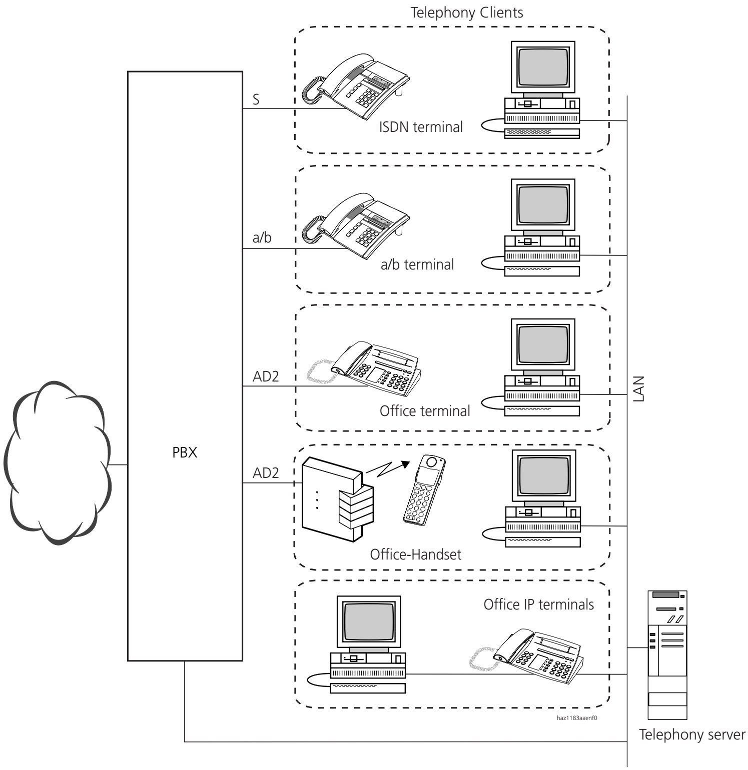


Fig. 1.49: Third-party CTI via Ethernet

Application example

In addition to the application examples with a first-party CTI solution, the following applications for example can also be implemented:

- Busy lamp field
- Group functionality
- Networked CTI solution
- Automatic Call Distribution (ACD)[1)]

1) Use of the ACD features is subject to a CTI Professional licence (see "Licence-related System and Expansion Limits", page 593)

9.3 Open Interfaces Platform

The Open Interfaces Platform (OIP) is a middleware that provides the application interfaces as an homogeneous interface for the applications. OIP does not merely access the PBX data but also external data sources such as databases or Microsoft Exchange.

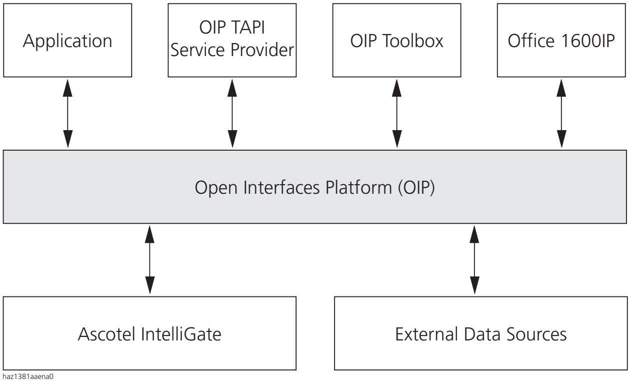


Fig. 1.50: Open Interfaces Platform (OIP) as middleware between PBX, external data sources and applications

OIP Services are a component part of the Open Interfaces Platform (OIP) and responsible for controlling the system. Individual interface functions of the PBX such as Call Control or configuration are implemented in the OIP Services.

Compared with Ascotel IntelliGate's previous CTI interface the Open Interfaces Platform (OIP) offers the following possibilities:

- OIP provides the applications with more than just telephony functions, e.g. the configuration of Office terminals or sending messages. The OIP Toolbox contains various applications for management and operation.
- OIP can be used as a telephony server to provide CTI functionalities on telephony clients. This means that the Microsoft telephony server no longer has to be used. Added security is also provided with the different rights assignment.
- With OIP it is possible to grant several applications simultaneous access to the Ascotel IntelliGate network.
- Simpler installation using a web-based installation.

Network Integration of OIP

The Open Interfaces Platform (OIP) is integrated into the Ascotel IntelliGate network via Ethernet, i.e. local connections are not required.

OIP Services

The standard settings of the OIP Services are configured during the installation of the OIP server so that the system is able to run without modifications. Details of OIP services, their setting options and access rights can be found in the Application Interfaces System Manual.

9.4 Handset Interface Audio

Audio is an audio interface with a housing similar to that of the Pocket Adapter. It is connected to the handset socket of a system terminal and is compatible with the following applications:

- Monitoring with a second handset
- Monitoring via a loudspeaker
- Recording a call with a recorder

See "Audio", page 824 for installation.

Dimensions and weight:

- Height: 26 mm
- Width: 61 mm
- Depth: 121 mm
- Weight: approx. 180 g

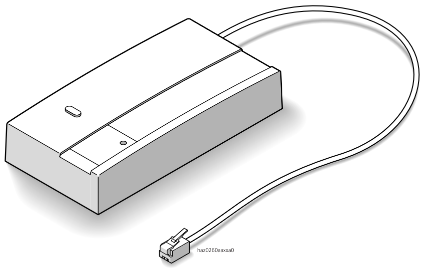


Fig. 1.51: Audio

9.5 S Bus Extension PT 10

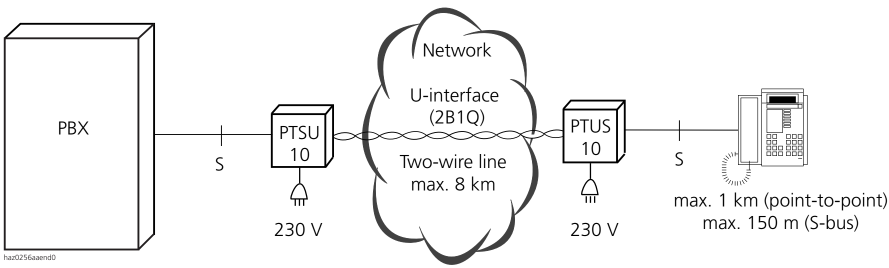


Fig. 1.52: The PT 10 S bus extension

The PT 10 S bus extension consists of two interface converters, the PTSU and the PTUS. It allows the user to convert S-bus signals into U-signals and back into S bus signals. In this way ISDN terminals (e.g. phones, radio units, PCs). at distances of up to 8 km can be connected with the PBX without problem.

Dimensions and weight:

- Height: 125 mm
- Width: 67 mm
- Depth: 88 mm
- Weight: approx. 400 g

With the S bus extension PT 10, it is also possible to network two PBXs with each other (see also "Tying-in an individual subscriber", page 697).

9.6 Ascotel Mobility Interface (AMI)

For the implementation of very large DECT systems Ascotel IntelliGate 2065 can be connected to the DECT radio system DCT 1800[1)]. The Ascotel Mobility Interface (AMI) is available for this purpose. It provides the interface between the radio system and the public ISDN network or an up-circuit PBX.

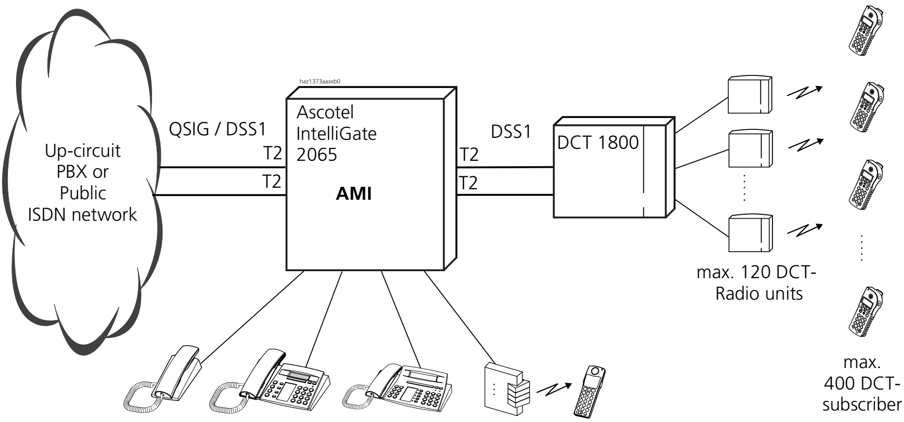


Fig. 1.53: Ascotel Mobility Interface AMI

Up to 400 DECT subscribers can be connected to the DCT 1800 radio system. A wide range of 9d generation terminals is available from the Ascom Wireless Solutions product portfolio.
For more information on Ascom Wireless Solutions products see http://www.ascom.com/ws

1800 other terminals can be connected to the Ascotel IntelliGate 2065 system in parallel with the terminals on the DECT radio system DCT 1800, providing the maximum number of subscribers does not exceed the system limits (see Tab. 3.20).

A licence is required to activate the functionality of the Ascotel Mobility Interface for connecting the DECT system DCT 1800 (see also "Licence-related System and Expansion Limits", page 593).

The DECT terminals on the DECT 1800 respond in the same way as internal S-bus subscribers. In addition the Voice Mail functionality can also be used. Information

1) An Ascom Wireless Solutions product

on newly received Voice Mail messages is indicated to the DCT subscribers on the display.

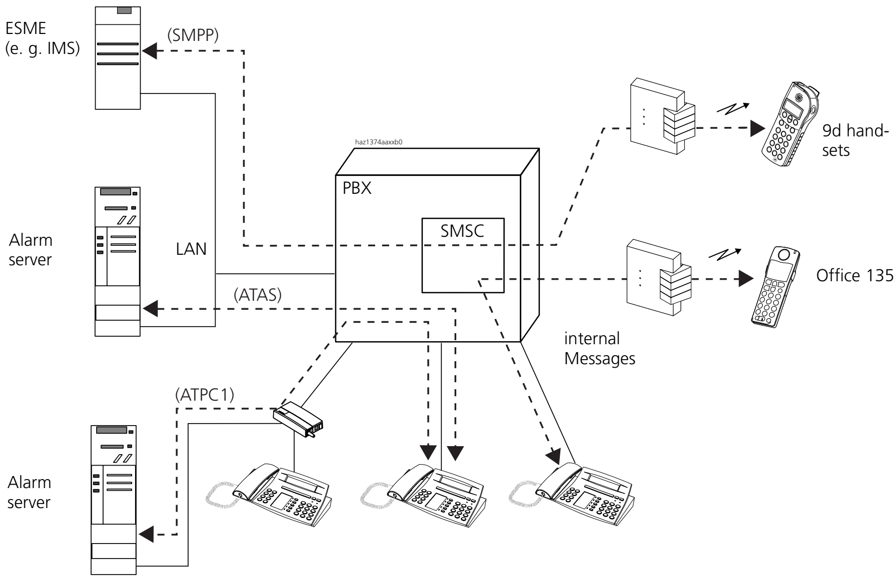
Listed below is a rough overview of the main supported features that can be used by DCT subscribers.

- Display the number / name to the called party (CLIP / CNIP)
- Display the number / name to the calling party (COLP / CONP)
- Redirecting information displayed to the caller and the called party
- Call Forwarding Unconditional (CFU, CFB) and Call Forwarding on No Reply (CFNR)
- Dealing with the call if party is unobtainable
- Callback to busy / free subscriber (CCBS / CCNR)
- Call waiting
- Hold / Enquiry call / Brokering / Call transfer
- Conference / preconfigured conference
- Differentiated ringing for internal call / external call / callback
- Notification on the display of Voice Mail messages received

As with other S-bus subscribers there is also a number of other configuration options available for call routing, call logging, subscriber settings, etc. In addition virtually all PBX features are available using */# procedures.

9.7 Message and alarm systems

Ascotel IntelliGate supports several message formats and message protocols for implementing messaging and alarm systems.



ESME (External Short Message Entity): External entity that processes short messages (SMS)
SMSC (Short Message Service Center): Software responsible for the flow of messages within the PBX
SMPP (Short Message Point-to-Point Protocol): SMS protocol

Fig. 1.54: Message and alarm systems

9.7.1 Internal messaging system for Office terminals

The internal messaging system for Office terminals allows users to exchange pre-defined or user-defined text messages between Office terminals. With the Office 45 system terminal five private text messages can also be stored for use at a later date. Text messages can also be sent to individual subscribers or subscriber groups. A maximum of 16 text messages can be stored per terminal.

The internal messaging system for Office terminals is licence-free (see also "Sending and reading text messages", page 493).

9.7.2 External messaging, monitoring and alarm systems

External messages in Short Message format (SM) are signalled to the PBX by an SM server (e.g. IMS: Integrated Message Server) via the Ethernet interface using the SMPP protocol. All the messages (internal and external) are always handed over to the SMSC (Short Message Service Centre) first, which then forwards the messages to the corresponding destination terminal. The SMSC is a software package integrated in the PBX that is responsible for the flow of messages within the PBX. The SMSC is configured using the AIMS Configuration Manager.

For external alarms from an alarm server the protocol ATAS or ATPC1 is used. The alarms are not handled by the SMSC but sent directly to the corresponding destination terminal. Additional storage locations for 16 alarms are also available for each terminal.

Alarms take priority over messages.

See also:
"Operation with external messaging, monitoring and alarm systems", page 556

Connection via V.24 with ATPC1

External alarms, faults and messages from building energy management systems, nurse paging systems, security systems, etc., can be signalled as text messages to Office terminals via the V.24 interface of the Pocket Adapter. The connected systems must also be capable of sending and receiving AT commands defined in the Ascotel standard ATPC1. A licence is not required for connecting external alarm sources or messaging systems in this way.

Connection via V.24 / Ethernet with ATAS

Compared with ATPC1 the ATAS protocol provides additional possibilities for display on the system terminals (Fox menu) and allows an alarm to be triggered using the Redkey (see "Function Redkey", page 561). The connection is also monitored, and the connection set-up is password-protected. An ATAS licence is required for enabling the protocol. This licence expands the possibilities for connecting external alarm and messaging sources to the PBX by providing the Ethernet interface and the V.24 interface of the system, in addition to the Pocket Adapter.

Interface descriptions

The ATAS and ATPC1 protocols can be disclosed to interested manufacturers of messaging, monitoring and alarm equipment on request. Contact Support or go directly to "**open.interfaces@aastra.com**".

9.7.3 Expanded messaging system with 9d handsets

With the expanded, licensed messaging system Ascotel IntelliGate can be used to implement user-friendly messaging and alarm systems. The licence enables the use of the SMPP protocol and 9d handsets to be logged on as system terminals. A wide range of alarm and message applications as well a cordless DECT terminals can then be used from the Ascom Wireless Solutions product portfolio.

The PBX is capable of communicating with up to 10 different ESMEs. Examples of ESMEs include the IMS (Integrated Message Server) and Mailgate (both Ascom Wireless Solutions products).

Ascotel IntelliGate ensures the connections between the IMS and the 9d handsets. 9d handsets do not register with Ascotel IntelliGate under the GAP standard but as system terminals. The IMS communicates with the PBX via the LAN interface. The SMPP protocol is used for this purpose.

A web-based configuration is loaded into the browser via the AIMS Shell for the configuration of the ESME.

10 Ascotel Information Management System (AIMS)

The Ascotel Information Management System (AIMS) is a software package used for the planning, configuration and monitoring of the system.. The planning and configuration can be prepared offline by the telecom specialist and then loaded on to the system locally, via LAN or via the ISDN network Remote access means that changes and expansions can be carried out independently of time and location, and is used for the remote maintenance of the system. The remote alarming function ensures that malfunctions are automatically reported to a maintenance centre via the System Event Manager (SEM). The broad spectrum of this functional management concept is complemented by call data acquisition, Least Cost Routing (LCR) and a hotel application. For the installation and settings of the interfaces for AIMS of the dial-up networking, see "Connection options", page 837 e "Settings", page 840.

AIMS runs under all the usual Windows operating systems. AIMS comprises the following program managers:

- Configuration Manager (CM)
- Fault & Maintenance Manager (FM)
- Account Manager (AM)
- Hotel Manager (HM)
- Information Manager (IM)
- Project Manager (PM)
- Upload Manager (UM)
- Voice Mail Manager (VMM)
- IP Gateway Manager (GM)
- System Event Manager (SEM)

The Upload Manager (UM) and the System Event Manager (SEM) are not available in all sales channels.

10.1 AIMS Shell

The AIMS Shell is used to administer the PBX and PINX, to regulate the access authorizations and to set the online parameters. The numbering plan for a private leased line network can be determined here on a cross-PINX basis. The following tasks can be completed from the Shell:

- PBX management
- Global operations
- PISN management
- Access to the AIMS Managers

Other useful functions include for example the consistency check, the backup function and the partial upload.

When you start up AIMS, the main window of the AIMS Shell is opened. It consists of a menu bar and a toolbar along the top edge as well as a search area. The middle section is divided into a left-hand and a right-hand pane. Displayed on the left is the menu tree with the PBX groups (network, work group, area and customer); on the right, information on the groups or the PBX.

The tabs in the lower part contain the following information:

PBX tab

- Name
- System (Ascotel 2025, Ascotel 2045, Ascotel 2065)
- PBX generation (I5, I6)
- Type of online connection (local access, dial-up access, LAN), selectable
- PBX identification number (system ID), editable
- Dial-in number for remote access to the PBX, editable
- PBX IP address, editable
- PBX software version
- Last Download

Editable fields can only be modified by users who are registered with the PBX.

"Password" tab

To enter and edit passwords for the following authorizations:

- Attendant
- Installer
- System Manager

- Support / Service Centre

The tabs "PINX connections", "Common PISN subscribers", "Common abbreviated dialling numbers" and "ISDN-4.5 Integration" are only available if a PBX configured accordingly has been selected in the network.

"PINX connections" tab

The PBXs connected with one another in the PISN are displayed complete with name. Here, provided you are authorized to do so, you can define the routes to the individual PBXs.

"Common PISN subscribers" tab

Define virtual PISN subscriber numbers. With this function it is possible to signal internally any incoming calls from a virtual subscriber and to display an internal number by way of CLIP. A separate route can be introduced for outgoing routing.

"Common abbreviated dialling numbers" tab

Set up PISN abbreviated dialling numbers defined uniformly for the entire PISN. With the function "Tools / PISN / PISN Synchronization" the common abbreviated dialling numbers can be compared under the PBX in the PISN.

10.2 AIMS Managers

The AIMS Managers are called up via the "Manager" menu or using an icon on the toolbar. Tab. 1.17 shows which Managers are password-protected and which ones are available offline and online:

Tab. 1.17: Availability of AIMS Managers

Symbol	Manager	Prior to password input	After password input	Offline (after password input and File / Open)	Online (after password input and File / Connect)
	Configuration Manager (CM)			✓	✓
	Fault & Maintenance Manager (FM)			✓	✓
	Account Manager (AM)			✓	✓
	Hotel Manager (HM)			✓	✓

Symbol	Manager	Prior to password input	After password input	Offline (after password input and File / Open)	Online (after password input and File / Connect)
	Information Manager (IM)	✓	✓	✓	✓
	Project Manager (PM)	✓	✓	✓	✓
	Upload Manager (UM)		✓	✓	✓
	Voice Mail Manager (VMM)			✓	✓
–	IP Gateway Manager AIP 6400 (GM)		✓	✓	✓

10.2.1 Configuration Manager

The Configuration Manager (CM) provides the following functions:

- Configure system and customer data offline
- Configure system and customer data online (via local access, dial-up access or LAN)
- Adapt system or customer data flexibly and quickly

10.2.2 Fault & Maintenance Manager

The Fault & Maintenance Manager (FM) provides the following functions:

- Configure remote alarming
- Display, evaluate and analyse event messages

10.2.3 Account Manager

The Account Manager (AM) provides the basis for transparent call data management with the following functions.

- Configuration of OCL and ICL data
- Configuration of the OCL and ICL output interface
- Recording of ICC data per subscriber, network interface or cost centre (totalizer only)

- Configuration and allocation of the surcharge calculator to the ICC counters
- Data import from LCR tables
- LCR management

10.2.4 Hotel Manager

The Hotel Manager (HM) can be used in hotels or establishments with similar organizational structures, for instance hospitals and retirement homes. It provides the following functions:

- Check-in and check-out
- Configure room phones
- Acquisition and printout of call charges
- Room management (room available, occupied)
- Wake-up calls for guests

10.2.5 Information Manager

Information Manager (IM) supports the AIMS user with helpful offline documentation. They are stored in Acrobat format (*.pdf) in the ...\aims\im\... subdirectory and can be opened, read or printed out using Acrobat Reader. The Information Manager contains the following documents, among others:

- AIMS Installation and Operating Instructions
- AIMS Application Notes
- Project Manager Operating Instructions
- Current information under https://pbxweb.aastra.com

10.2.6 Project Manager

The Project Manager (PM) is an Excel software program for planning the hardware components of a system.

The program determines the terminals required, the system options, user-network interfaces and network interfaces, system type, licences, system and expansion cards, connection concept, cables, etc.

It is capable of automatically generating a subset of configuration data. The data can then be printed out separately and also directly incorporated into the Configuration Manager.

Note:
The PM is not included in the AIMS Client Management Set.

10.2.7 Upload Manager

The Upload Manager (UM) is used to update the software of a system from the PC. Its characteristics are as follows:

- The software transfer can be made locally via a V.24 connection, via a LAN or via an ISDN connection (remote).
- The progress and status of the upload process are shown in a diagram.
- The precise time at which the newly loaded software is to be activated can be specified.

Note:
The UM is not included in the AIMS Client Management Set.

10.2.8 Voice Mail Manager

The Voice Mail Manager (VMM) is characterized by the following functions and properties:

- Configuration of the Voice Mail System (optional online and offline configuration)
- 2 authorization levels: Attendant and Administrator

10.2.9 IP Gateway Manager

The IP Gateway Manager (GM) is used to integrate systems in database networks via the Internet Protocol.

The Gateway Manager is characterized by the following functions and properties:

- Online configuration of the IP Gateway
- Control via web browser (as the GM is HTML-based)

10.2.10 System Event Manager

The System Event Manager (SEM) is characterized by the following properties:

- Comprehensive centralized monitoring of event messages from systems
- Installation on several PCs possible
- Particularly well suited for monitoring networked systems
- Receives and processes messages via ISDN or TCP / IP

The System Event Manager runs under Windows 95, 98, ME, 2000 or NT 4.0, based on centralized data management.

3 system components are integrated in the SEM:

- **SEM Configurator**
  Setup of the ISDN connection of the SEM server, management of the data stock, access authorization, type of responses to event messages, language selection

- **SEM Server**
  Permanent monitoring of several PBXs via ISDN, event message watchdog in the background, allocation of event messages to AIMS PBX configurations, event messages acoustic, per e-mail, via Gateway to SMS or in printed form

- **SEM Viewer**
  Event message management with priorities and status reports, coupled with the corresponding AIMS data management systems, event message filters and searches, event message export on spreadsheets, possibility of selecting trigger criteria for event messages

Note:
The SEM is not included in the AIMS Client Management Set.

10.3 Installing AIMS

To install AIMS follow the instructions contained in the "readme.txt" file. The file is included with every AIMS package and is stored as an electronic document on each AIMS CD.

10.4 Access concept

To ensure that the PBX can only be configured with AIMS by authorized personnel, access to the PBX in AIMS and in the PBX is password protected. AIMS and PBX password management can be synchronised. The initialization passwords are the same in the PBX and AIMS.

10.4.1 Authorization

AIMS keeps a list of AIMS users, complete with respective authorization levels. When AIMS is started, it checks whether the user's OS (operating system) name (as of Win 95/98, Win NT 4.0) is listed on the AIMS user list. If so, the user will have access to whichever AIMS functions correspond to his authorization level. He can log on to a PBX without having to enter a password. Under Windows 95 it is possible to start without a password or name. In such cases access to a PBX is possible only after the user has entered a PBX password.

AIMS has four password-protected authorization levels:

- Attendant
- Installer
- System Manager
- Support / Service Centre

Tab. 1.18 shows which features can be modified with each authorization level:

Tab. 1.18: Password-protected authorization levels and features

Features	Support / Service Centre	Installer	System Manager	Attendant
Create a PBX group	✓	✓	✓	✓
Delete a PBX group	✓	–	–	–
Change the name of a PBX group	✓	✓ 1)	✓ 1)	✓ 1)
Add a PBX to a group	✓	✓ 1)	✓ 1)	✓ 1)
Remove a PBX from a group	✓	–	–	–
Access to all PBX features	✓	✓	–	–
Change the password of the Service Centre / Support authorization level	✓	–	–	–

Features	Support / Service Centre	Installer	System Manager	Attendant
Change the password of the Installer authorization level	✓	✓	–	–
Change the password of the System Manager authorization level	✓	✓	✓	–
Change the password of the Attendant authorization level	✓	✓	✓	✓

1) Possible only if the user's name is listed on the user list for PBX groups.

10.4.2 Administrator

The administrator has password-free access to the PBX; he can administer the list of AIMS users, assign user rights and change the passwords for all authorizations.

Once AIMS has been installed, all users initially have administrator rights. For security reasons the user rights should therefore be changed immediately.

10.4.3 Password syntax

The following rules apply to password selection and spelling:

- A password must consist of a minimum of 4 and a maximum of 10 alphanumerical characters.
- Upper and lower case spelling is irrelevant.
- The following special characters can be used: ?, /, <, >, -, +, *, #, =,. and space.
- Do not use German "umlauts".

10.4.4 Change password

Any password on the corresponding authorization level can be replaced by a new password. Users at a higher authorization level are also authorized to change the passwords of lower authorization levels.

AIMS password lost

If the password for the highest authorization level is known, the passwords of the lower authorization levels can be changed.
If it is no longer known, contact the PBX support.

10.5 Exchanging data between PBX and PC

The system and user data is stored in the PBX and on the PC's hard disk. Both databases are serviced by the AIMS Managers. To keep the databases at the same level, you need to exchange data between the databases.

The figure below illustrates the interplay between the PC and the two databases:

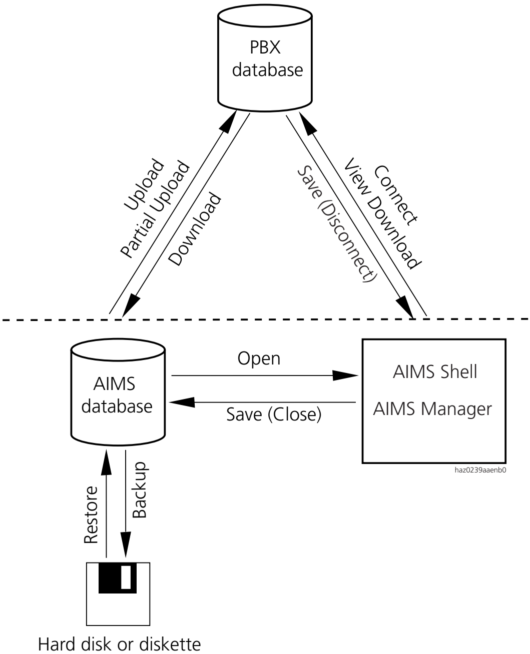


Fig. 1.55: Functions involved in exchange data between the PBX and AIMS

To edit or complement system and user data, load the data either directly from the PBX (PBX database) or from the PC's hard disk (AIMS database) into the main memory.

Note:
The data of the Voice Mail card and IPI card is administered in the corresponding Managers. The same applies to the data of the Project Manager.

10.5.1 Working offline (AIMS database)

With the AIMS database you can only work in offline mode. The following functions are available:

Open

All the data from the selected PBX is loaded from the AIMS database into the PC's main memory and made available for editing on the AIMS Shell or in an AIMS Manager. Modified parameters are signalled in the menu tree by a check mark.

Close

The AIMS database is closed. If data was modified, the system will ask whether you want to save the changes.

Note:
The changes and the check marks are retained even if the database is closed without saving.

Save

After editing, the modified data is written from the PC's main memory into the AIMS database and the check marks on the modified parameters are deleted.

Backup

The "Backup" function on the AIMS Shell stores the PBX data from the PC's main memory on to a backup file specified by the user. If the current PBX data is to be saved, it must first be loaded into the main memory using "Download".

Tip:
Make two backup diskettes of the PBX data: Keep one with the system and give the other to the Installer.

Restore

The Restore function loads all the PBX data from a backup file into the AIMS database. The data can then be transferred to the PBX using the Upload function.

10.5.2 Working online (PBX database)

Data in the PBX database can only be accessed in the online mode. The following functions are available:

Connect

Connects the PBX with the AIMS Shell. The function uses the date and time to determine whether the PBX data matches the data stored in the AIMS database. If the data does not match, all the PBX data will automatically be downloaded into the AIMS database.

Disconnect

Closes all the AIMS Managers and disconnects the online connection between the PC and the PBX.

If data was modified, the system will ask whether you want to save the changes in the AIMS database.

Save

Saves the modified data in the PBX. The system will then ask you whether you want to update the AIMS database.

Download

During a download via the AIMS Shell all the PBX data is loaded from the PBX to the AIMS database.

During the download the PBX is automatically prebarred and then automatically released once the operation is completed. Existing connections are retained. The progress of the download operation is indicated by a progress indicator bar in a window. A download can be carried out when the system is operating under full-load conditions.

Download view

The View Download function loads the PBX data of the current window in an AIMS Manager from the PBX into the PC's main memory. This partial download is available in the "Online mode" in the online mode of individual Managers (e.g. Configuration Manager, Fault & Maintenance Manager). "Download View" is used to register new or modified hardware in AIMS.

Upload

During an upload via the AIMS Shell all the PBX data is written from the AIMS database into the PBX.

During a data upload the PBX is automatically prebarred and then automatically released once the operation is completed. The progress of the upload operation is indicated by a progress indicator bar in a window.

An upload is carried out in the following cases:

- Putting a new system into operation.
- Restoring a system if the configuration has been lost.
- Replicating a special configuration, e.g. copying customer data to a different system.

Note:
A restart will be carried out automatically once the upload is completed. Any existing telephone connections will be disconnected.

Partial Upload

The Partial Upload function is used to load the following data from the PC's main memory on to the PBX.

- Abbreviated dialling numbers
- PISN subscriber data
- Terminal data (individual subscribers or block by block)
- LCR (Least Cost Routing) data

With a partial upload, configuration data that changes frequently can be transferred more quickly into the system.

Note:
A partial upload is only followed by a restart and disconnection of existing phone connections if system data is loaded up onto a PBX. In this case the system generates an appropriate message.

10.6 Import / Export

The Import / Export function allows the user to import data tables (abbreviated dialling numbers, DDI numbers and names, subscriber numbers and subscribers names, terminal data) into the AIMS database or to export such tables from the database. The exported tables are stored in Excel format and can then be sorted or modified.

10.7 Importing data from older systems

In AIMS you can import system data from AIMS 4.x databases. For this to work, AIMS 4.x has to be installed before.

Similarly it is possible to transfer files with system data from the Project Manager to the AIMS database using AIMS.

The Application Notes and Frequently Asked Questions (FAQs) in connection with AIMS can be downloaded from the Internet under "https://pbxweb.aastra.com".

10.8 Licensing concept

Some of the features (e.g. system licence upgrade 2025 → 2045, CTI interface, QSIG, etc.) are protected ni the PBX by a licence code. This type of data can also be configured without the corresponding code.
The licence conditions are set out in a list and can be viewed under "Basic setup" in the system configuration in the AIMS Configuration Manager.

Checking the features subject to licensing conditions

If the "Licensing" feature is defined and activated in AIMS, features subject to licensing can be modified offline in AIMS; if not, the appropriate warning will appear on the screen.
If the Upload function is used to load the configuration data into the PBX, it will first check the licence code. If the code is invalid or missing altogether, the user will be warned by an error message. The configuration is retained and the Upload process is continued.

Part 2 System Functions and Features

1 Overview of Chapters

Numbering Plan

This part is based on the principles of the networking philosophy discussed in Part 1 and features the different types of internal and external numbering plans available in the various systems. It explains the differences between internal numbering plans for the private network and external numbering plans for the public network. It tells you what you need to know when creating numbering plans for each particular network.

Identification Elements

Correctly identifying and displaying a call is the essential requirement for adequately implementing the system's networking philosophy. Chapter 3 looks at how the origin of a call is identified using different ringing tone patterns and how the caller's number (CLIP) or name (CNIP) is displayed. It describes how CLIP and CNIP displays are created under different system conditions, how they can be influenced, and how to suppress the CLIP display.

Routing elements

The purpose of a routing element is to distribute incoming and outgoing calls to their destinations. A good, adaptable routing architecture is extremely important in connection with private networks in particular. The system resolves call routing by using modular routing elements that ideally support any adapted, situation-based configuration. Chapter 4 features all the elements involved in call routing.

The multitude of setting options does, however, involve a considerable amount of configuration. That is why the initialization configuration has been selected in such a way that many settings no longer have to be adapted when configuring a stand-alone PBX.

Call Routing

Chapter 5 describes the interplay between the routing elements for the various types of traffic: call routing for internal, incoming and outgoing traffic. Other topics include Least Cost Routing, exchange-to-exchange traffic, transit routing in the private leased-line network, overflow routing and break-out.

Data services

Chapter 6 deals with outgoing and incoming data service connections. It looks at types of data services, the configuration of data service destination tables, and how data services are routed in the private leased-line network. Other topics include user-to-user signalling and the transition to the X.25 data network.

Call logging (CL)

Call data and call charges can be logged and evaluated in great detail with the aid of the system. Chapter 7 explains the concept of individual charge counting (ICC) and the setting options for logging call data for outgoing (OCL) and incoming (ICL) calls. It also examines other aspects such as the output concept, interface configuration for call data output, output types and the various output formats.

Features

The PBX offers a multitude of features that can be activated by the subscriber. Chapter 8 contains a systematic description of all these features.

Features Overview

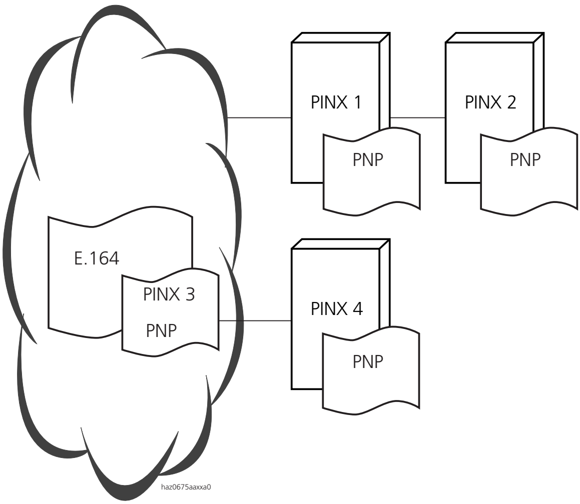
Chapter 9 contains a systematic, alphabetical list of the features. The complete overview, in tabular form, contains all the features and equipment of a system, including terminal-related data and procedures for the individual features.

2 Numbering Plan

2.1 Numbering Plan Identifiers

The numbering plan is used to analyse numbers and allocate them to an addressable destination. Two types of numbering plans (Numbering Plan Identification, NPI) are relevant to the system:

- The public network uses numbering plan identifier E.164, which is defined and standardized by the ITU-T.
- Private networks use numbering plan identifier PNP (Private Numbering Plan). The internal numbering plan of a PBX or PINX is also of the PNP type, as is the private numbering plan supplied by the public network provider.



PINX 3 is a virtual PINX (Centrex)

Fig. 2.1: Numbering plan identifiers in the public network and in the PISN (in PINXs)

Numbers in a numbering plan are analysed with the aid of the Type Of Number (or TON).

Numbering Plan Identifier E.164

Numbering plan E.164 comprises the following types of number:

Tab. 2.1: E.164 types of number

Type Of Number	Structure	Example
Subscriber	[SN]	624 11 11
National	[NDC] [SN]	32 624 11 11
International	[CC] [NDC] [SN]	41 32 624 11 11
Unknown	[NP] [NDC] [SN] [IP] [CC] [NDC] [SN]	032 624 11 11 0041 32 624 11 11

[SN] Subscriber Number
[NDC] National Destination Code
[CC] Country Code
[NP] National Prefix
[IP] International Prefix

The national and international prefixes (in Switzerland 0 for national and 00 for international long-distance traffic) are not part of the type of number. Prefix digits are sometimes also referred to as trunk prefixes.

PNP Numbering Plan Identifier

The PNP numbering plan comprises the following types of number:

Tab. 2.2: PNP types of number

Type Of Number	Structure	Example
Level 0	[RIN]	1313
Level 1	[RP1] [RIN]	60 1313
Level 2[1)]	[RP2] [RP1] [RIN]	62 60 1313

1) The system supports private networks up to and with Level 1

[RIN] Regional Intern Number: all destination numbers within a Level 0 region
[RP1] Regional Prefix 1: Prefix for a Level 1 region
[RP2] Regional Prefix 2: Prefix for a Level 2 region

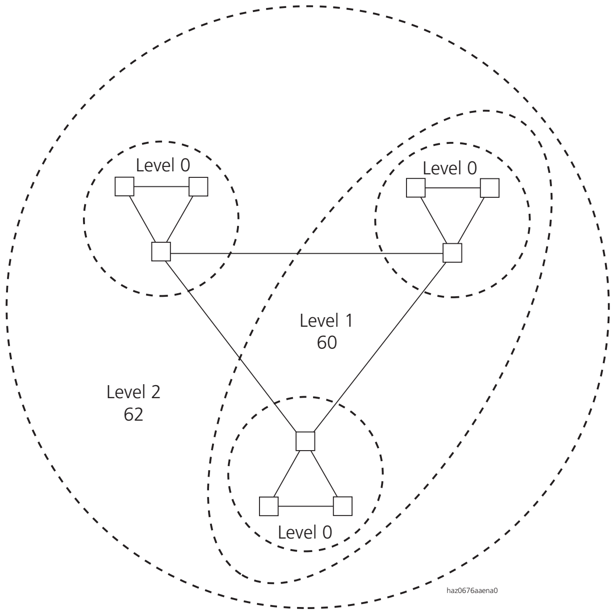


Fig. 2.2: Levels as per PNP definition

2.2 The System's Numbering Plan

The system's internal numbering plan is the numbering plan used for a stand-alone PBX or a PINX in a private network. The numbers entered in the numbering plan are used both to dial up call destinations in the PBX and to execute control functions. Call destinations and functions are grouped into categories.

The internal numbering plan:

- Assigns number ranges to the categories.
- Allocates their numbers to call destinations and control functions, making them obtainable and executable respectively.

As far as the call destination numbers are concerned, the system's numbering plan is a PNP-type numbering plan.

2.2.1 Categories in the Numbering Plan

The allocation of categories to numbers and number ranges can be freely configured, provided a number of rules are observed. Initialization values vary from country to country.

Rules for an Internal Numbering Plan

Numbers are always interpreted starting from the left..

The various categories must be unequivocally separated through number allocation.. If, for example, the PC Operator has been allocated number 11, the numbers 11n cannot be allocated to any other categories. If, however, the PC Operator has been allocated the number 111, the numbers 112 to 119 can be allocated other categories.

Numbers within a category do not necessarily have to constitute a coherent range; instead, they can be spread over the entire number range (e.g.: subscribers 200, 404, 550, 551, ...). However, for the purposes of clarity, we recommend that you define coherent ranges.

The number length is variable and can consist of 1 to 12 digits. Numbers with more than 12 digits will be truncated from the right.

Tab. 2.3: Categories in the system's numbering plan with allocated numbers

Category		Number / Number Range		
Name	Explanation	Number[1)]	Number range	Explanation
Exchange access, Business:	Call charges are added up on the "Business Telephony" or "Business Data Service" cumulative counter.	0	<ext. call No.>	Prefix, truncated before dialling out into the network
Exchange access, Private:	Call charges are added up on the "Private" cumulative counter	10	<ext. call No.>	Prefix, truncated before dialling out into the network
Operator Console	The PC Operator is stored under this number	11	–	
Emergency number	Emergency destinations are stored under this number, depending on the switching group	12	–	

Category		Number / Number Range		
Name	Explanation	Number[1]	Number range	Explanation
Exchange access with cost centre selection	The call charges are explicitly allocated to the selected cost centre.	13	<CC No.> <ext. call No.>	Prefix, truncated together with the CC No. before dialling out into the network
Exchange access with route selection	Routes the outgoing call via the selected route	170 to n[2]	<ext. call No.>	Prefix, truncated before dialling out into the network
Internal subscribers	Internal subscribers on the PBX	200 to n[3]	–	
Virtual subscribers	Virtual subscribers respond in the same way as analogue subscribers	Not allocated	–	
Abbreviated dialling numbers	Other, user-definable numbers are stored under these numbers	7000 to 7999	–	
Door intercom 1	Selects the door intercom system	851	–	
Door intercom 2	Selects the door intercom system	852	–	
User groups	User groups can be selected internally, directly with these numbers	860 to n[2]	–	
Remote maintenance access PPP	Selects the configuration interface via PPP	898	–	
Call distribution	Call distribution elements can be selected internally, directly with these numbers	Not allocated	–	
PISN subscribers	Subscribers on another PINX in the PISN	Not allocated	–	
Separate regional prefix	Level 1 prefix for region allocation in the PISN	Not allocated	–	Prefix, truncated on detection
Ascotel DECT-Subscribers	Ascotel DECT subscribers on the PBX	Not allocated	–	
Pager subscribers (S-bus PSI)	Pager subscribers on the PBX	Not allocated	–	
*-substitute	Substitute digit for pulse dialling sets without *-key	Not allocated	<Function code>	

1) Initialization values for Switzerland

2) Depends on the system type (see "System Limits", page 592).

3) Depends on the number of user-network interfaces installed. On the system 2025 / 2045 the range is 20 to m.

2.2.2 Exchange Access Categories

Tab. 2.4: Exchange access categories in the internal numbering plan

Category		Name		
Name	Explanation	Number[1)]	Number range	Explanation
Exchange access, Business:	Call charges are added up on the "Business Telephony" or "Business Data Service" cumulative counter.	0	<ext. call No.>	Prefix, truncated before dialling out into the network
Exchange access, Private:	Call charges are added up on the "Private" cumulative counter	10	<ext. call No.>	Prefix, truncated before dialling out into the network
Exchange access with cost centre selection	The call charges are explicitly allocated to the selected cost centre.	13	<CC No.> <ext. call No.>	Prefix, truncated together with the CC No. before dialling out into the network
Exchange access with route selection	Routes the outgoing call via the selected route	170 to n[2)]	<ext. call No.>	Prefix, truncated before dialling out into the network

1) Initialization values for Switzerland

2) Depends on the system type (see "System Limits", page 592).

A call can be transmitted to the public network by selecting a prefix from one of the exchange access categories.

The cost type (Business, Private), cost centre (cost centre selection) or route (route selection) is determined according to the prefix selected.

Route selection prefixes are the internal call numbers of the routes.

Route selection can also be used for routing in the private leased-line network.

2.2.3 Categories for Abbreviated Dialling and Emergency Number

Tab. 2.5: Abbreviated dialling category in the internal numbering plan

Category		Name	
Name	Explanation	Number[1)]	Explanation
Abbreviated dialling numbers	Other, user-definable numbers are stored under these numbers	7000 to 7999	
Emergency number	Emergency destinations are stored under this number, depending on the switching group	12	

1) Initialization values for Switzerland

Abbreviated dialling numbers

Abbreviated dialling numbers facilitate exchange traffic in the case of frequently used numbers. They can also be used to activate */# procedures more quickly.

An internal or external call number or a */# procedure and a name can be stored under any abbreviated dialling number

Stored Numbers

If an external number is stored, the exchange access prefix must also be entered at the same time. Prefix and number must be separated with a hyphen. The hyphen ensures that when the number is dialled via a line key, the exchange access prefix is truncated.

Only the front portion of a number can be entered at any time. The rear portion must then be suffix-dialled manually. Example:
The number 0-001212 and the name "NY" (for New York) are stored under the abbreviated dialling number 7500. Any user who wants to call Manhattan, New York, simply dials "NY" by name, then adds the local number.

Name

The name is used:

- To dial by entering the name rather than the call number (dialling by name).

- To display the name on the subscriber’s own system terminal when the CLIP number of an incoming call matches the number stored under the abbreviated dialling (see "Replicating the Name Display in the PBX", page 184).

Digit Barrings and Exchange Access Rights

When an external destination is dialled via an abbreviated dialling number the number stored bypasses the digit barring and the exchange access authorization.

When an external destination is dialled using dialling by name via abbreviated dialling, only the exchange access rights are bypassed (more on digit barrings and exchange access rights see "Digit Barring Facilities", page 288).

Emergency number

The emergency number is used to quickly dial an emergency number destination. Depending on the switchover, up to three internal call numbers or one external call number can be stored under it.

Stored Numbers

One number can be stored for each of the three switchover positions of switching group 1 (more on the switching groups see "Switch Groups", page 233).

Entering the numbers is subject to the same rules as for abbreviated dialling.

Digit Barrings and Exchange Access Rights

The same rules apply as for abbreviated dialling

2.2.4 Subscribers Categories

Tab. 2.6: Subscriber categories in the internal numbering plan

Category		Number / Number Range
Name	Explanation	Number[1)]
Internal subscribers	Internal subscribers on the PBX	200 to n[2)]
Virtual subscribers	Virtual subscribers respond in the same way as analogue subscribers	Not allocated

Category		Number / Number Range
Name	Explanation	Number[1)]
PISN subscribers	Subscribers on another PINX in the PISN	Not allocated
Ascotel DECT-Subscribers	Ascotel DECT subscribers on the PBX	Not allocated
Pager subscribers (S-bus PSI)	Pager subscribers on the PBX	Not allocated

1) Initialization values for Switzerland

2) Depending on the number of user-network interfaces installed.

Internal subscribers

The numbers within this category are allocated terminals on user-network interfaces of the PBX. Pager and cordless subscribers are not part of this category.

The specific settings for an internal subscriber will be examined in the subscriber configuration.

Allocating a name to an internal subscriber in the subscriber configuration makes it possible to:

- Dial the subscriber internally by entering his name instead of the call number (dialling by name)
- Display the name on the terminal of a destination subscriber on the system's own PBX / PINX or on another PINX in the PISN (CNIP)

Virtual subscribers

Virtual subscribers respond in the same way as analogue internal subscribers except that they

- do not physically occupy a port as there is no hardware involved,
- do not require a B channel.

Other properties

- When the caller dials a virtual subscriber he obtains the ring-back tone or the busy tone (if the subscriber is already in a call).
- Virtual subscribers are capable of sending and receiving messages via third-party CTI interface.

- Virtual subscribers belong to the group of subscribers with their own DDI number, the maximum number of which is restricted by the system limits per system.
- Virtual subscribers have their own recall time, which can be set throughout the system. It is used if no recall time is defined in the subscriber setting (see also "Recall", page 472).

Application examples:

- During an explicit call transfer without prior notice to a virtual subscriber a call can be "parked" for up to 900 seconds and then fetched using *86 <SC No.>.
- To integrate a PISN subscriber into a user group, it is possible to accept a virtual subscriber in the user group using a CFNR to the PISN subscriber.
- In third-party CTI applications virtual subscribers can be used to send and receive messages.

Cordless subscribers

Although Ascotel DECT-subscribers are also part of the group of internal subscribers, they are allocated to a separate category in the numbering plan as their destination address is not a physical user-network interface; instead, a handset identification has to be stored under each number. This is done using a procedure for logging handsets on to the PBX.

The specific settings for a cordless subscriber will be examined in the subscriber configuration.

Allocating a name to a cordless subscriber in the subscriber configuration makes it possible to:

- The subscriber to be dialled internally by entering his name rather than the call number (dialling by name)
- Display the name on the terminal of a destination subscriber on the system’s own PBX / PINX or on another PINX in the PISN (CNIP)

PISN subscribers

This category comprises subscribers who belong to the same PISN but are connected to a different PINX. They can also be subscribers of a virtual PINX.

The numbers of user groups, call distribution elements, abbreviated dialling destinations, routes or door intercoms can also be entered as PISN subscribers, besides the numbers of internal subscribers..

The specific settings for a PISN subscriber will be examined in the subscriber configuration (stored call number, route, name, see "Subscriber Configuration", page 246).

Entering PISN Subscribers

There are two ways of entering PISN subscribers:

- A PISN subscriber's call number is entered in full and unequivocally (Fig. 2.3, PINX 2).
- One number with wildcards is entered for several PISN subscribers (PISN subscriber group,.Fig. 2.3, PINX 1, PISN subscribers D and E).

These variants can also be combined (Fig. 2.3, PINX 1).

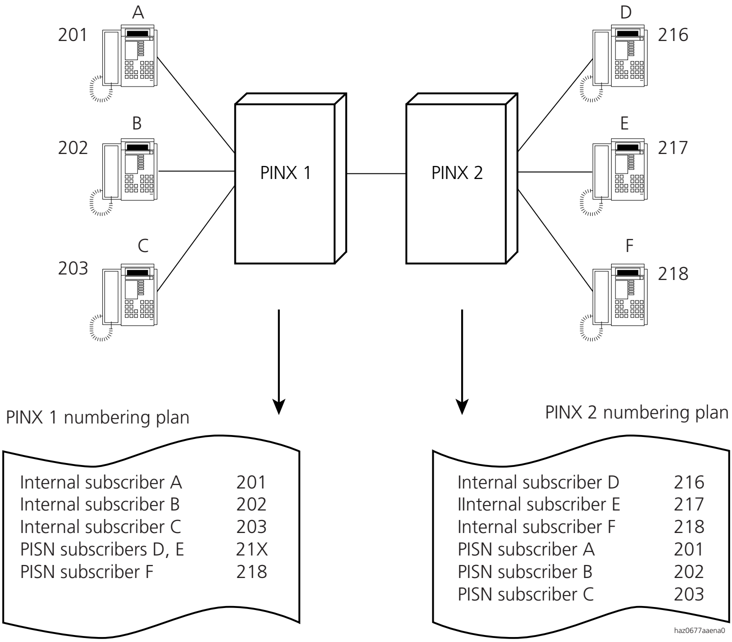


Fig. 2.3: PISN subscribers entered with and without wildcards

Entering the Number of a PISN Subscriber in Full

A complete PISN subscriber number unequivocally identifies a subscriber at another PINX or a virtual subscriber.

Each unequivocal number of a PISN subscriber can be allocated a name in the subscriber configuration. This enables:

- these subscribers to be dialled by entering the name rather than the call number (dialling by name)
- the name of a virtual PISN subscriber to be displayed (CNIP)

Entering Wildcards for a PISN Subscriber Group

A number with wildcards identifies a PISN subscriber group (Fig. 2.3, PINX 1). They can be:

- Internal subscribers of one or more PINXs
- PISN subscribers of another region

The wildcard is entered as an upper case (e.g. 21X).

This method of entering PISN subscribers helps to reduce the number of entries made. Moreover, not all the changes made to the internal subscribers of a PINX need to be updated in the other PINXs. However, neither the call numbers nor the names of the individual subscribers in the group are stored in a phone book (it is not possible to retrieve the number from a phone book nor is dialling by name possible, except if the number and name are also stored locally in a private phone book).

Tip:
It is advisable to enter PISN subscribers first with wildcards in an initial stage so that the numbering plan is quickly and transparently available throughout the PISN, and is also already operational.
All the PISN subscribers to be available with dialling by name can then be entered individually at a later stage

Entering a Regional Prefix

If an individual or group entry belongs to another PISN region, the entry for the PISN subscriber must be preceded by the regional prefix.

Example of Entering PISN Subscribers

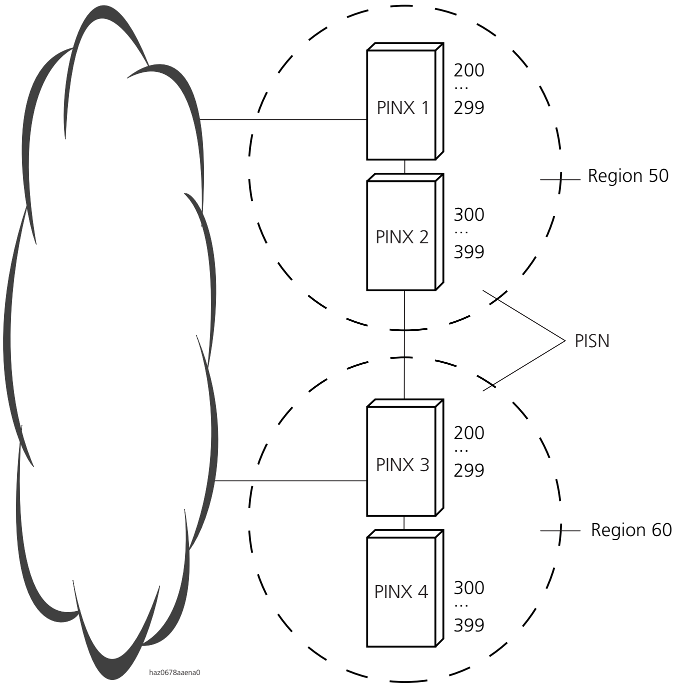


Fig. 2.4: PISN with two regions

Tab. 2.7: Entering PISN subscribers in PINX 2

Variant	Number of entries	PINX 1	PINX 3	PINX 4
Number in full	300	200,201...299	60200, 60201...60299	60300, 60301...60399
Numbers partly with wild-cards	12	20X, 21X...29X	602XX	603XX
			PINX3 and PINX 4	
Numbers with maximum possible wildcards	2	2XX	60XXX	
Combination: number in full and number with wildcards	5	2XX, 211	60XXX, 60211, 60311	

2.2.5 Separate Regional Prefix Category

Tab. 2.8: Category for separate regional prefix in the internal numbering plan

Category		Number / Number Range	
Name	Explanation	Number	Explanation
Separate regional prefix	Level 1 prefix for region allocation in the PISN	Not allocated	Prefix, truncated on detection

This regional prefix allocates a PINX to a PISN region..
The PINX compares its own regional prefix entry with the first few digits of the call numbers of the following calls:

- All outgoing calls
- All incoming calls routed via a trunk group with the setting "Network type = private"

If the first few digits match up with the PINX's own regional prefix, they will be truncated.. The remaining number is then analysed and forwarded

2.2.6 Shared Numbering Plan

PISN subscribers are structured in the internal numbering plans of the PINX.

From the PINX's viewpoint its own subscribers are internal subscribers and the subscribers of the other PINXs are PISN subscribers..

If two or more PINXs are structured in such a way that they split the range of subscriber numbers among themselves, we talk of a shared numbering plan. Together they form a region, within which all subscribers can be reached under the internal call number.

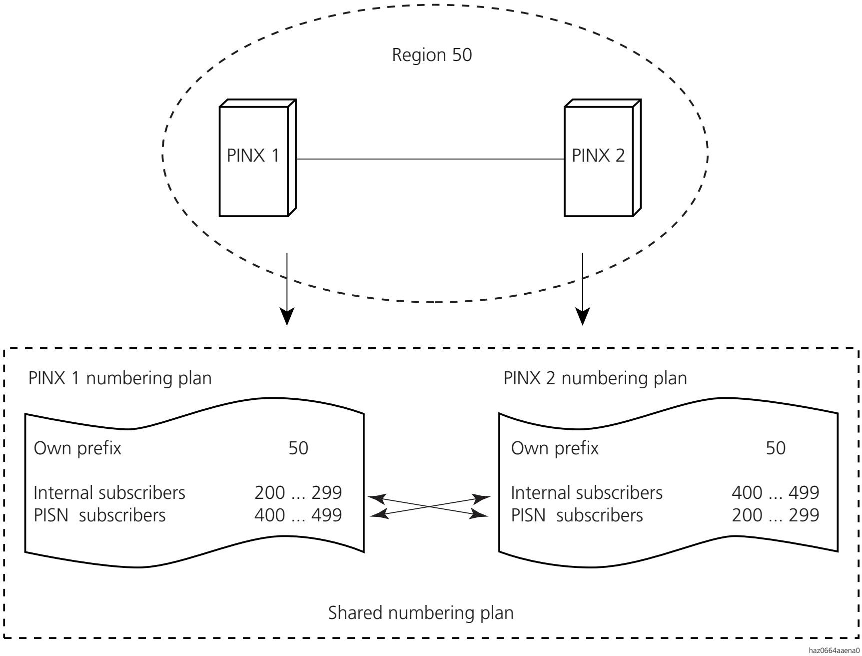


Fig. 2.5: Shared numbering plan: two PINXs share the numbers of a numbering plan.

2.2.7 PISN with different Regions

A PISN can be divided into several regions. Each region is identified by its regional prefix.
Subscribers who call a subscriber in a different region first dial the prefix of the destination region, then the internal number of the subscriber they want.

Their specific regional prefix is specified in the internal numbering plan of each PINX.

The organization of the numbering plans does not depend on the PISN topology.

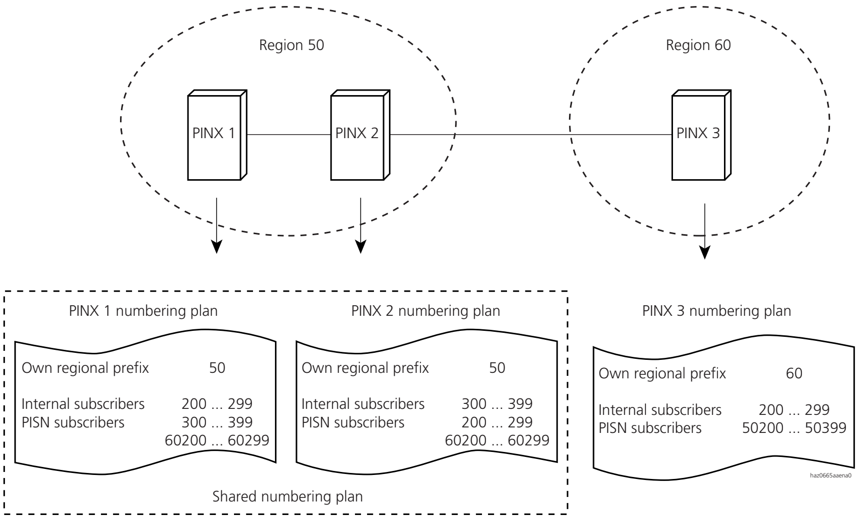


Fig. 2.6: PISN with two regions and shared numbering plan for Region 50

Entering a Regional Prefix

In the example above the PISN subscribers of a different region are entered with the regional prefix (for example 60200 to 60299).

Another possibility is to define a route with call number 60 and to enter the PISN subscribers without regional prefix (route method).
The subscriber dials exactly the same number, for example 60250, but this time the call is routed as a route selection. It uses the route with call number 60 and not the one allocated to the PISN subscriber in the subscriber configuration. (In the example above the numbers would have to be distributed differently since number ranges cannot be assigned twice.)

3 Identification Elements

In keeping with the system's networking philosophy, it is essential for calls to be correctly identified and displayed. Many settings therefore serve the purpose of correct call identification.

A call is identified firstly by the type of acoustic ringing (i.e. ringing pattern) and, secondly, by the display on the terminal.

The initialization values are selected in such a way that the ringing patterns and displays appear correctly in most cases. Changes to the settings are necessary only in exceptional cases.

3.1 Internal and External Ringing Patterns

The ringing pattern provides a means of identifying whether the call originates from within the PBX (internal call) or from the outside (external call). The rhythm of the ringing pattern differs in each case

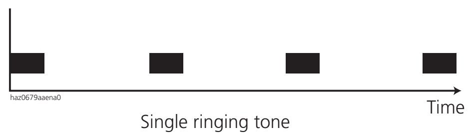


Fig. 2.7: Internal ringing pattern[1)]

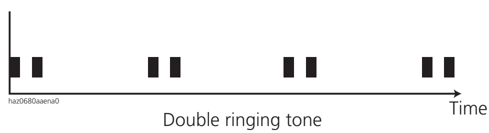


Fig. 2.8: External ringing pattern[1)]

Calls with the "Internal" Ringing Pattern.

- Calls from internal subscribers
- Calls from the public network if "Ringing pattern = single ringing tone" is set in the subscriber configuration.
  This setting is recommended if the subscriber is a terminal (not a system terminal) that automatically answers a call (e.g. a fax machine) as not all devices are capable of correctly interpreting the external ringing pattern.

1) In some countries the internal and external ringing patterns may be used in precisely the opposite way.

- Calls from subscribers from the private network (PISN subscriber):
  - Calls from the private leased-line network
  - Calls from virtual network PISN subscribers
- An enquiry call from a subscriber with an outside call on hold if "Ringing pattern internal / external = internal" is set in the common settings (see "Hold (enquiry call)", page 460).

Calls with the "External" Ringing Pattern:

- Calls from the public network
  - if they do not originate from a virtual network PISN subscriber and
  - if "Ringing pattern = automatic" is set in the subscriber configuration.
- An enquiry call from a subscriber with an outside call on hold if "Ringing pattern internal / external = external" is set in the common settings (see "Hold (enquiry call)", page 460).
  The "Ringing pattern internal / external" setting is valid throughout the system.

Identifying the Origin of a Call

If an incoming call's CLIP number corresponds to numbering plan identifier E.164, the system assumes that the call comes from the public network.

If an incoming call's CLIP number corresponds to numbering plan identifier PNP, the system assumes that the call comes from the PISN.

If the CLIP number's numbering plan identifier is unknown, the trunk group configuration is used to decide whether the call is signalled internally or externally (setting "NPI call unknown").

See also:
"Numbering Plan Identifiers", page 161

3.2 Displaying Numbers (CLIP) and Names (CNIP)

During both the ringing phase and the call itself the caller's call number or name (or both) are shown on the display of a system terminal or ISDN terminal.

- The indication of the caller's phone number is referred to as CLIP (Calling Line Identification Presentation).

- The indication of the caller's name is referred to as CNIP (Calling Name Identification Presentation).

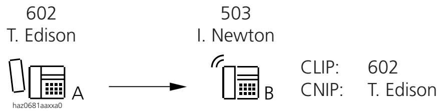


Fig. 2.9: CLIP and CNIP

When the destination subscriber answers the call, the number or name of the destination subscriber is transmitted and displayed to the caller:

- The indication of the number is referred to as COLP (Connected Line Presentation).
- The indication of the name is referred to as CONP (Connected Name Presentation).

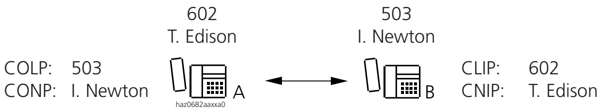


Fig. 2.10: COLP and CONP

These identification elements allow the use of other features such as logging unanswered calls on the destination subscriber's call log; the destination subscriber can then return the call by dialling the CLIP number.

These identification elements are available only in ISDN networks and even then only to a limited extent. As CNIP and CONP are not supported by the public network, the system tries to replicate them by searching through the internal phone books for a number that matches the CLIP or COLP number. If there is a match, the name entered there is displayed (see "Replicating the Name Display in the PBX", page 184).
CNIP and CONP are supported in the private network under QSIG. They are both accepted and do not need to be recreated in the PBX.

The CLIP and COLP numbers also contain the information of the NPI numbering plan type and the TON Type of Number (see "Numbering Plan Identifiers", page 161).
The system needs this additional information for a correct number analysis, particularly as a PINX in a PISN. It is not displayed on the user's terminal.

CLIP Numbers Outside the Registered Number Range

Sometimes the CLIP number transmitted to the public network is not within the registered number range Network providers have different ways of responding to this situation:

- The network operator uses the PINX master number as the CLIP number and sends it on to the destination subscriber.
- The network operator sends on to the destination subscriber the CLIP number received.
  Usually this requires an agreement with the network provider (special arrangement).

In the following cases a PINX sends the CLIP outside the registered number range:

- If a freephone number (0800...) is to be displayed as the CLIP
- In the case of overflow routing via a different gateway PINX (see page 343 and example in Tab. 2.16).
- In the case of break-out routing (see page 348)
- If a break-in situation is to be forced

Displaying the CLIP

CLIP functions process incoming and outgoing calls.

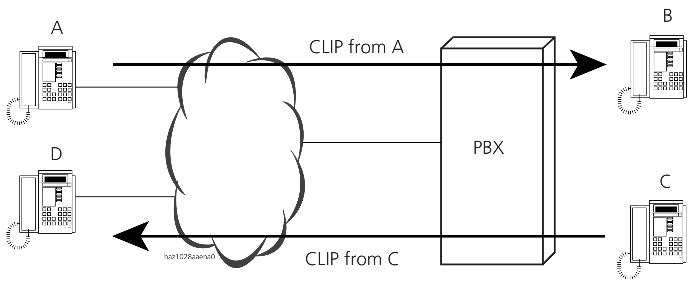


Fig. 2.11: CLIP of an incoming and an outgoing call

CLIP of an Incoming Call

Subscriber A calls subscriber B:
Subscriber A sends his CLIP, which is received in the PBX by the trunk group, processed and displayed to subscriber B.
For more details see as of page 187.

CLIP of an Outgoing Call

Subscriber C calls subscriber D:
Subscriber C sends his CLIP number, which is processed in the PBX If there already is a direct dialling and a corresponding allocation, the CLIP number is adapted and sent to subscriber D.
For more details see as of page 187.

The initialization configuration has been selected so that the CLIP display is correct. The relevant settings do not normally have to be adjusted.

3.3 CLIP with Incoming Calls

The CLIP number of an incoming call is processed and presented in two stages:

- Analysis and processing of the CLIP number
- Presentation of the CLIP number on the destination subscriber's terminal

3.3.1 Analysing and Editing the CLIP

The following information is necessary for specifying the CLIP properties in a PISN correctly. . This sub-chapter can be skipped in the case of the configuration of a stand-alone PBX.

The system analyses and adapts the CLIP number of an incoming call as accurately as possible so that the CLIP number is always displayed correctly, even in a PISN. For this purpose CLIP number prefixes such as regional prefix, prefix and code are evaluated, and the type of number adapted.

The tables below show how the system handles the type of number and the CLIP number of an incoming call.

Tab. 2.9: Handling a CLIP number with NPI-type "PNP" or "unknown.

TON of the CLIP number	Specific Regional prefix[1)]	Conversion
unknown, level 1, level 2	yes	Regional prefix is truncated, TON is set to "level 0".
	No	CLIP number and TON remain unchanged
level 0	No	CLIP number and TON remain unchanged

[1)] CLIP number has a regional prefix that matches the separate PINX.

Tab. 2.10: Handling a CLIP number with NPI-type "E.164"

TON of the CLIP number	Prefix	Conversion
Unknown	International prefix	Prefix is truncated, TON is set to "international". Further processing, see TON = international
	National prefix	Prefix is truncated, TON is set to "national" Further processing, see TON = national
	No prefix	CLIP number and TON remain unchanged
International	Country code that matches the separate PINX	Code is truncated, TON is set to "national" Further processing, see TON = national
	No matching country code	CLIP number and TON remain unchanged
National	Long-distance code that matches the separate PINX	Code is truncated, TON is set to "subscriber".
	No matching long-distance code	CLIP number and TON remain unchanged
Subscriber		CLIP number and TON remain unchanged

See also the examples in "Examples of CLIP Displays in the PISN", page 199.

3.3.2 Presentation of the CLIP on the Terminal

Call from the Public Network

If a call originates from the public network, the prefix for Business exchange access followed by a hyphen is added to the CLIP number (e.g. 0-333 33 33) so that the called party can call back simply by dialling the number displayed.

Call from a Subscriber in a Virtual Network

If a call originates from a virtual network subscriber, the call number to the PISN subscriber ("Number" setting in the subscriber configuration) is used to convert the CLIP number into the PISN subscriber number and NPI is set to "PNP" (see also examples on page 207).

Destination Subscriber not a System Terminal

If the destination is not a system terminal; the CLIP number is handled in the same way as with system terminals but without adding a hyphen.

Call via a V.24 Interface

The CLIP numbers are forwarded to the V.24 interface of an Office Pocket Adapter in the same way as they are displayed on the system terminals.

With transferred calls the CLIP number can be forwarded with the Office Pocket Adapter via V.24.

Calls with suppressed CLIP (CLIR)

If a caller uses the CLIR function to suppress his CLIP display to the called party, the system terminal displays "Number suppressed" instead of the CLIP.

Calls without CLIP

"Number unknown" is displayed on the system terminal for calls without CLIP.

3.3.3 Replicating the Name Display in the PBX

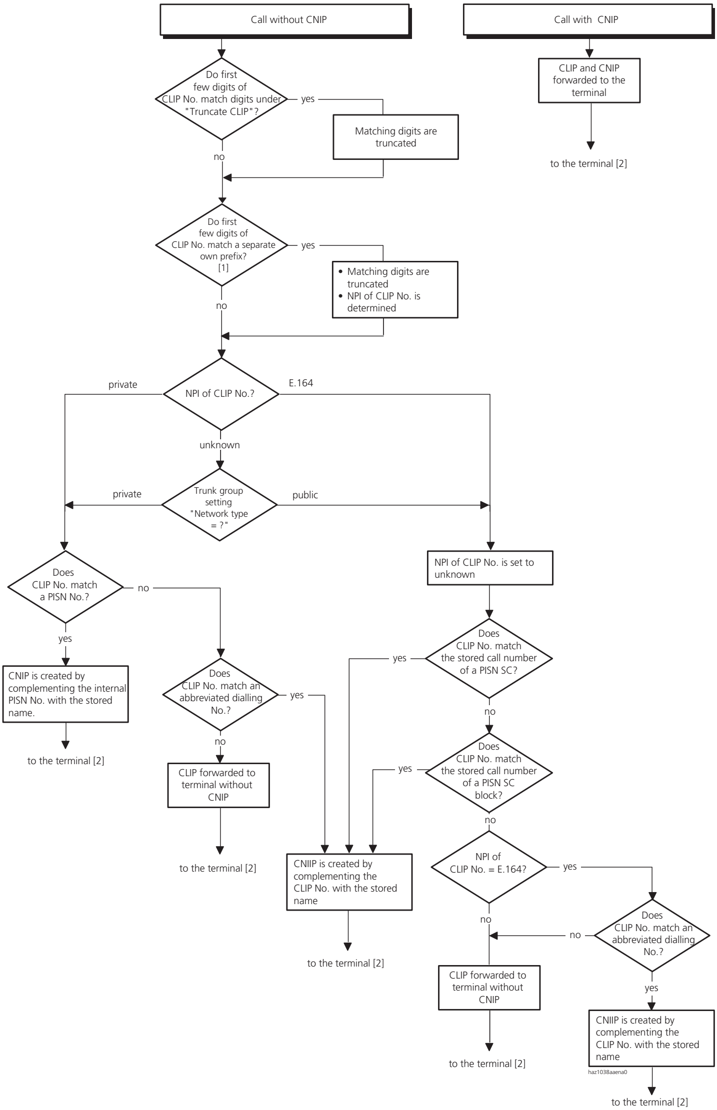
The PBX will try to assign a name to the CLIP number of an incoming call from the public network and to display that name on the system terminal (CNIP). A search is therefore carried out in the PBX card files for a match for the CLIP number. The card files are searched in the following sequence:

- PISN subscriber list
- Abbreviated dialling list
- Local card files of the system terminals.

A name will be displayed depending on the search result as shown in Fig. 2.12.

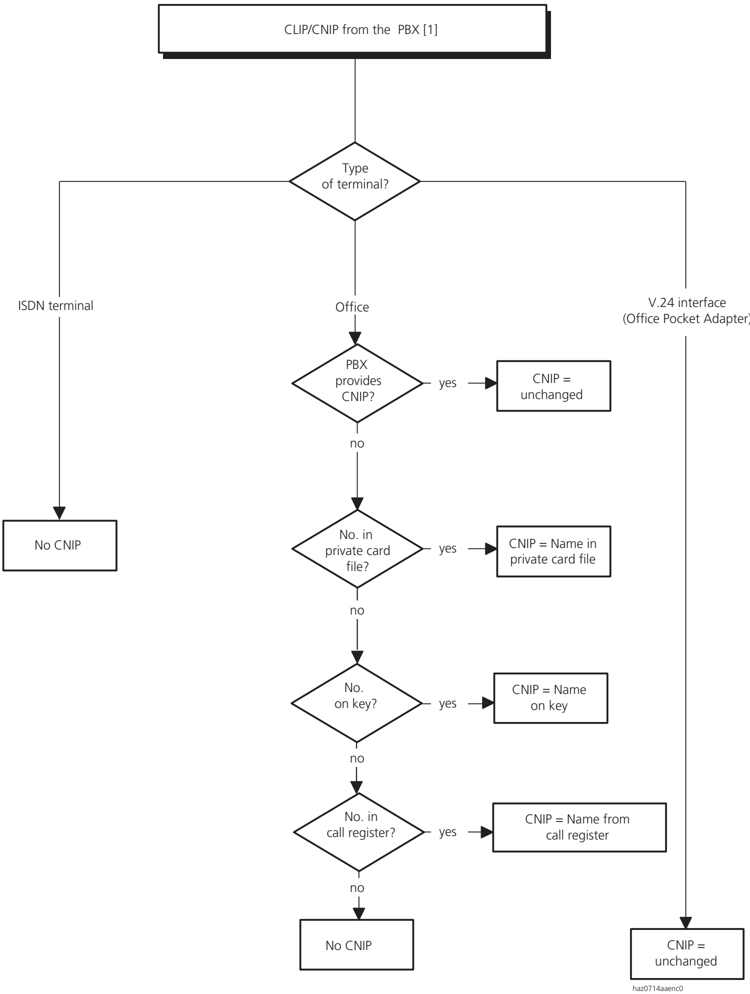
CNIP and CONP are supported in the private leased-line network under QSIG. They are both accepted and do not need to be recreated in the PBX.

3.3.4 Flow charts for name identification (CNIP)



[1] Possible prefixes: own prefix, country code, area code or own regional prefix.
[2] Continues on Fig. 2.13.

Fig. 2.12: Analysis and processing of an incoming call in the PBX



[1] From Fig. 2.12.

Fig. 2.13: Presentation of the CLIP / CNIP of an incoming call on the terminal

3.4 CLIP with Outgoing Calls

With an outgoing call the CLIP number is transmitted along with the NPI and TON information. In principle there are two possible variants for creating a CLIP number:

- The PBX creates the CLIP number automatically, based on the origin and routing of the call.
- A number is entered permanently as the CLIP number in the subscriber configuration.

3.4.1 Creating the CLIP in the PBX

With the "Automatic CLIP = Yes" setting in the subscriber configuration, the PBX generates a CLIP number. If there is a suitable DDI number for the calling subscriber, that number will be used.
A suitable DDI number is a number in a direct dialling plan which

- is linked directly or through a user group to the calling subscriber via a call distribution element, and
- is linked with the same trunk group via which the outgoing call is routed.

If there is more than one suitable DDI number, the lowest one is used.

The trunk group settings are used as the numbering plan identifier and type of number.

If there is no suitable DDI number, the trunk group settings are used for calls into the public network (Fig. 2.14), for calls into the private leased-line network it also depends how the automatic CLIP is set in the trunk group configuration (Fig. 2.16).

3.4.2 Entering a fixed CLIP

In practice a permanent CLIP number is used if the CLIP of the subscriber concerned is always to remain the same in the public network, regardless of the path used for routing an outgoing call. Break-out is a typical application (see page 348).

If a call goes out to the public network, the permanent CLIP number is retained unchanged together with the numbering plan identifier NPI and the type of number TON, even if the call is routed via another PINX (see example on page 204).

The CLIP number required, the numbering plan identifier NPI and the type of number are entered in the subscriber configuration. The "Automatic CLIP" setting, also in the subscriber configuration, must be set on "No".

"E.164" is normally set for numbering plan identifier NPI.

3.4.3 Suppressing CLIP / COLP (CLIR / COLR)

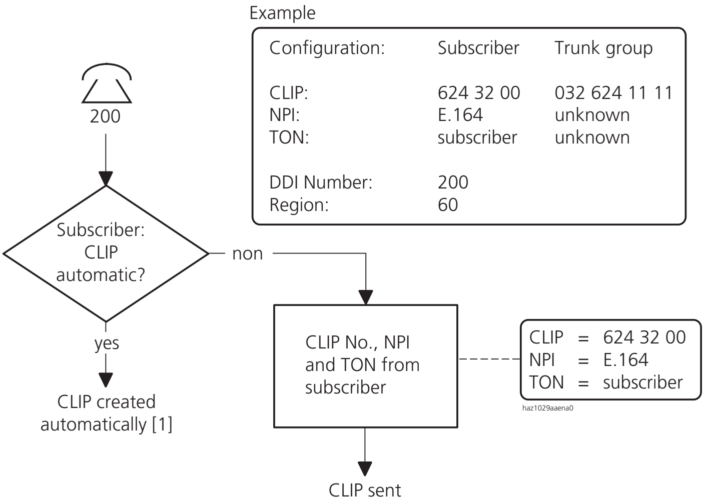
If "CLIR = Yes" has been set in the caller’s subscriber configuration, the information sent along with the CLIP and COLP numbers specifies that they are not to be displayed to the call’s recipient (CLIR: Calling Line Identification Restriction, COLR: Connected Line Presentation Restriction). In this case the network provider does not forward the CLIP number to the recipient (the CLIP number may nonetheless be sent to a number of public authorities, such as the police, see also "Displaying CLIR", page 199).

The same setting is also used to prevent the name being displayed to the call's recipient. The suppression of CNIP (Calling Name Identification Presentation) and CONP (Connected Name Identification Presentation) is called CNIR (Calling Name Identification Restriction) and CONR (Connected Name Identification Restriction).

Depending on the network provider it may be necessary to subscribe to CLIR.

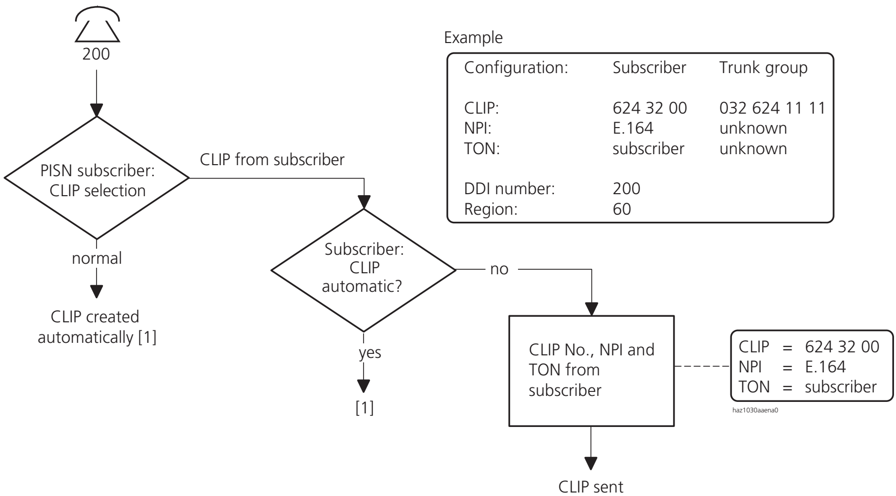
CLIR can also be temporarily activated for an outgoing call (see "Suppression of the call number display for each call", page 530).

3.4.4 CLIP flowcharts for Outgoing Calls



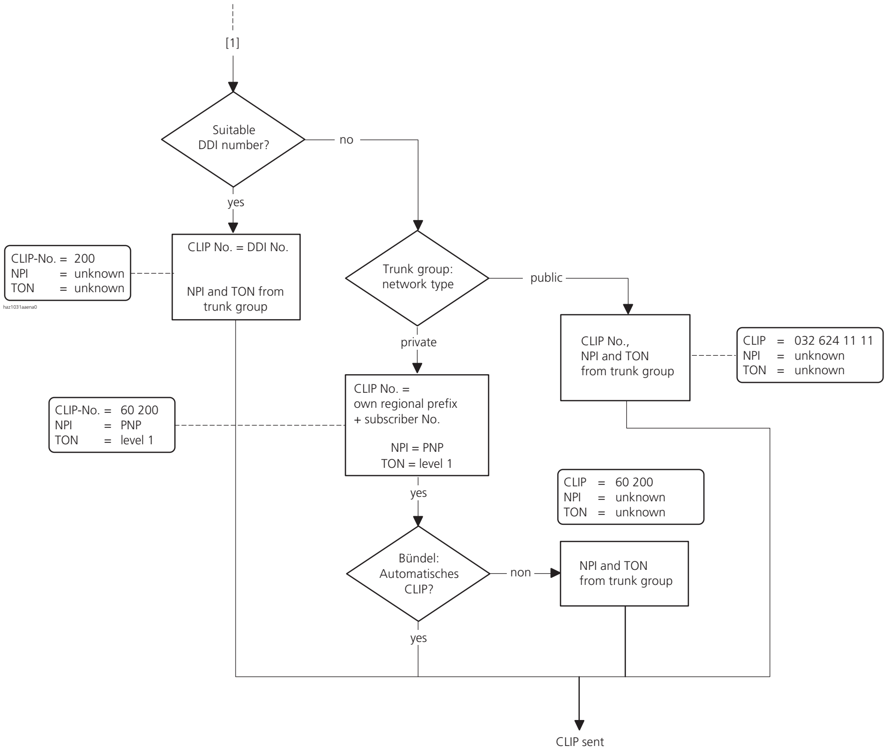
[1] Continues in Fig. 2.16.

Fig. 2.14: CLIP of an outgoing call to an external subscriber in the public network



[1] Continues in Fig. 2.16.

Fig. 2.15: CLIP of an outgoing call to a PISN subscriber



[1] From Fig. 2.14 or Fig. 2.15

Fig. 2.16: Creating an automatic CLIP for outgoing calls

3.4.5 CLIP Display with a Virtual PISN Subscriber

A public network subscriber can be set up as a virtual PISN subscriber in the PBX. Internal subscribers will then perceive the subscriber as another internal subscriber: A call is signalled with the internal ringing pattern. The internal number can also be dialled for outgoing calls.

Individual mobile subscribers or entire number blocks can be integrated in this way.

Setting up a Virtual Subscriber

A PISN subscriber is set up for this purpose (see "Numbering Plan", page 199). Enter the public network subscriber's full number under Number" For outgoing calls the configured number will be dialled via the configured route instead of the dialled PISN subscriber number. This mechanism is similar to the one used for abbreviated dialling.

When the subscriber calls up from the public network, his CLIP number will be compared with the numbers of all the PISN subscribers. If there is a match, the called subscriber is shown the PISN subscriber number by way of CLIP instead of the CLIP sent from the public network.

3.5 Display for Call Forwarding Unconditional

When Call Forwarding Unconditional is activated, it is useful for subscribers to know that the call was redirected, by whom and to whom. This means the called subscriber is able to answer the call on behalf of the subscriber who redirected the call to him. With this information the calling subscriber is better prepared for the call. This redirecting information is available on system terminals and ISDN terminals both internally and in private networks. If the public network operator supports the function, the redirecting information is also available to virtual PISN subscribers and subscribers in the public network.

3.5.1 Information displayed to the called subscriber

The called subscriber sees not only the caller's name and number but also that the call was redirected and who redirected it (redirecting information).

Example:
Subscriber A calls subscriber B, who has redirected to subscriber C. The display on an Office 45 system terminal at subscriber C reads:

<CNIP A> / <CLIP A> forwarded from <CNIP B> / <CLIP B>

This redirecting information at subscriber C is available for "CFU", "CFB", "CFNR" and "Call Deflection (CD)". (With CD "forwarded from" is displayed instead of "Deflected from".)

Outgoing call with local call forwarding

The configuration possibilities for the redirecting information depend on the destination subscriber:

If the destination subscriber is

- an internal subscriber in the local PINX, the redirecting information is always transmitted to the called subscriber.
- a PISN subscriber, a virtual PISN subscriber or a public network subscriber, you can select in the trunk group configuration whether the redirecting information is to be sent to the called subscriber or suppressed ("Send redirection/redirecting information= Yes / No").
- a public network subscriber and if CLIR is activated at the subscriber who carried out the redirecting, the called subscriber will see neither the originator of the call nor that it has been redirected. This even though the calling subscriber did not activate CLIR. To prevent this, you can set the "CLIR" parameter in the trunk group configuration to "Yes, but not for redirecting".

In a call forwarding chain with several subscribers the name/number of the first subscriber in the chain is displayed as redirecting information to the called subscriber.

Incoming call with CDE overflow

If in the event of a CDE overflow the call is routed from one call distribution element to another due to entries under "CDE if busy" or "CDE if no answer", the redirecting information provided to the called subscriber depends on the new destination:

If the destination is

- an internal subscriber or a subscriber in a private QSIG network, the name/number of the CDE is transmitted.
- a virtual PISN subscriber, the direct dial number to which the called is made is transmitted.
- an external subscriber in the public network, no redirecting information is transmitted.

Incoming call that is already redirected

The redirecting information is also available to the called subscriber in the case of an incoming call redirected via a PISN subscriber or a subscriber in the public network. If the call is routed via a call distribution element it is useful in certain cases if the name/number of the CDE is displayed instead of the redirecting information. For this, set the parameter "Show redirecting information instead of CDE name" to "No" in the CDE configuration ("Initialization value = Yes").

3.5.2 Information displayed to the calling subscriber

The calling subscriber sees not only the called party's name and number but also that the call is being redirected and to whom (redirecting information).

Example:
Subscriber A calls subscriber B, who has redirected to subscriber C. The display on an Office 45 system terminal at subscriber A reads:

<CNIP B> / <CLIP B> forwarded to <CNIP C> / <CLIP C>

This redirecting information at subscriber A is available for "CFU", "CFB" and "Call Deflection (CD)". (With a CD "forwarded to" is displayed instead of "Deflected to".)

Incoming call with local call forwarding

The caller's configuration possibilities for the redirecting information depend on the call's origin:

If the caller is

- a subscriber in the local PINX, the redirecting information is always transmitted to the calling subscriber.
- for a PISN subscriber, a virtual PISN subscriber or a public network subscriber, you can select in the trunk group configuration whether the redirecting information is to be sent to the calling subscriber or suppressed ("Send redirecting information = Yes / No").

- a public network subscriber or if the subscriber who redirected the call has activated COLR, the caller will not see that he is being redirected. If this setting is required only for internal redirected calls but not external ones, the "COLR" parameter in the trunk group configuration can be set to "Only for redirection to external".

In a call forwarding chain with several subscribers the name/number of the last subscriber in the chain is displayed as redirecting information to the calling subscriber.

Incoming call with CDE overflow

If in the event of a CDE overflow the call is routed from one call distribution element to another due to entries under "CDE if busy" or "CDE if no answer", the redirecting information provided to the calling subscriber depends on the new destination:

If the destination is

- an internal subscriber or a subscriber in a private QSIG network, the name/number of the CDE is transmitted.
- a virtual PISN subscriber or an external subscriber in the public network, no redirecting information is transmitted.

Outgoing call with non-local redirection

The redirecting information is also available to the calling subscriber in the case of an outgoing call that is not redirected via his own PBX but via a PISN subscriber, a virtual PISN subscriber or a public network subscriber.

3.6 CLIP / COLP Settings

The following settings affect the CLIP and, by analogy, the COLP, too.

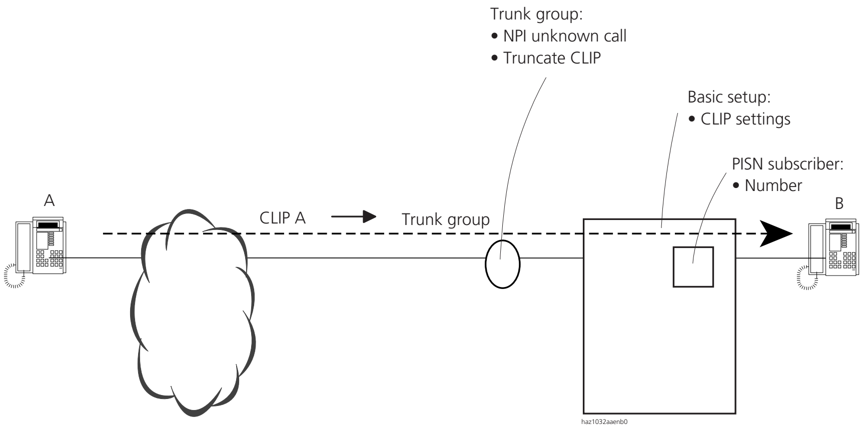


Fig. 2.17: CLIP incoming

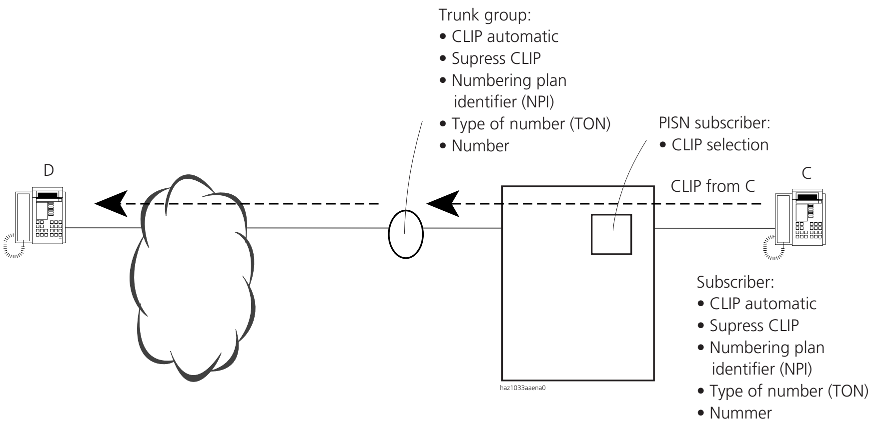


Fig. 2.18: CLIP outgoing

Tab. 2.11: CLIP related settings

Configuration Element	Parameter	Affect on CLIP	
		Incoming	Outgoing
Subscriber	Automatic CLIP Suppress CLIP Numbering plan NPI Type of number TON Number		✓ ✓ ✓ ✓ ✓
PISN Subscriber	Number CLIP selection (normal, subscriber CLIP)	✓	 ✓
Trunk groups	Call NPI unknown Automatic CLIP Suppress CLIP Numbering plan NPI Type of number TON Number Truncate CLIP	✓ ✓	 ✓ ✓ ✓ ✓ ✓
CLIP settings	International prefix Country code National prefix Long-distance code (national code) Display CLIR	✓ ✓ ✓ ✓ ✓	
Numbering Plan	Separate Regional Prefix	✓	✓

Subscriber

Call to the Public Network

Call to the public network with exchange access prefix via a trunk group with "Type of network = public":

If "CLIP automatic = yes", the DDI number will be used as CLIP if the subscriber is himself reachable by incoming calls via the path trunk group → direct dialling plan → CDE. If there is no direct dialling plan or corresponding DDI number, the CLIP number entered in the trunk group will be used instead.

The numbering plan and type of number are always taken from the trunk group.

If "CLIP automatic = no" the configured number is used without any further changes.

Call to a PISN Subscriber

The creation of the CLIP number depends on the configured PISN subscriber. If the PISN subscriber has "CLIP selection = normal", the DDI number is used as CLIP, providing the subscriber is himself reachable by incoming calls via the path trunk group → direct dialling plan → CDE.

If there is no direct dialling plan or corresponding DDI number (which is normally the case), the subscriber's internal call number is used instead.

If the subscriber has the setting "CLIP selection = SC CLIP", the CLIP number is created in the same way as for a call to the public network. This means that a permanently defined CLIP number can also be transmitted in the private network.

Call with Route Selection via Trunk Group with "Network Type = Private"

By analogy with the call to a PISN subscriber with the setting "CLIP selection = normal".

PISN Subscriber

"Number" Setting

The call number entered under "Number" is compared with the CLIP number of an incoming call. If the two numbers match up, the PISN subscriber number is displayed as the CLIP, with "NPI = private" and "TON = level 0".

"CLIP Selection" Setting

See "Call to a PISN Subscriber", page 197.

Trunk groups

"NPI unknown Call" Setting

If a call with "NPI = unknown" is received, it is signalled with the internal or external ringing pattern on the basis of this setting. It is also decided at the same time whether the exchange access prefix (0-) should precede the CLIP number.

"Truncate CLIP" Setting

A digit sequence can be configured here. If the sequence matches the initial digits of the CLIP number received, the digits will be truncated. This setting is normally used to remove any superfluous "0".

"CLIP automatic" Setting

This setting is effective only if "Network type = private" has been set in the trunk group configuration.

If "CLIP automatic = yes", the numbering plan identifier and type of number are left unchanged.

If "CLIP automatic = no", the numbering plan identifier and type of number are taken from the trunk group setting, but not the actual CLIP number. This may be necessary in cases where connected third-party systems do not process numbering plan identifiers and types of number correctly.

Numbering Plan identifier, Type of Number, Number

These settings are used if the CLIP number could not be created automatically. This is the case when there is no suitable DDI number available with a call to the public network.

CLIP/CLIR settings

These settings are used to truncate prefixed access digits so that the CLIP number is as short as possible.

To enable the PBX to interpret CLIP numbers correctly, the system's own regional prefixes need to be entered under "CLIP/CLIR settings":

- International and national prefixes for the PBX locations ("00" and "0" for Switzerland, "00" and "-" for France)
- Country code and long-distance code of the PBX location (for Switzerland "41", for Geneva "22"", see also "Numbering Plan Identifier E.164", page 161).

Displaying CLIR

When CLIR is activated (suppress CLIP) the public network provider will still send a CLIP to special customers, for instance the fire brigade and the police. The CLIR information will, however, include the CLIP (see also "Suppressing CLIP / COLP (CLIR / COLR)", page 188).

In the private leased-line network a CLIP is always sent with an activated CLIR. It is also provided with the CLIR information.

If "Display CLIR = yes", a CLIP with CLIR information is still displayed with incoming calls.

In internal traffic, a suppressed CLIP is always displayed.

Numbering Plan

The CLIP number is prefixed with the regional prefix for outgoing calls to a PISN subscriber or via a trunk group with "Type of network = private".

For incoming calls, the regional prefix is removed from the CLIP number (provided it begins with that digit sequence).

3.7 Examples of CLIP Displays in the PISN

Various scenarios are used in a sample network to illustrate how CLIP displays are handled in a PISN. Fig. 2.19 shows the sample network.

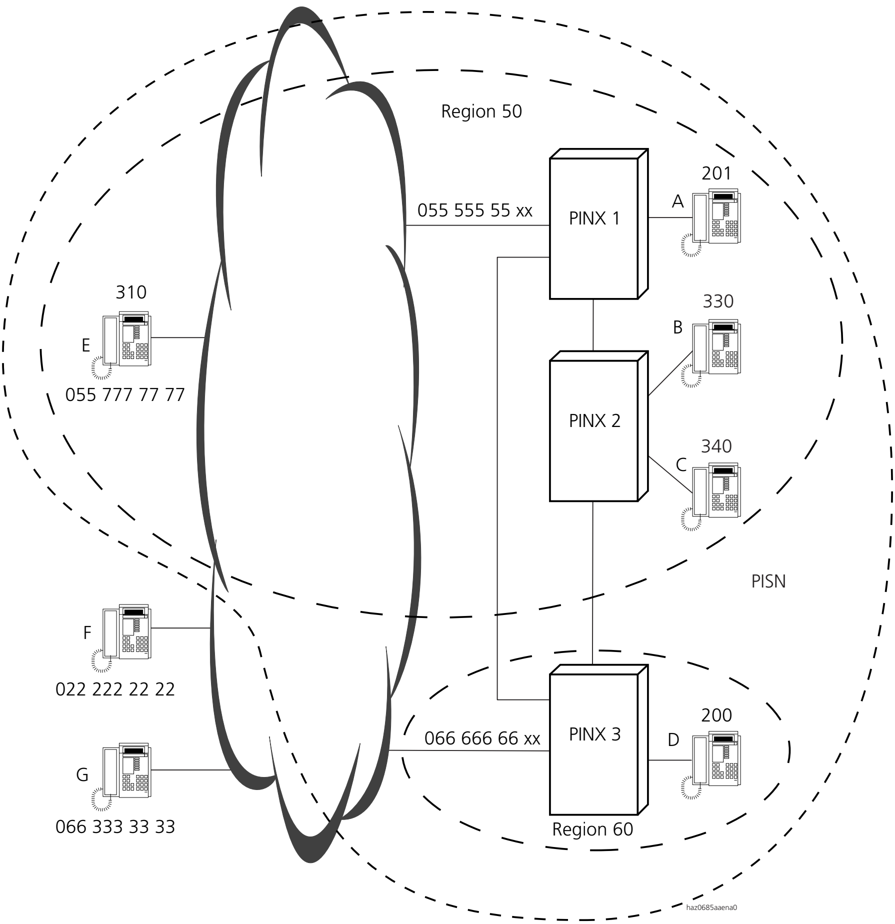


Fig. 2.19: Sample network: PISN with two regions and one virtual network subscriber

3.7.1 PISN-Internal Calls

Ordinary PISN-Internal Call

Subscriber C (340) on PINX 2 calls subscriber A on PINX 1 by a direct route. Both subscribers belong to the same region.

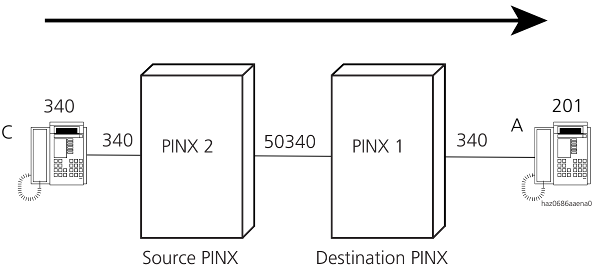


Fig. 2.20: Example 1: Subscriber C calls subscriber A (excerpt from Fig. 2.19)

Tab. 2.12: Example 1: Creating and presenting subscriber C's CLIP number

Step	CLIP number	NPI	TON	Description
1	340	PNP	level 0	Subscriber C → PINX 2 • There is no suitable DDI number.
2	50340	PNP	level 1	PINX 2 → PINX 1
3	340	PNP	level 0	PINX 1 • The system's own regional prefix is deleted • TON is adapted.
4	340			PINX 1 → Subscriber A • Presentation on the system terminal

PISN - Internal Call with Overflow Routing

Subscriber C (340) on PINX 2 calls subscriber A on PINX 1 via PINX 3 as all the available communication channels between PINX 2 and PINX 1 are busy. PINX 3 belongs to Region 60.

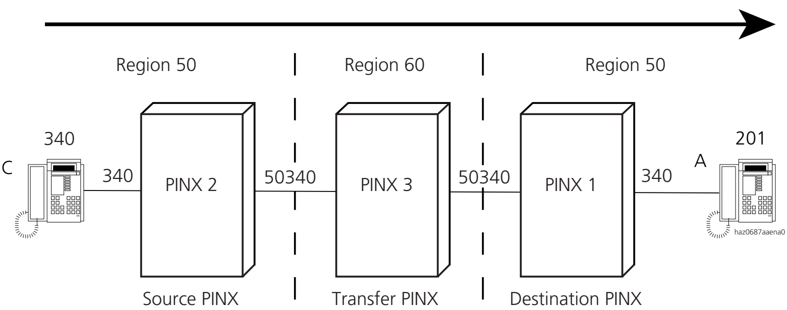


Fig. 2.21: Example 2: Subscriber C calls subscriber A, Overflow Routing (excerpt from Fig. 2.19)

Tab. 2.13: Example 2: Creating and presenting subscriber C's CLIP number

Step	CLIP number	NPI	TON	Description
1	340	PNP	level 0	Subscriber C → PINX 2 • There is no suitable DDI number.
2	50340	PNP	level 1	PINX 2 → PINX 3
3	50340	PNP	level 1	PINX 3 • There is no suitable DDI number.
4	50340	PNP	level 1	PINX 3 → PINX 1
5	340	PNP	level 0	PINX 1 • The system's own regional prefix is deleted • TON is adapted.
6	340			PINX 1 → Subscriber A • Presentation on the system terminal

3.7.2 Outgoing Calls to the Public Network

Call to the Public Network via a Gateway PINX

Subscriber C (340) on PINX 2 calls subscriber F on the public network via PINX 1. PINX 1 has a DDI number for subscriber C (54).

The following CLIP characteristics are set in the trunk group configuration of PINX 1:

- CLIP number = 50
- NPI = unknown
- TON = unknown

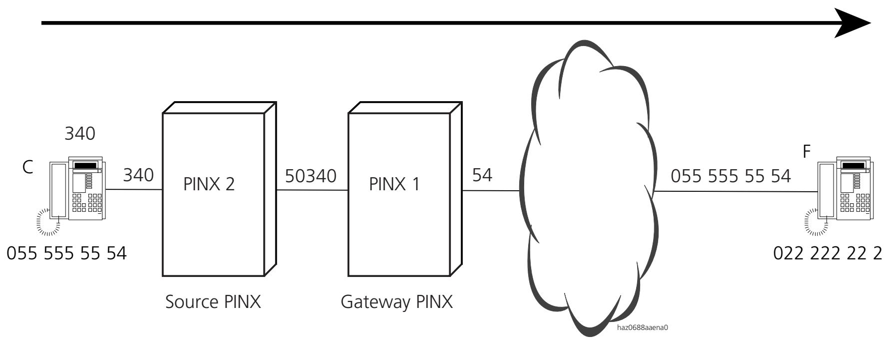


Fig. 2.22: Example 3: Subscriber C calls subscriber F in the public network (excerpt from Fig. 2.19)

Tab. 2.14: Example 3: Creating and presenting subscriber C's CLIP number

Step	CLIP number	NPI	TON	Description
1	340	PNP	level 0	Subscriber C → PINX 2 • There is no suitable DDI number.
2	50340	PNP	level 1	PINX 2 → PINX 1
3	340	PNP	level 0	PINX 1 • The system's own regional prefix is deleted • TON is adapted.
4	54	Unknown	Unknown	PINX 1 → Exchange • There is a suitable DDI number, which is used as a CLIP No. and sent to the public network.
5	055 555 55 54			Exchange → Subscriber F • Presentation on the terminal

Call to the Public Network via a Gateway PINX with Overflow Routing

Subscriber C (340) on PINX 2 calls subscriber F on the public network via PINX 3 as all the available communication channels between PINX 2 and PINX 1 are busy. PINX 3 does not have a DDI number for subscriber C.

The following CLIP characteristics are set in the trunk group configuration of PINX 3:

- CLIP number = 60
- NPI = unknown
- TON = unknown

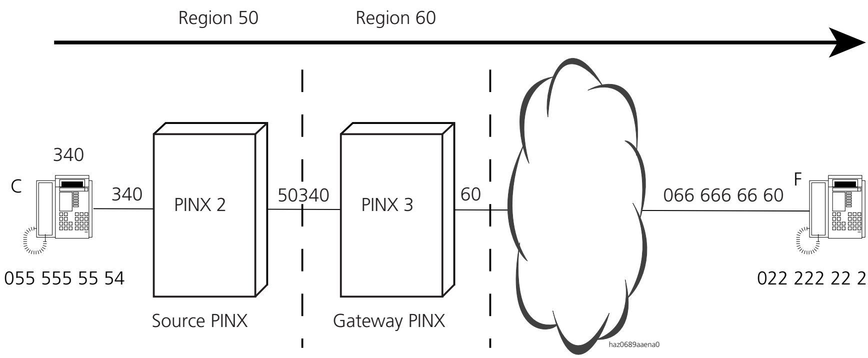


Fig. 2.23: Example 4: Subscriber C calls subscriber F via an alternative path (excerpt from Fig. 2.19)

Tab. 2.15: Example 4: Creating and presenting subscriber C's CLIP number

Step	CLIP number	NPI	TON	Description
1	340	PNP	level 0	Subscriber C → PINX 2 • There is no suitable DDI number.
2	50340	PNP	level 1	PINX 2 → PINX 3
3	50340	PNP	level 1	PINX 3 • There is no suitable DDI number.
4	60	Unknown	Unknown	PINX 3 → Exchange • The CLIP number entered in the trunk group configuration is sent to the public network.
5	066 666 66 60			Exchange → Subscriber F • Presentation on the terminal

Call to the Public Network via a Gateway PINX with Overflow Routing and "Automatic CLIP = no"

Subscriber B (330) on PINX 2 calls subscriber F on the public network via PINX 3 as all the available communication channels between PINX 2 and PINX 1 are busy.

PINX 3 does not have a DDI number for subscriber B.

"Automatic CLIP = no" is set in the configuration subscriber of subscriber B. The CLIP settings of the subscriber configuration are used:

- CLIP number = 55 555 55 53
- NPI = E.164
- TON = national

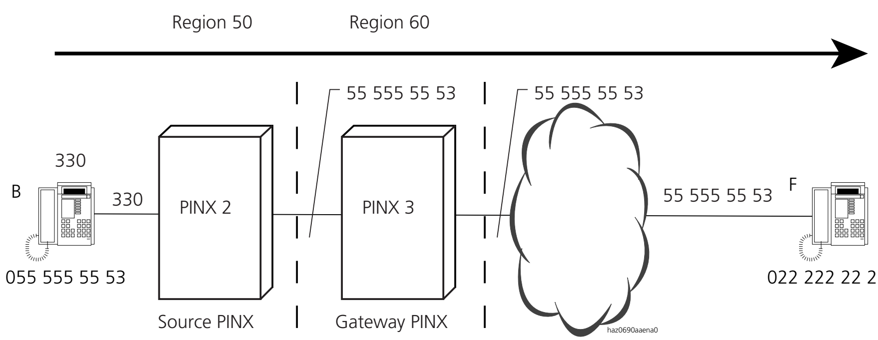


Fig. 2.24: Example 5: Subscriber B calls subscriber F (excerpt from Fig. 2.19)

Tab. 2.16: Example 5: Creating and presenting subscriber B's CLIP number

Step	CLIP number	NPI	TON	Description
1	330	PNP	level 0	Subscriber B → PINX 2 • A suitable DDI number is not searched for.
2	55 555 55 53	E.164	National	PINX 2 → PINX 3
3	55 555 55 53	E.164	National	PINX 3 • CLIP No. is buffered unchanged • A suitable DDI number is not searched for.
4	55 555 55 53	E.164	National	PINX 3 → Exchange • CLIP No. is sent unchanged to the public network.
5a	055 555 55 53			Exchange → Subscriber F • Presentation on the terminal if special arrangement is available (see page 181).
5b	066 666 66 60			Exchange → Subscriber F • Presentation on the terminal if special arrangement is not available (see page 181).

3.7.3 Incoming Calls from the Public Network

Subscriber G on the public network calls subscriber C on PINX 2 via PINX 1. He dialls 055 555 55 54.

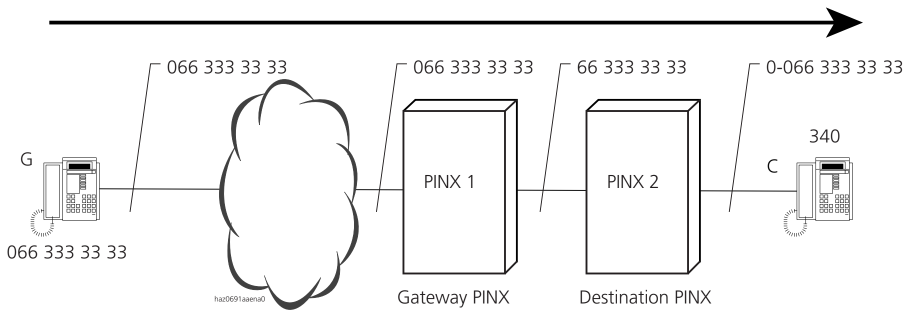


Fig. 2.25: Example 6: Subscriber G calls subscriber C (excerpt from Fig. 2.19)

Tab. 2.17: Example 6: Creating and presenting subscriber G's CLIP number

Step	CLIP number	NPI	TON	Description
1	066 333 33 33	E.164	Unknown	Subscriber G → Exchange → PINX 1
2	66 333 33 33	E.164	National	PINX 1 • Prefix is truncated • TON is set to "national".
3	66 333 33 33	E.164	National	PINX 1 → PINX 2
4	66 333 33 33	E.164	National	PINX 2 • CLIP number is not altered.
5	0-066 333 33 33[1)]			PINX 2 → Subscriber C • Presentation on the system terminal

1) In PINX 3's trunk group configuration 066 666 60 is entered as the master number.

Call from the Public Network with Overflow Routing

Subscriber G on the public network calls subscriber C on PINX 2 via PINX 1 and PINX 3 as all the available communication channels between PINX 2 and PINX 1 are busy. He dials 055 555 55 54.

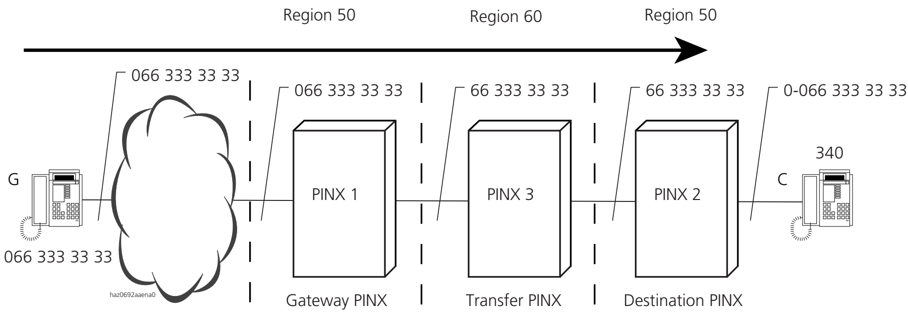


Fig. 2.26: Example 7: Subscriber G calls subscriber C via PINX 3 (excerpt from Fig. 2.19)

Tab. 2.18: Example 7: Creating and presenting subscriber C's CLIP number

Step	CLIP number	NPI	TON	Description
1	066 333 33 33	E.164	Unknown	Subscriber G → Exchange → PINX 1
2	66 333 33 33	E.164	National	PINX 1 • Prefix is truncated • TON is set to "national"
3	66 333 33 33	E.164	National	PINX 1 → PINX 3

Step	CLIP number	NPI	TON	Description
4	333 33 33	E.164	Subscriber	PINX 3 • Long-distance code is truncated as it is the same as the system’s own long-distance code • TON is set to "subscriber"
5	66 333 33 33	E.164	National	PINX 3 → PINX 2
6	66 333 33 33	E.164	National	PINX 2 • CLIP number is not altered.
7	0-066 333 33 33			PINX 2 → Subscriber C • Presentation on the system terminal

Call made by a PISN subscriber in the public network

PISN subscriber E (310) on the public network calls subscriber C on PINX 2 via PINX 1. He dials 055 555 55 54.

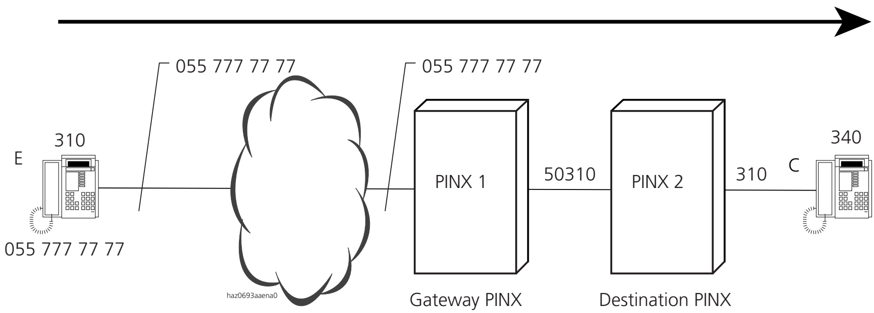


Fig. 2.27: Example 8: Subscriber E calls subscriber C (excerpt from Fig. 2.19)

Tab. 2.19: Example 8: Creating and presenting subscriber E’s CLIP number

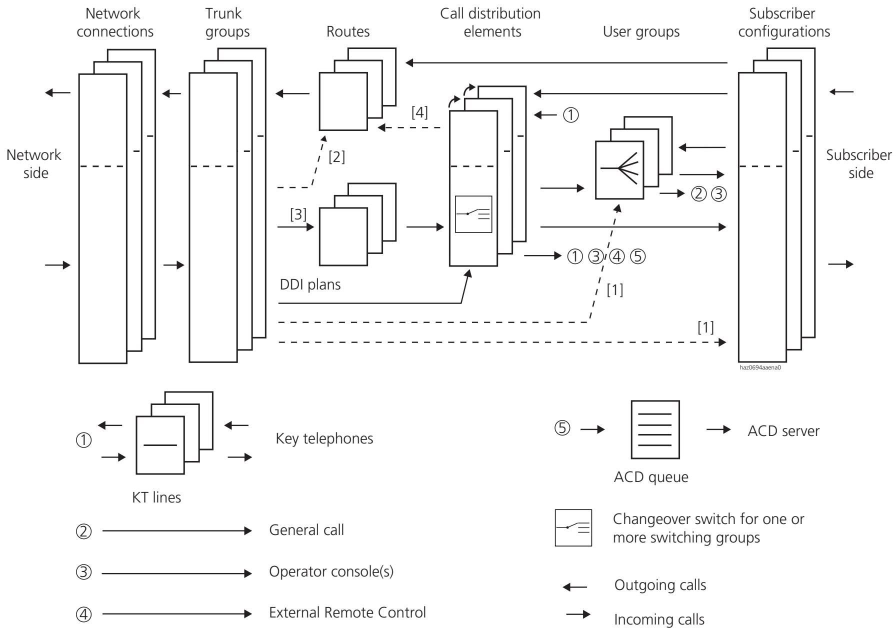
Step	CLIP number	NPI	TON	Description
1	055 777 77 77	E.164	Unknown	Subscriber E → Exchange → PINX 1
2	55 777 77 77	E.164	National	PINX 1 • Prefix is truncated. • TON is set to "national".
3	777 77 77	E.164	Subscriber	• Long-distance code is truncated as it is the same as the system’s own long-distance code. • TON is set to "subscriber".
4	310	PNP	level 0	• CLIP number matches the call number to PISN subscribers: PISN SC No. is used • A suitable DDI number is not found.

Step	CLIP number	NPI	TON	Description
5	50310	PNP	level 1	PINX 1 → PINX 2 • Regional prefix is added and TON is adapted.
6	310	PNP	level 0	PINX 2 • System’s own regional prefix is deleted and TON adapted.
7	310	Unknown	level 0	PINX 2 → Subscriber C • Presentation on the system terminal

4 Routing elements

The role of routing elements is to route and distribute internal, incoming and outgoing calls to destinations. From the system's viewpoint a destination is an interface (e.g. network interface or user-network interface). In this context user groups or subscriber configurations are also routing elements, not destinations.
The settings allocated to a routing element are carried out in the system configuration. This chapter describes the individual routing elements; the interplay between the routing elements is described in "Call Routing", page 267.

Fig. 2.28 shows how all the routing elements relate to one another:



[1] Routing via the numbering plan to one of the elements. Applies only to calls from the permanent network PISN (see page 276)
[2] Routing via a transit route (see page 338) or as [1]. Applies only to calls from the permanent network PISN
[3] Does not apply to calls from the analogue network
[4] Outgoing KT calls

Fig. 2.28: How calls are routed in the system

Network interfaces

Network interfaces provide the access to the PBX from the outside. The settings for the network interfaces are used to specify network-specific characteristics (e.g. point-to-point or point-to-multipoint connection or the distribution of B channel groups at the primary rate access).
As network interfaces are not routing elements per se, they are not discussed further in this chapter. For more detailed information see Part 1, page 60.

Trunk groups

Network interfaces with the same characteristics are grouped together in a trunk group. For each trunk group, for example, it is specified whether the grouped network interfaces are connected to a private network or the public network (see page 212).

Direct Dialling Plans

Direct dialling is used to reach internal subscribers or PISN subscribers directly from the public network. The direct dial portion of an incoming call number is used to link the call with a specific call distribution element (see page 223).

Routes

All outgoing calls are routed to a trunk group via a route. They also include calls routed via the Least Cost Routing function and transit calls in a PISN (see page 219).

Call Distribution Elements

Call distribution elements are used to route a call to a destination or combination of destinations. The destination (or combination of destinations) can vary depending on the allocated switch position. If the original destination is busy or does not answer after a certain time, calls can be routed to alternative destinations (see page 226).

Switching Groups

Certain destinations and functions are selected depending on the switching position of a switching group. Each switch group has three switch positions, which are used for example for Day, Night and Weekend (see page 233).

User Groups

In a user group incoming and internal calls are routed to a group of internal destinations in accordance with a pre-configured call distribution pattern (see page 236).

Subscriber Configuration

All the subscriber-specific settings are grouped together in the subscriber configuration. This chapter deals exclusively with settings that are specific to routing and identification (see page 246).

Operator Console

The system has one switching station, which is defined under the name "Operator Console" in the internal numbering plan. Several Operator Consoles can be operated in parallel (see page 248).

General Bell

Calls with the general bell as destination can be signalled via an external supplement (see page 253).

Key Telephones

Many of the system terminals can be operated as key telephones with line keys. The line keys are linked to a call distribution element via "KT lines" (see page 253).

ACD Server

With an ACD application on the third-party CTI interface (ACD server), routing control can be shifted from the PBX to the ACD server (see page 264).

4.1 Trunk groups

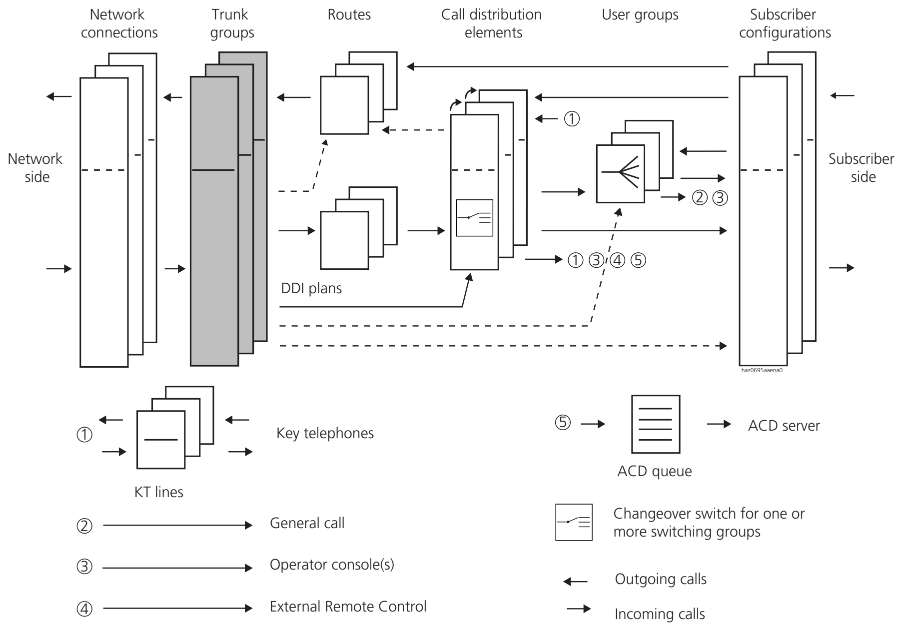


Fig. 2.29: Trunk groups in relation to the other routing elements

Network interfaces with the same characteristics are grouped together in a trunk group. For example, it is specified whether the network interfaces allocated to a trunk group are connected to a private network or the public network.

The trunk group is the key element for call traffic with the network. The trunk group configuration is allocated important routing and identification functions, essentially for incoming traffic. A number of settings are used for setting up special network configurations, for instance the optimum integration of PINXs by third-party manufacturers. The initialization values of these settings are such that they do not need to be adapted any further for conventional configurations.

4.1.1 Trunk Groups of Network Interfaces

General Rules and Settings

A network interface can only be assigned to a single trunk group.

A trunk group contains either analogue or digital network interfaces (the network interface entered first is decisive).

The digital network interfaces of a trunk group lead either

- to the permanently networked PISN if "network type = private" is set,
- to the public network if "network type = public" is set.

The following rules apply to the setting of the transmission protocol ("Protocol") for the network interfaces of a trunk group:

- Trunk groups with "type of network = private" are usually set to the QSIG protocol.
  PSSI (QSIG) set.
- Trunk groups with "Network type = public" are set to protocol DSS1.

Tip:
It is advisable to enter network interfaces that have the same destination in the same trunk group, for example to set up one trunk group for the public network, one trunk group for PINX 1 and one trunk group for PINX 2, etc. (see also planning suggestions in Part 3).

Initialization Settings

Newly set up digital network interfaces are automatically entered in trunk group 1.

Trunk group 1 is set to "Network type = public" and "Protocol = DSS1".

Newly set up analogue network interfaces are automatically entered in trunk group 2.

Trunk group 2 is set to "Network type = public".

Seizure Sequence for Outgoing Calls

The system will first try and seize the network interface that was entered last (right at the end of the list). If for whatever reason this interface is not available, it will then attempt to seize the second last interface, then the third last, etc. (see also Fig. 2.32).

This is repeated for each outgoing call using the same principle.
This means the call charges tend to accumulate on the network interface entered last.

S Interface as the Network Interface

An S interface set as "external" is also classified as a network interface and can be assigned to a trunk group.

Note:
If an S interface is reconfigured within a trunk group (to ETSI or V2), it is no longer a network interface and is removed from the trunk group.

B Channel Groups

The two user-information channels of a basic access and the 30 user information channels of a primary rate access can be divided into 2 and 4 B channel groups ("B channel list" setting[1)]). This classification is carried out only if, for example, not all the B channels of the primary rate access are available. B channel groups can be separately allocated to a trunk group, for example the primary rate access on network interface 3.17 can be entered as follows:

- "3.17": All four B-channel groups are allocated to the trunk group.
- "3.17/2": Only B-channel group 2 is allocated to the trunk group.

Initialization value: All B channels are in B-channel group 1.

Planning tips:

- B channels can only be grouped in sequence (e.g. channel group 1 contains B channels 1 to 6.
- A B channel can only be allocated to one channel group.
- If the B-channel groups of a primary rate access are spread among different trunk groups, the same protocol must be set for all trunk groups.

1) Dividing up B channel groups is not supported by all network providers.

Configuration:

Once a trunk group contains a B-channel group, the trunk group's protocol can no longer be changed. For this reason it is important to proceed using the following configuration stages:

1. Enter the network interface of the basic or primary rate access in the first trunk group (e.g. "3.17").
2. Set the trunk group protocol (e.g. "DSS1").
3. Divide the B channels of the basic or primary rate access into B-channel groups. The network interface already entered is changed to B channel group 1 (entry changes to 3.17/1).
4. Enter the other B-channel groups in the required trunk groups. The protocol of the first trunk group is set automatically.

Subscriber Line Group (SLG) in ISDN

Digital outside lines that are to have the same traffic characteristics can be grouped into subscriber line groups (SLG) in the public network (e.g. several basic accesses with the same DDI block).
An SLG must also be recreated in the PBX. For this the network connections of the exchange lines of an SLG must be allocated to the same trunk group (see Fig. 2.30).

A subscriber line group can consist of basic accesses, primary rate accesses or individual B-channel groups of primary rate accesses (also mixed).

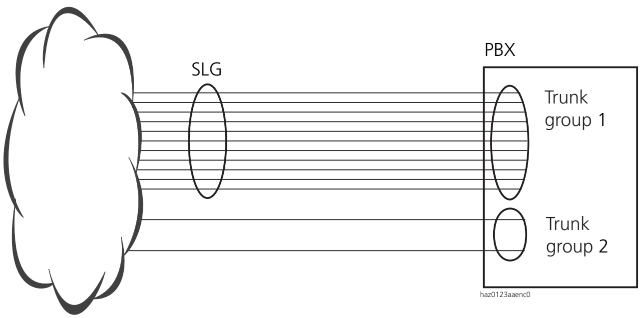


Fig. 2.30: Recreating an SLG in the PBX

4.1.2 Routing Functions of the Trunk Group for Incoming Calls

The following incoming routing functions are assigned to the trunk group:

- Restricting the number of calls incoming simultaneously per trunk group
- Routing a call to one of the following elements:
  - Direct dialling plan (see page 223)
  - Call distribution element (see page 226)
  - Destination of the internal numbering plan (see page 276)
- Adapting the numbering plan identifier of an incoming call

Restricting the Number of Calls Incoming Simultaneously per Trunk Group

Once the set limit is reached ("Incoming connections" setting), no more calls are routed via the trunk group. This is signalled to the caller by means of the congestion tone.

After an initialization the limit is set to approx. 80% of the available B channels.

4.1.3 Trunk Group Identification Functions

The CLIP numbers of outgoing calls can be influenced by the following settings in the trunk group configuration:

- Automatic CLIP
- Numbering plan identifier (NPI)
- Type of number (TON)
- Permanent CLIP number

For more details see "CLIP with Outgoing Calls", page 187and following.

Truncate CLIP

See "Trunk groups", page 197.

4.1.4 Other Trunk Group Functions and Settings

Name of the trunk group

"Name" is used to assign a name to each trunk group The name's main purpose is to provide orientation It is displayed on some system terminals whenever an outgoing connection is set up.

Tip:
It is a good idea to name trunk groups according to the origin of their lines (e.g. "Public ISDN", "Analogue", "Leased Line Geneva", etc.). This ensures greater clarity during configuration work.

Generating a Ring-back Tone

The system can use the settings "Outgoing call: Ring-back tone" and "Incoming call: Ring-back tone" to control within certain limitations the generation of the ring-back tone on digital trunk connections. These settings do not have to be altered in normal PBX operation.

- In the case of a stand-alone PBX on the public network the ring-back tone is supplied by the local exchange and does not have to be generated by the PBX.

- In a PISN with QSIG networking the ring-back tone always has to be generated in the destination PINX. The setting "Incoming call: Ring-back tone = Generate" is permanent and cannot be altered.

Here are two applications in which the settings do have to be adapted:

- In a PISN with networking via DSS1 protocol the ring-back tone is normally also generated in the destination PINX ("Incoming call: Ring-back tone = Generate"). There are however exceptions (e.g. PBX integrated in CENTREX) where the ring-back tone does not have to be generated internally. In such cases select the setting "Incoming call, ring-back tone = Do not generate".
- It is possible that the destination does not generate a ring-back tone. (e.g. external IP gateways). In such cases it is possible to generate the ring-back tone locally. To do so, select the setting "Outgoing call, ring-back tone = Generate".

Note:
If several PINXs are cascaded, generate the ring-back tone only once whenever possible and as close to the called party as possible.

Rerouting in the Exchange

The "Rerouting in the exchange" setting can be used to specify whether the system is authorized to place exchange-to-exchange connections to the exchange via the trunk group's exchange lines. If the exchange lines of two trunk groups are involved, this connection privilege must be granted to both trunk groups (see also "Transferring Call Forwarding Unconditional to the Exchange", page 328).

This setting is possible only for trunk groups with protocol = DSS1.

Hold and Three-party Conference in the Exchange

For three-party conference in the exchange see "Three-Party Connections in the Exchange", page 331.

"Truncate DDI" Setting

See "Direct Dialling Plan (DDI plan)", page 223.

Other Trunk Group-related Subjects:
Network interfaces, Route, Incoming traffic, Outgoing traffic, Traffic in the PISN, Identification elements.

4.2 Route

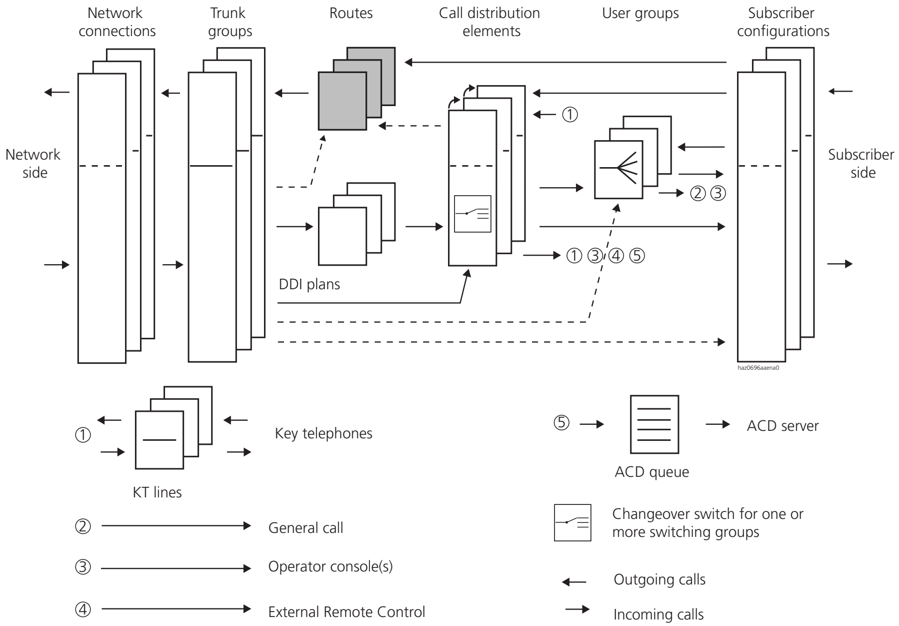


Fig. 2.31: Routes in relation to the other routing elements

The route function applies only to outgoing calls.

A route determines the direction of outgoing calls through allocation to trunk groups. All outgoing calls are routed via a route to one or more trunk groups. They also include calls routed via the Least Cost Routing function and transit calls in a PISN. Normally a separate route is set up for each PINX

The route elements can be allocated internal call numbers in the internal numbering plan. In this way a route element can be selected directly (route selection, see page 292).

"Name" is used to assign a name to each route. The main purpose of the name is to provide orientation; it does not have a routing function.

Tip:
It is advisable to name the routes according to their function. For example "Transit Routing", "Remote Alarming", "to PINX 3", etc. It makes the configuration work all the clearer.

4.2.1 The Route's Routing Functions

The route is allocated the following outgoing routing functions:

- Routing an outgoing call to one or more trunk groups
- Restricting the number of calls outgoing simultaneously
- Polling an external digit barring
- Deleting the exchange access prefix
- Adding a prefix to the call number (where required)
- Specifying numbering plan identifier NPI
- Specifying how many digits need to be dialled before a call is set up

4.2.2 Routing an Outgoing Call to a Trunk Group

Up to 8 trunk groups can be entered for each route ("Trunk group" setting). The system will first try to seize the network interfaces of the trunk group that was entered last. If these interfaces do not come first, it will then attempt to seize the network interfaces of the second trunk group, etc.

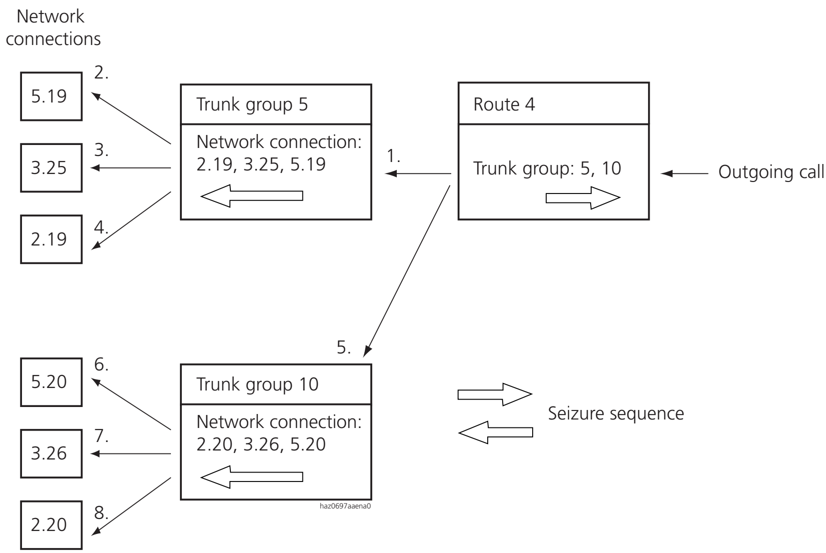


Fig. 2.32: Seizure sequence for network connections in the case of an outgoing call

If both analogue and digital network interfaces are being used, one trunk group has to be entered in each route for the analogue interfaces and one trunk group for the digital interfaces since a trunk group can only contain either analogue or digital interfaces.

Initialization Settings

- After initialization, route 1 is allocated trunk groups 1 and 2.
- After initialization, route 3 is allocated trunk group 1 (route for remote alarming).
- On 2025 and 2045 systems all routes are allocated the numbers from 170 upwards in the numbering plan.
- On the 2065 system the first 30 routes are allocated numbers from 170 upwards in the numbering plan.

4.2.3 Other Routing Functions for Outgoing Calls

Restricting Calls Outgoing Simultaneously

The "Outgoing connections" setting is used to specify the number of outgoing calls that are possible simultaneously. Once the limit has been reached, subscribers can no longer make outgoing calls with the allocation of this route. This is signalled by means of the congestion tone.

Activating / Deactivating External Digit Barring

Normally an outgoing call is compared with the external digit barring allocated in the subscriber configuration.

The "digit barring, ext" setting is used to deactivate the external digit barring for each route. This is useful when a route is set up for calls into the private leased-line network.

Deleting the exchange access prefix

If the call number of an outgoing call has an exchange access prefix, it will be truncated before the call is forwarded.

Adding a Prefix to the Call Number

"Send access code" is used to define a prefix which is added to a call number (which no longer has an exchange access prefix).

The prefix can be used to transmit a call to the public network via a third-party PINX by specifying a route number as the exchange access prefix for the gateway PINX..

Specifying the Numbering Plan Identifier NPI and the Type of Number TON

The call number of an outgoing call is assigned the NPI defined under "Numbering plan identifier NPI".

- For routes that are used for routing outgoing calls whose end destination is in the public network, set "E.164".
- For routes that are used for routing outgoing calls via dedicated lines with end destination in the PISN, set "PNP".

"TON = unknown" is always assigned as the type of number. This cannot be modified in AIMS.

Send Delay

Send delay is used to specify how many digits need to be dialled before a call is set up. The dial tone will be supplied by the PBX as long as the line is not seized. This setting is useful in the following situations:

- When calls are routed to the public network via third-party PINXs
- When the destination PBX can only evaluate whole call numbers (Overlap Receiving not supported)
- To save line resources under heavy traffic conditions

Other Subjects Related to Routes:
Trunk group, Call distribution, Subscriber configuration, Operator Consoles, key telephone, Outgoing traffic, Least Cost Routing, Traffic in the PISN, Numbering plan.

4.3 Direct Dialling Plan (DDI plan)

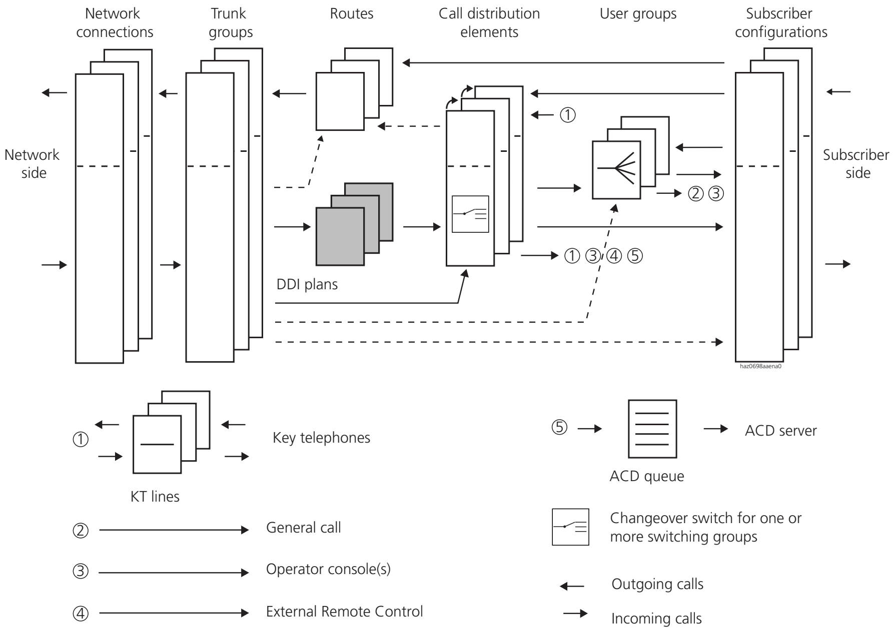


Fig. 2.33: Direct dialling plans in relation to the other routing elements

Direct dialling is used to reach internal subscribers directly from the public network or from another PINX. The incoming call is linked with a call distribution element on the basis of the call number's direct dial portion.

Within a direct dialling plan, number ranges are created by agreement with the public network operator; these numbers match up with the anticipated direct dial portions of the call numbers. In a three-digit direct dialling plan, for example, number ranges of 300…399 and 500…549 are created.

Depending on the country in which the PBX is operated, the ISDN public exchange may send the complete call number or only a part of it. If the complete call number is sent, the digits that do not form part of the direct dialling number can be truncated starting from the left using the setting "Truncate DDI" in the trunk group configuration.

Several Direct Dialling Plans per PBX / PINX

Several direct dialling plans are available. This ensures that the same subscriber can be reached from the outside via different network accesses and that the correct CLIP is also transmitted in outgoing traffic.

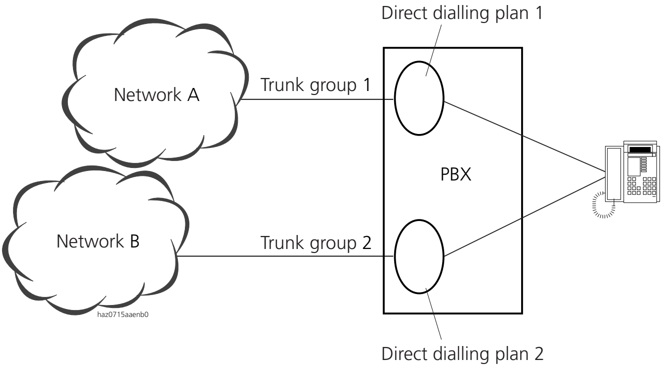


Fig. 2.34: Several Direct Dialling Plans per PBX / PINX

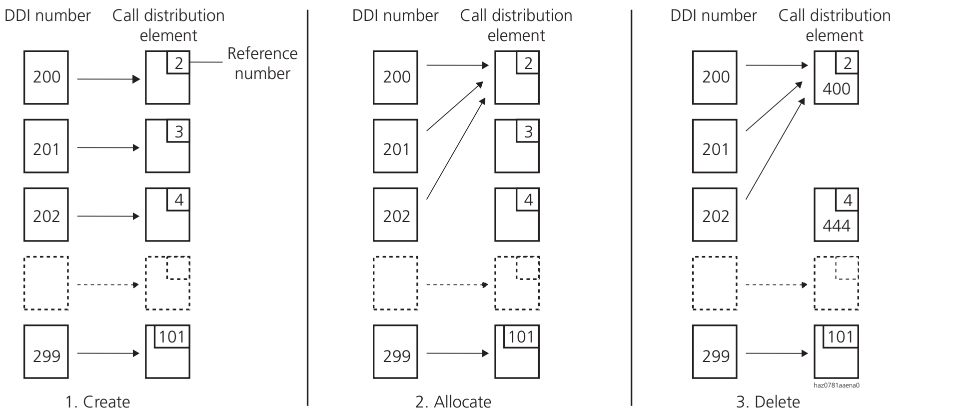
Tip:
Use a separate direct dialling plan for each individual network access to the public network (e.g. for different network providers, point-to-point / point-to-multipoint connections, different SLGs or different direct dialling ranges).

Direct Dialling Plans in the Private Leased-line Network

Direct dialling plans can also be used in the private leased-line network. This is the case in particular if incoming calls from the private leased-line network are to be routed depending on the switching position (see page 357).

Linking a Direct Dial Number with a Call Distribution Element

Direct dial numbers are created as blocks with 1 to several numbers. When the number range is created each direct dial number is automatically linked with a call distribution element. A call distribution element can also be subsequently allocated to several numbers.



1. When direct dial numbers are created, call distribution elements are assigned automatically
2. Several direct dial numbers can be allocated to one call distribution element.
3. For performance reasons unused call distribution elements should be deleted.

Fig. 2.35: Linking direct dial numbers with call distribution elements

Destinations can be allocated to the linked call distribution elements as soon as they are created. Various options can be selected using the setting "Link matching subscribers" (Tab. 2.20).

Tab. 2.20: Allocating CDE destinations using the setting "Link matching subscribers"

Parameter value	Result
No	A common call distribution destination can be allocated to all the DDI numbers, depending on the switch position. Initialization value: UG 16. Initialization value: UG 16
yes, do not create non-matching	A DDI number will only be created if an internal subscriber with the same number already exists. The subscriber will be allocated as the call distribution destination , whatever the switch position.
yes, create non-matching also	All the DDI numbers of the new block will be created. If internal subscribers with the same numbers exist they will be allocated as call distribution destination. A common call distribution destination can be allocated to all the other DDI numbers, depending on the switch position. Initialization value: UG 16

The system only provides direct dialling plans for digital network interfaces.

Other Subjects Related to Direct Dialling Plans:
Trunk group, Call distribution, Incoming traffic, Traffic in the PISN, Identification elements, Numbering plan

4.4 Call Distribution Element (CDE)

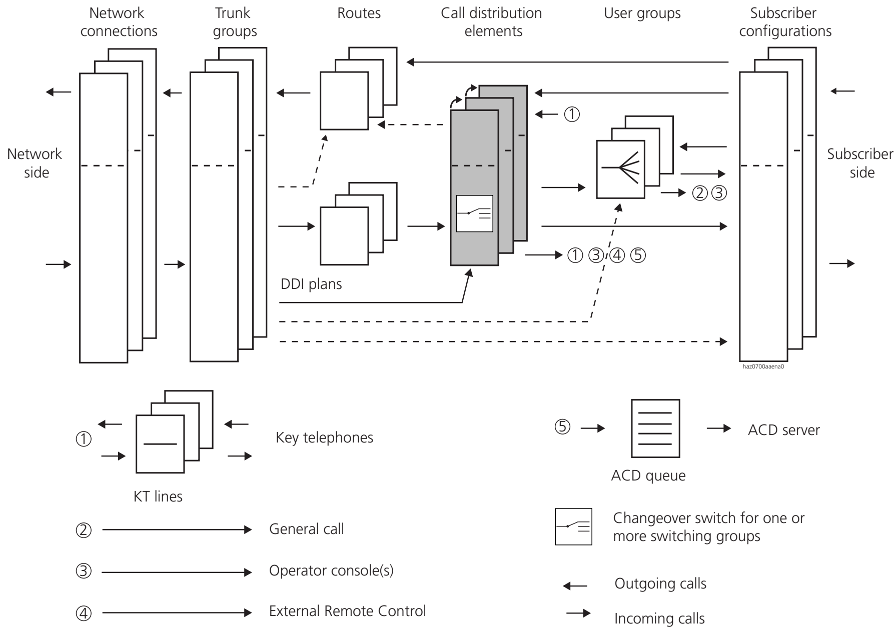


Fig. 2.36: Call distribution elements in relation to the other routing elements

Call distribution elements are used to route an incoming call to an individual destination or to a combination of destinations.

Each call distribution element is assigned a switch group. The destinations can be specified differently for all three switch positions of the switch group.

Each call distribution element can be linked with two other call distribution elements for the routing to alternative destinations if either the original destination is busy or the call is not answered.

A call distribution element can be addressed both internally and externally. It can route a call to an internal or an external destination.

Call distribution elements can be allocated call numbers in the numbering plan. Internal calls can then be routed to a call distribution element by selecting one of these numbers (but not with name selection).

Restrictions:

- Call forwarding unconditional and call forwarding on no reply cannot be applied to a call distribution element.
- The features Call waiting / Intrusion and automatic callback cannot be activated on call distribution elements.
- A call distribution element cannot be stored under a team key.
- In addition a call distribution element cannot be part of a pre-configured conference or of a user group.
- A call distribution element cannot be called using name selection.

Tip:
To be able to dial a call distribution element using name selection, you can use an abbreviated dialling number with the stored call number of the call distribution element.

4.4.1 Call destination

With the destination information of a call distribution element an internal call or an external incoming call can be routed to individual destinations or combinations of destinations.

Individual Destinations

A call is routed is to one of the following destinations:

- SC: Subscriber (internal subscribers or PISN subscribers)
- UG: User Group (see page 236)
- KT: KT line (see page 253)
- OC: Switching centre (see page 248)
- ACD: ACD queue (see page 264)
- ERC: External remote control (see page 544)

Multiple destinations

Calls can be routed to the following multiple destinations

- SC+UG
- SC+KT
- UG+KT

- SC+UG, busy
- SC+KT, busy

If the first destination is busy with multiple destinations busy, the second is not called and the caller obtains the busy tone.

The destinations are defined for each of the three switch positions of the selected switching group (e.g. for Day, Night, Weekend). Other destinations can be defined for each position ("Switching group" setting, "Position 1", "Position 2" and "Position 3").

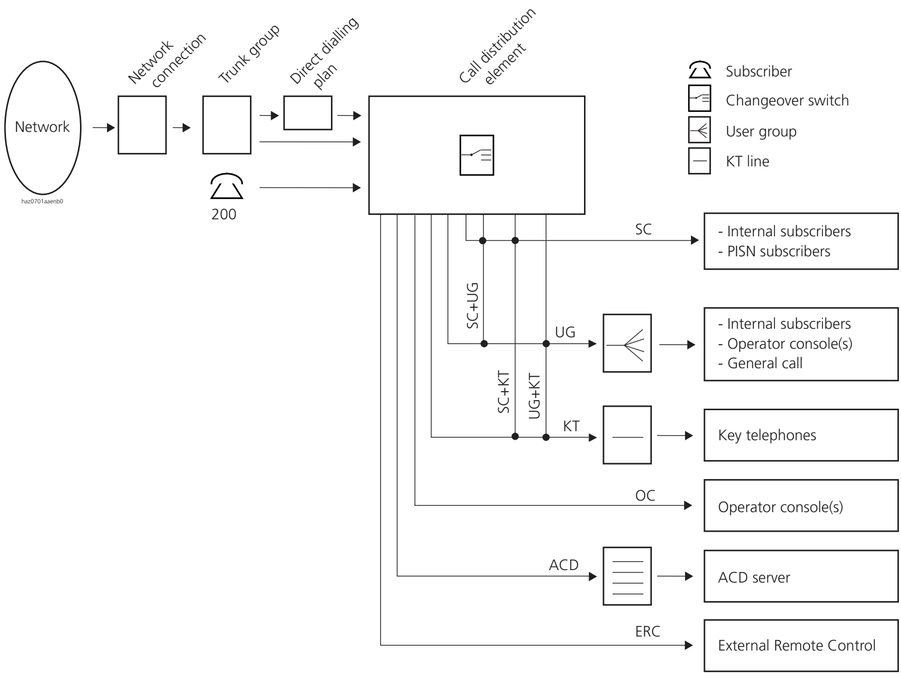


Fig. 2.37: Destinations of the call distribution element

Alternative Destinations

A call distribution element can be linked with two other call distribution elements for the routing to alternative destinations:

- One of the call distribution elements is used for the routing to alternative destinations if a call at the original destination is not answered.
- The other call distribution element is used for the routing to alternative destinations if the original destination is busy.

Alternative Destination if no Answer

If at the original destination the call is neither answered nor forwarded within a configurable period of time ("CDE forwarding time" setting), it is routed to the call distribution element entered under "CDE if no answer". The original destination will then stop ringing

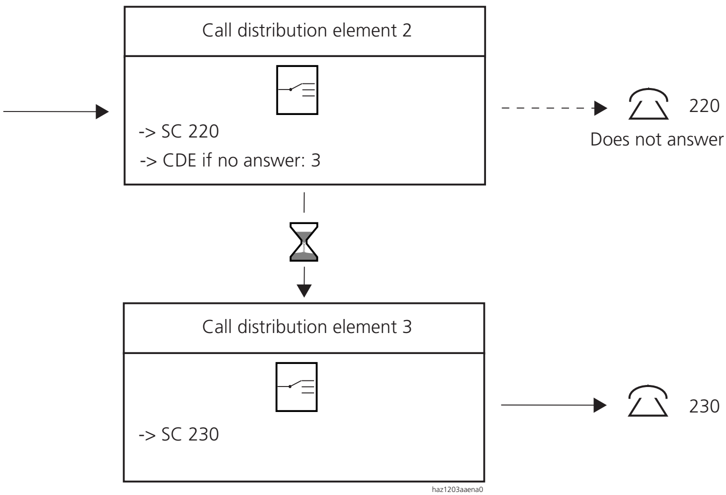


Fig. 2.38: Routing via CDE if no answer

If the call is not answered at the alternative destination either, it will be routed to another call distribution element if such an element has been entered under "CDE if no answer".

If the alternative destination is busy, the call is not forwarded.

The CDE forwarding time can be set individually for each call distribution element.

Note:
If a CDE destination is also defined under "Default CF if no answer" in the subscriber configuration, the call will be redirected after the internal or external delay configured there (see "Default Call Forwarding per Subscriber", page 277).

Courtesy (announcement prior to answering)

A previously activated Courtesy will remain activated if the call is routed to the alternative destination. Courtesy is not reactivated at the next CDE.

Alternative Destination is Busy

If the original destination is busy, the call is routed to the call distribution element entered under "CDE if busy". If the alternative destination is also busy, the call is routed to the next alternative destination -- if such a destination has been configured. This process can be repeated up to the fifth call distribution element. If the destination of the fifth call distribution element is also busy, the caller will obtain the busy tone.

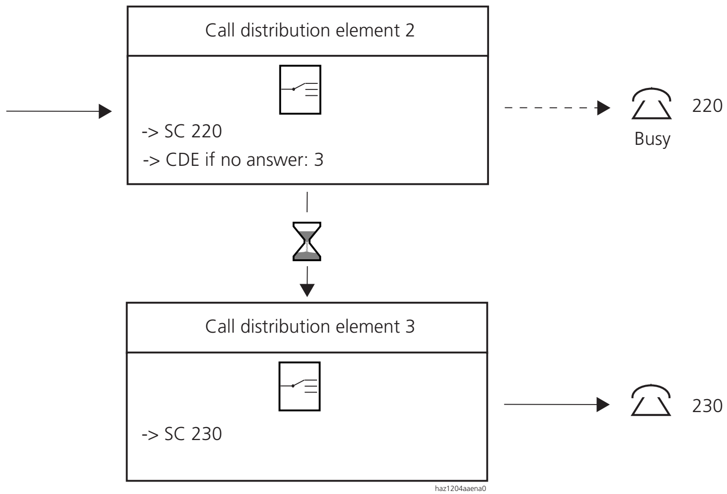


Fig. 2.39: Routing to an alternative destination if the original destination is busy

Note:
There is little point in using "CDE if busy" together with the destination combinations "Subscriber and user group, busy" and "Subscriber and KT line, busy".

Application Example of an Overflow

Implementing an overflow from a busy user group (e.g. Purchasing group) to another user group (e.g. Customer Service group).

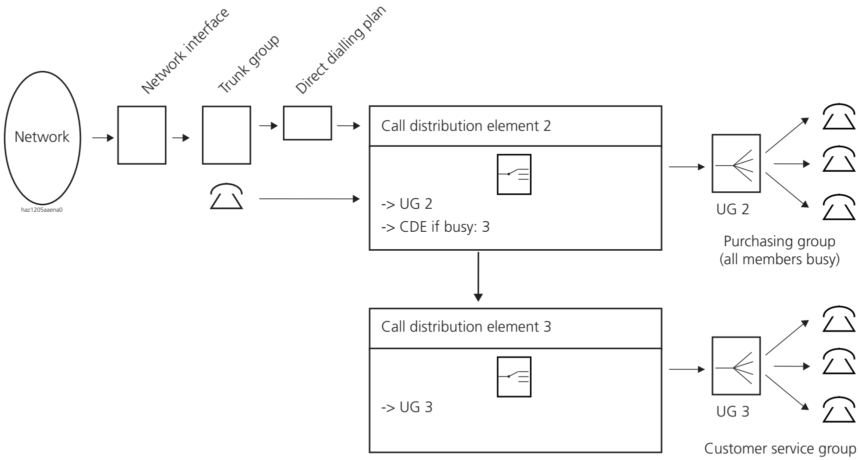


Fig. 2.40: Application example of the configuration of an alternative destination if busy

4.4.2 Routing Functions for Incoming Calls

Call distribution is allocated the following incoming routing functions:

- Routing a call to a destination, depending on the position of the allocated switching group (see "Call destination", page 227).
- Routing a call to an alternative destination if the original destination is either busy or if the call goes unanswered (see "Alternative Destinations", page 229).
- Restricting the number of calls incoming simultaneously on each call distribution element ("Incoming connections" setting) As soon as this limit is exceeded, any subsequent caller will obtain busy, provided no alternative destination "CDE if busy" has been defined.

- Routing a call to data service destinations:
  Data service destinations can be configured for each call distribution element (see "Data services", page 353).
- Routing a call to external remote control:
  This destination is normally allocated only once in every system to enable access to external remote control via a DDI number. (See "Remote controlling features from outside the system", page 544.)

4.4.3 Routing Functions with Outgoing Calls

Outgoing calls via the line keys of a key telephone are routed via the route entered under "KT route" (see "Key Telephones", page 253).

4.4.4 Other Functions and Settings of the CDE

Name

"Name" is used to provide each call distribution element with a name. The name is used for identification purposes.

- With incoming calls it is displayed on the system terminal.
- With outgoing calls via KT lines it is provided as CNIP.

The name cannot be used for dialling by name.

Displaying DDI

With incoming calls the direct dialling number can also be displayed instead of the name of the call distribution ("DDI instead of CDE name = yes"). This is needed for CTI applications in particular.

Activating / Deactivating Incoming Call Logging (ICL)

Incoming call logging can be activated or deactivated for each call distribution element (see "Call logging for incoming calls (ICL)", page 379).

Specifying the Company Configuration

The "Company" setting specifies whether or not the call distribution element is to be used in a two-company configuration (see "Two-company system", page 250).

Courtesy (announcement prior to answering)

A Courtesy group can be allocated to each call distribution element; likewise Courtesy can be deactivated for each element (see "Courtesy Service (announcement prior to answering)", page 523).

KT Cost Centre

Call charges for calls via the KT lines of a call distribution element are logged at the cost centre entered under "KT cost centre" (see also "Outgoing Calls via a KT Line", page 259).

Other Subjects Relating to Call Distribution:
Trunk group, Direct dialling plan, User group, Key telephones, Subscriber configuration, Internal traffic, Incoming traffic, Outgoing traffic, Traffic in the PISN, Switching groups, Numbering plan.

4.5 Switch Groups

With the aid of the switch groups the routing configuration for the system can be conveniently adapted to the time and situation-related requirements of the customer. This means for example that calls during the day can be routed differently to calls at night, or differently at times with a high call volume to times with a low call volume (e.g. at radio stations or in telemarketing.

Certain destinations and functions are selected depending on the switching position of a switching group. Each switching group has three switching positions. The switch position can be used for example for Day, Night and Weekend. There are nine switching groups. The switching groups have changeover switches for

- Routing incoming calls to internal destinations in a call distribution element
- Routing incoming calls to a Courtesy destination in a call distribution element

Switching group 1 also has changeover switches for

- Allocating an external digit barring for each internal subscriber
- Allocating an internal digit barring for each internal subscriber
- Routing an emergency call (outgoing)
- Allocating an internal destination for the door bell
- Switch groups 2 to 9 only have a changeover function for the routing of incoming calls

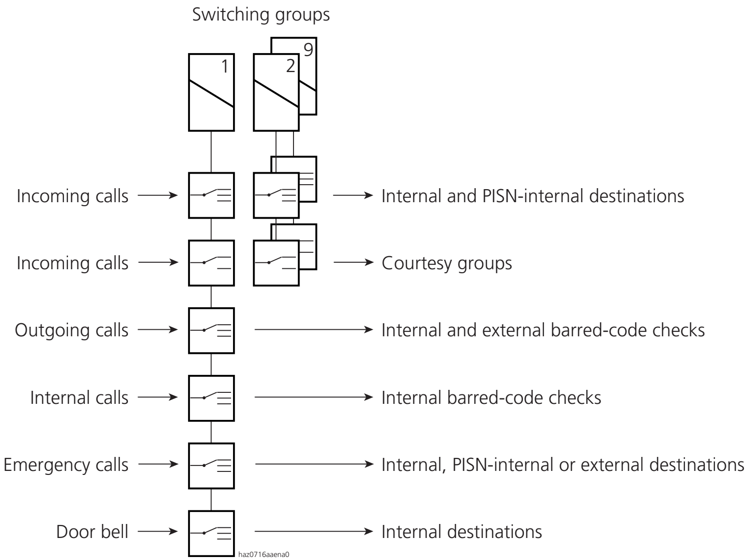


Fig. 2.41: Switching groups and how their changeover switches are used

The choice of switch group and the assignment of the switch positions are carried out in the appropriate menus of the system configuration. After initialization the changeover switches are allocated throughout to switching group 1.

Switching groups are selected via menu selection or by dialling */# procedures on a terminal (see "Switching switch groups", page 535). The relevant authorization can be regulated individually for each internal subscriber. ("Changeover authorization" setting). Digit barrings can also be used to limit changeover authorizations to individual switching groups.

Switching group 1 can also be selected via the ports of the control inputs (see "Switch group interface", page 813). Selection via the control inputs takes precedence over selection via */# procedures. This means that the */# procedure cannot be executed as long as a signal is imposed at the control inputs.

Application Example for Switch Groups

If the secretary is the last person to leave the office at 6.30 p.m., she activates the night service using the menu selection on the Office 45 or an external switch. The PBX will then respond as follows:

- From this moment onwards external calls to the customer service number will be diverted to a telephone answering machine.
- Calls to the main numbers will be informed of the office hours using Courtesy.
- The DDI numbers of office workers will be routed to the Voice Mail service.
- While dialling out will not be permitted in principle, emergency numbers are enabled.

To achieve the above, the following allocations were made for switch position 2 ("Night") of switch group 1 in the system configuration:

- All customer service DDI numbers are routed to the internal number of the telephone answering machine in the call distribution elements.
- The main call number is allocated Courtesy group 1 in the call distribution element. (Courtesy group 1 must be activated).
- All office worker DDI numbers are routed in the call distribution elements to UG 17 (System 2025 / 2045) or UG 25 (System 2065) in which Voice Mail is located.

As the subscriber-specific allocation of the digit barring is also switch-related, they need to be adapted accordingly.

Other Subjects Related to Switching Groups:
Call distribution, Subscriber configuration, Operating switching groups.

4.6 User Group

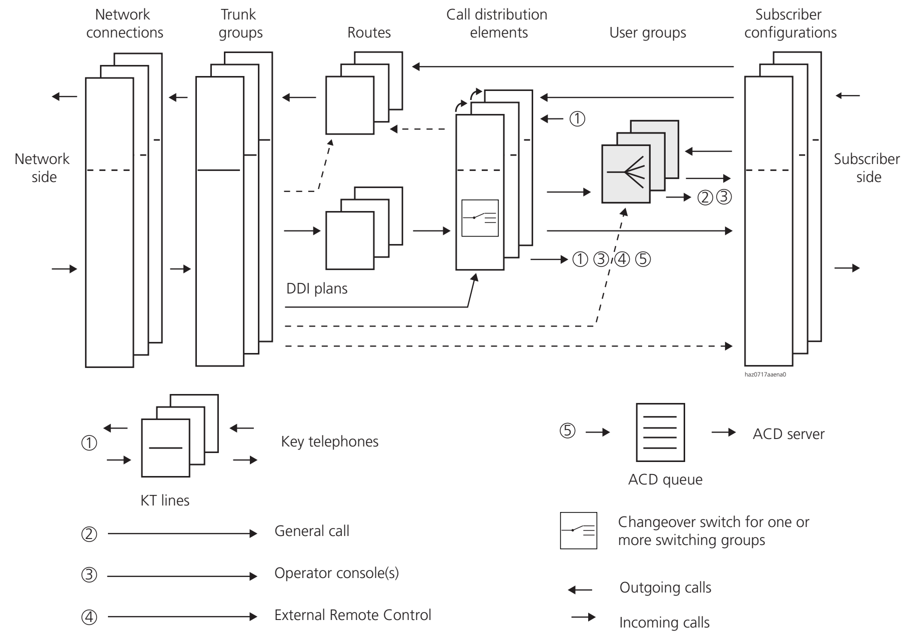


Fig. 2.42: User groups in relation to the other routing elements

In a user group incoming and internal calls are routed to a group of internal destinations in accordance with a pre-configured call distribution pattern.

Incoming Calls

User groups are selected by means of their call numbers or names (name selection). The call numbers of user groups are a separate category of the numbering plan.

CFU or CFNR cannot be made to user groups (except for user groups with special functions and user groups configured as "large").

Outgoing Calls

User groups do not affect outgoing routing.

User group types

There are three types of user groups:

- Ordinary user groups
- Large user groups
- User Groups for Voice Mail and Other Applications

4.6.1 Ordinary user groups

Elements of a User Group

A user group consists of one or more of the following elements:

- Subscriber line group (SLG):
  Group with up to 16 internal subscribers (members).
- Operator Console:
  The call is signalled in parallel on all Operator Consoles (see "Operator Console", page 248).
- General bell:
  Centralized acoustic signalling of a call (see "Answering general bell", page 521).

All the elements can be connected to each user group in the user group configuration (see Tab. 2.21).

Tab. 2.21: How user group elements are connected

Element	Added on by:
Subscriber group	entering at least one subscriber as member of the group
Operator Console	Connect [yes / no]
General Bell	Connect [yes / no]

Note:
If the element Operator Console or General Bell is connected without an Operator Console or General Bell actually being connected, calls to this destination will simply idle.

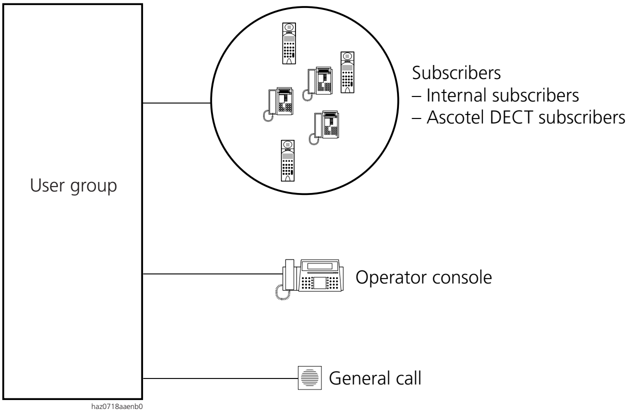


Fig. 2.43: Elements in a user group

Call Distribution to the Elements

A call is distributed in parallel to the connected elements of a user group. Each element can be individually delayed. The delay time can be set globally to 3, 5 or 7 ringing cycles and applies throughout the system to all line groups.

Call Distribution in the Subscriber Group Element

There are three possibilities for call distribution to the members within a subscriber line group:

- global
- linear
- cyclic

Global Call Distribution

In a global call distribution all the available members in the group are called simultaneously. As soon as any member answers the call, the call to the other members is cleared down.

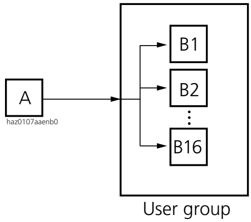


Fig. 2.44: Global Call Distribution

Linear Call Distribution

In a linear call distribution the first member in the group is called first. If he does not answer, the call is forwarded to the next member after 3, 5 or 7 ringing cycles. Linear call distribution bypasses busy members.

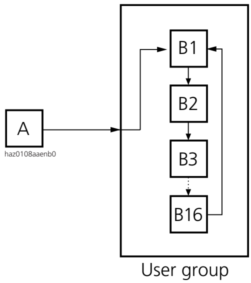


Fig. 2.45: Linear Call Distribution

Cyclic call distribution

Call distribution is the same as in the linear variant except that each new call is first signalled in each case to the next member in the row.

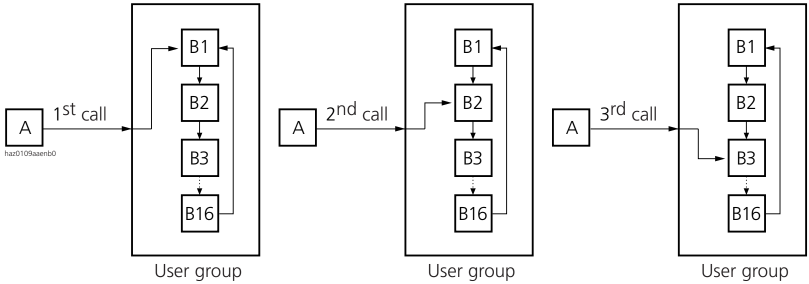


Fig. 2.46: Cyclic call distribution

Delayed Calls to Subgroups

The members of the subscriber group element can also be subdivided into a main group and a subgroup (setting "UG subgroup as of member").

The subgroup is called according to the set call distribution:

- If call distribution is set on "global", the subgroup rings once the configured delay time has elapsed.
- If call distribution is set on "linear or cyclic", the subgroup rings once the configured forwarding time has elapsed after the call has been ringing at the last member of the main group.

The members of the subgroup are always called according to the "global" call distribution.

In Summary

In a user group there are two selectable times that can be used for controlling call distribution. Both are preconfigured in the system configuration:

- The delay time affects
  – The user group elements. It can be activated and deactivated for each element.
  – The subgroup of members of the subscriber group set on global.
- The forwarding time for linear and cyclic call distribution among the members of the subscriber group.

The duration of the delay time and forwarding time can be set globally to 3, 5 or 7 ringing cycles.

Other delay times can also be specified on a subscriber terminal, e.g. the delayed signalling on a line key of a key telephone or on a team key.

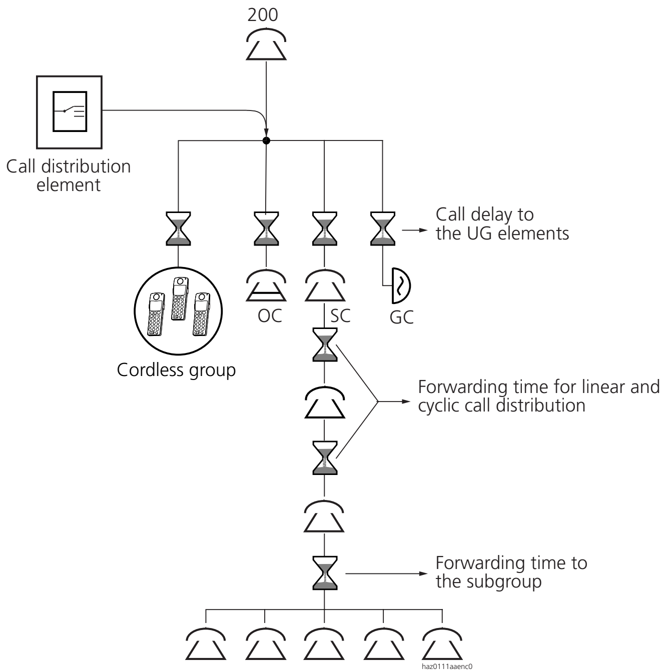


Fig. 2.47: Call distribution in a user group

Rules Applying to the Subscriber Group

Any member of a subscriber group can use the menu selection or a */# procedure to log out of (#48xx) or log into(*48xx) a user group (see also "User group: Logging in and logging out", page 533. Members who have logged out of a group are ignored during call distribution. The last remaining member in a group does not have the possibility of logging out of the group.

A subscriber may belong to several user groups at the same time. Logging out of or in to user groups applies simultaneously to all the user groups or specifically for one particular user group.

Any call forwarding to internal destinations that have been activated by group members are observed.
Any call forwarding to external destinations and PISN subscribers will result in the members concerned being logged out of the user group. The last member remaining in the subscriber group does not have the possibility of activating an external Call Forwarding Unconditional and cannot log out of the user group. Call Forwarding on No Reply can be activated at any time.

If all the members are busy, the system will respond as follows:

- An external call will be routed in accordance with the emergency routing concept (see "Response if busy", page 279.
- An internal call will be acknowledged with the busy tone.

Ascotel DECT-Subscribers

Ascotel DECT subscribers can be entered in a user group in the same way as internal subscribers.

GAP Subscribers

The following restrictions apply to entering GAP handsets as UG members:

- Only one GAP subscriber per user group
- A user group with one GAP subscriber must not contain any other DECT subscribers

4.6.2 Large user groups

Any user group in the PBX can be configured as a large user group in the System Configuration ("Large User Group = Yes"). Large user groups differ from ordinary user groups in the following ways:

- Apart from the general system limits (see Tab. 3.17) there is no other restriction on the number of members in a subscriber group.
- The elements Operator Console and General Bell are not possible.
- Global call distribution is not possible
- Subgroups are not available

- They can be the destination of a Call Forwarding Unconditional or Call Forwarding on No Reply, even if the diverting subscriber is still a member of a different user group.
- Any call diversion or call forwarding (even internal) by a UG member automatically results in the member being switched out of the user group. If there is only one member left in the subscriber group, that member cannot activate CFU or CFNR and therefore cannot be switched out of the user group.
  Note: In Twin Mode both the DECT handset **and** the corded terminal have to be entered in the user group.
- If a member makes outgoing external calls without a direct dialling number the direct dialling number of the user group is **not** used as CLIP.

4.6.3 User Groups for Voice Mail and Other Applications

User group 17 (System 2025 / 2045) and 25 (System 2065) have been designed to accommodate a Voice Mail server.

User groups 18 to 21 (System 2025 / 2045) and 26 to 29 (System 2065) are provided for applications that require a call forwarding to a user group.

These user groups differ from ordinary user groups in the following way:

- In the case of calls to these user groups UG subscriber redirections are not carried out. Callers who dial the subscriber in the user group directly are however diverted.
- They can be the destination of a Call Forwarding Unconditional or Call Forwarding on No Reply, even if the diverting subscriber is still a member of a different user group. In each case redirections to these user groups due to Call Forwarding will be carried out only once the call forwarding time has elapsed.
- It is not possible to divert these UG subscribers to the special user groups. This applies even if the subscriber logs out of the UG beforehand.
- Only the user group element "Subscriber group" is available.
- "Global" call distribution is not available.
- DECT subscribers are not allowed.

The following applies in particular for the Voice Mail user group:

- Communications with a Voice Mail server via the V.24 interface works only with the Voice Mail user group.

- Up to 16 voice channels (= user-group members) can be implemented for each user group.
- If the Voice Mail user group is not taken up by a Voice Mail application, it can be used for other applications.

User Groups 14, 15 and 16

After initialization, the element Operator Consoles and the first four subscribers are entered as members in a user group 16.

After an initialization each trunk group is allocated call distribution element 1. It is allocated user group 16 as the destination for all three switch positions.

User group 16 is used as the destination in the following cases:

- No suitable DDI number is found for an incoming call and call distribution element 1 is entered in the trunk group configuration.
- An incoming call reaches a busy user group, triggers call waiting and call waiting is rejected.
- An incoming call is routed to a Voice Mail system via the Voice Mail user group, and the system is out of order due to a fault.
- An incoming call is routed to a pager and the pager is not answered within a defined period of time (see Tab. 2.287).

Tip:
As the user group is used as an emergency routing destination, the elements or members configured in this user group must be suitable as alternative destinations.

User group 14, 15 and 16[1)]

User group 16 is reserved for Capolinea destination 1 and 2.

User group 14 is reserved for Capolinea destination 3.

User group 15 is reserved for the switching variant of Capolinea destination 1 and 2 (see "Capolinea", page 252).

1) For Italy only

User group 30 - 99

DECT group calls are not possible with user groups 30 - 99 (available only with the 2065 system), i. e. only one DECT terminal is allowed per user group.

Application example for a user group

In the call distribution pattern, general bell has been configured with a delay; along with the Operator Consoles. This means that if the Operator Consoles is overloaded the general bell will also start to ring after the configured ringing time (e.g. 3 ringing cycles). The call can then be taken from any terminal.

Other Subjects Related to the User Group:
Call distribution, Subscriber configuration, Operator Consoles, General bell, Internal traffic, Incoming traffic, User group: logging in and out, Numbering plan.

4.7 Subscriber Configuration

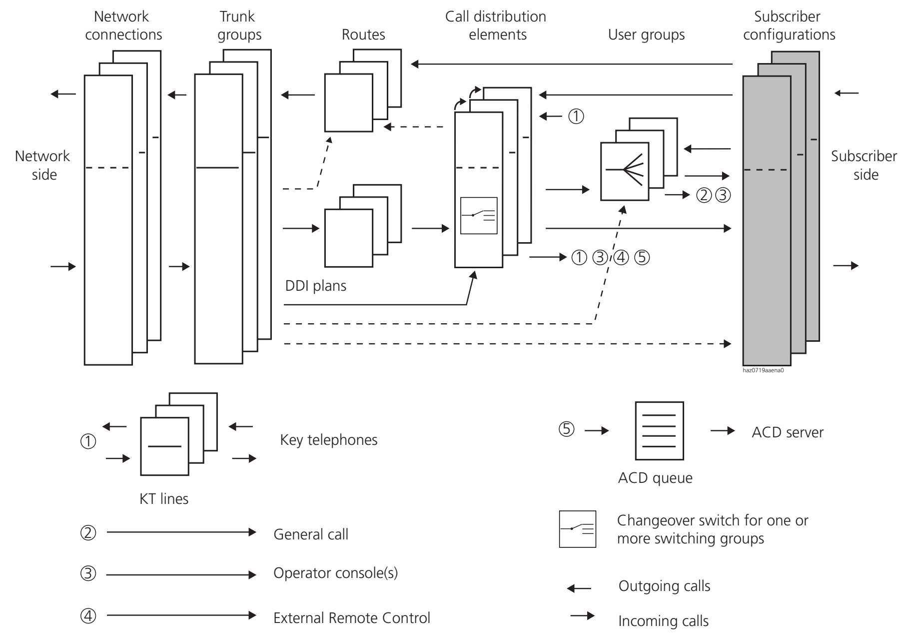


Fig. 2.48: Subscriber configuration in relation to the other routing elements

All the subscriber-specific settings are grouped together in the subscriber configuration. This chapter deals with the following topics:

- Routing and identification-specific settings
- Settings for PISN subscribers

4.7.1 Routing Functions for Incoming Calls

The incoming routing functions in the subscriber configuration are as follows:

- For a terminal the allocation of the internal subscriber number to a physical destination (user-network interface, terminal selection digit and terminal type)
- For a cordless handset the logical allocation to a subscriber identification stored in the handset

4.7.2 Routing Functions for Outgoing Calls

The following outgoing routing settings are grouped together in the subscriber configuration:

- Classes of service:
  - Exchange access authorization
  - Priority exchange allocation (see page 299)
  - Digit barring, external (see page 288)
  - Digit barring, internal (see page 268)
  - Partial rerouting (see page 328)
  - Least Cost Routing (see page 304)
- Outgoing call number for PISN subscribers
- Route allocation
- Forcing the route if the LCR function if activated (see page 317)

Classes of Service

Enable or restrict authorization to make outgoing phone calls to the public network from an allocated terminal. The following are excluded from the barring:

- Dialling abbreviated dialling numbers
- Dialling the emergency number
- Dialling PISN subscriber numbers

Call Number of the PISN Subscriber for Outgoing Calls

If a PISN subscriber is in a virtual network, his external (DDI) number will be listed here without the exchange access prefix. If a PISN subscriber is in a fixed network, a number is not usually entered here (see "Call to the private Leased-Line Network", page 301).

For a more detailed description of which subscribers of a different PINX can be entered as PISN subscribers, see "Shared Numbering Plan", page 175.

Route allocation

This setting allocates a route to the subscriber.

In the case of an internal subscriber this route is used to route calls that were dialled with an exchange access prefix (except route selection). If the LCR function is activated, the route is determined by the LCR unless the subscrib is authorized to force the route

When a PISN subscriber number is dialled, the route used is the one entered in the subscriber configuration for that PISN subscriber. If the LCR function is activated, the route will be determined by LCR.

Subjects Relating to Subscriber Configuration:
User-network interfaces, Call distribution, Route, User group, Operator Consoles, key telephones, Internal traffic, Incoming traffic, Outgoing traffic, Traffic in the PISN, Subscriber-related features, Numbering plan.

4.8 Operator Console

The system has one switching station, which is defined under the name "Operator Console" in the internal numbering plan. Several Operator Consoles can be operated on the same PBX. There are two types of Operator Consoles:

- The PC Operator Office 1550 is a PC terminal connected to the S user-network interface via the relevant Terminal Adapter.
- The system terminal Office 45 as an Operator Console connected to the AD2 interface.

With the exception of type-specific characteristics the following explanations apply to both types of Operator Console. Details and properties can be found in the type-specific documentation.

4.8.1 Routing Functions for Incoming Calls

Routing an Outside Call

Incoming calls are routed to the Operator Console(s) either directly or via a user group a call distribution element.

On an Office 45, Operator Console the calls are provided on the line keys. If all the line keys are busy, other calls will be sorted into the call queue.

On an Office 1550, the calls are entered in the external call queue. To answer the call the operator selects it directly from the call queue displayed on the graphic interface.

The operator can tell who the callers are from the call queue and can answer any of the call; the queue sequence does not have to be respected.

Routing Internal Calls

Internally the Operator Console is dialled up using the number of the switching centre defined in the numbering plan or via a call distribution element.

On an Office 45 (Operator Console) the calls are provided on the line keys. If all the line keys are busy, the calls will be placed into the internal call queue.

On an Office 1550, the calls are entered in the call queue for internal calls on the graphic interface. The operator select the call directly form the call queue.

Calls from the private leased-line network are handled in the same way as internal calls.

Routing a Personal Call (Internal or External)

The personal part of an Operator Console corresponds to an ordinary internal user. The calls are routed accordingly.

Call Signalling and Presentation on the Terminal

External and internal calls for the switching centre are signalled on all Operator Consoles.

Call Forwarding to a Substitution Destination

Calls to Operator Consoles can be diverted to a substitution destination (see "Substitution Circuit", page 262).

In a two-company system the call forwarding destination applies to both companies.

4.8.2 Routing Functions for Outgoing Calls

Routing an Outside Call

Seizing a line key enables direct network access and the network dialling tone is obtained. This means the subscriber does not have to dial an exchange access prefix to be able to dial out into the public network.
Calls are routed via route 1 except in the case of a two-company configuration (see "Two-company system", page 250).

With outgoing calls via the line keys the CLIP transmitted consists of the DDI number that is linked with the switching centre.

If a call number from the display or from a card file is preceded by an exchange access prefix with a hyphen, the prefix is truncated when dialling via a line key.

Example:
The display on the Operator Console indicates the number: 0-222 30 30. If a call is set up with this number via a line key, the number 222 30 30 is dialled and the call is transmitted to the public network via route 1.

Routing an Internal Call

Internal calls (on the Office 45 calls set up via the house key) are routed in the same way as an ordinary internal subscriber.

The CLIP consists of the personal internal subscriber number.

Routing a Personal Call (Internal or External)

The personal part of an Operator Console corresponds to an ordinary internal user. The calls are routed accordingly.

The CLIP consists of the personal internal subscriber number.

4.8.3 Two-company system

On a two-company system the Operator Console will indicate whether an incoming call is intended for Company A and B (see Fig. 2.49 as an example for Office 45).

The configuration as a two-company system only affects the display on the Operator Console. The following points need to be taken into account to ensure that the two-company operation is clearly separated:

- Use a separate direct dialling plan for each company.
- Allocate separate cost centres for each company.
- Use an internal digit barring,
  - if internal traffic between the companies is not possible.
  - to prevent outside cost centres from incurring charges through cost centre selection or route selection.

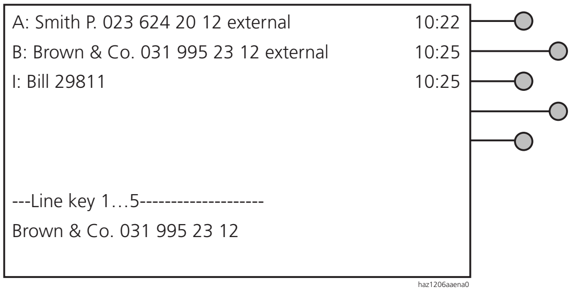


Fig. 2.49: Display on the Operator Console Office 45 in two-company mode

Routing an Incoming Call to the Operator Console

The company allocation of a call depends on the setting in the relevant call distribution element (see "Other Functions and Settings of the CDE", page 232).

Routing an Outgoing Call from the Operator Console

External outgoing calls from Company A are routed via route 1; external outgoing calls of Company B, via route 2.

Call Logging of Calls on the Operator Console

Call data, whether incoming or outgoing, is not logged separately according to company.

Initialization setting

Upon initialization all call distribution elements are configured for Company A (single-company system).

4.8.4 Capolinea[1)]

The purpose of the Capolinea feature is to ensure that each incoming call is answered. Therefore calls not answered by the destination subscribers are routed to alternative destinations (see "Response if busy", page 279). Operator Consoles are used as alternative destinations.

Capolinea Destinations

Unlike the standard operator function in the system, Capolinea has three destinations for Operator Consoles. They are defined throughout the system with the "Capolinea Destinations" setting (input of the internal subscriber numbers of the Operator Consoles).

Routing to a Capolinea Destination

An unanswered incoming call is routed to one of the user groups 16, 15 or 14. The following Capolinea destinations are allocated to the "Operator Console" user group elements:

- In user group 15 and 16
  - Capolinea destination 1 is allocated for Company A.
  - Capolinea destination 2 is allocated for Company B.
- In user group 14, Capolinea destination 3 is allocated.

User group 15 acts as a night service variant to user group 16.

An unanswered recall in response to "Transfer without prior notice" is also routed to a Capolinea destination (see "Call transfer without prior notice", page 469).

Configuration Notes

Tab. 2.22: Destination configuration in the call distribution element:

Capolinea destination	Switching position	Company	Destinations
1	1 (Day)	A	SC[1)]
1	2 (Night)	A	SC + UG 15
2	1	B	SC + UG 16
2	2	B	SC + UG 15
3	1	A	SC + UG 14

1) Here UG 16 is already configured and hidden as the destination; therefore it no longer has to be specially set (SC = SC+UG 16

1) For Italy only

Tab. 2.23: Configuration for user groups

User group	Elements configured	Initialization value
14	Operator Console, delayed	–
15	Operator Console, delayed, or General bell, delayed	Operator Console, delayed
16	Operator Console, delayed	Operator Console, delayed

Do not use the user groups for purposes other than Capolinea.

Other Subjects Relating to the Operator Console:
Terminals, PC operator, Subscriber-related features, Numbering plan

4.9 General bell

Calls with the general bell as destination can be signalled visually or acoustically using an external supplementary equipment. The call can be taken from any terminal (see "Answering general bell", page 521).

4.10 Key Telephones

Key telephones have several line keys and an internal key. For incoming traffic each line key of a key telephone is a routing destination addressed using the relevant call distribution element. This means for example that calls with a different DDI number can be offered on any line key.

For outgoing traffic each line key is linked with a separate routing. This means for example that a specific exchange line can be used for dialling by operating a line key.

With the internal key a key telephone can be operated like an ordinary featurephone.

Using Terminals as Key Telephones

The following system terminals can be configured as key telephones:

- Office 35/35IP
- Office 45/45pro
- Office 30
- Office 40

A system terminal automatically becomes a key telephone as soon as a KT line is placed on one of the terminal's line keys.

Key Functions

When a featurephone is converted to a key telephone one of the keys becomes an internal key, a number of keys become line keys, and the remaining keys can be freely configured in the same way as on the featurephone.

The internal key allows the key telephone to be addressed and used in the same way as an ordinary internal subscriber, in accordance with the settings in the subscriber configuration.

The maximum number of line keys possible depends on the terminal type.

The key telephone can be set in such a way that an incoming or outgoing call on a line key is either automatically allocated a KT line or automatically answered, as the case may be. Depending on the type of terminal the line keys can be provided with up to 9 priority levels (see the system terminal’s Operating Instructions).

Signalling

A call on a KT line is signalled both acoustically and visually. The status of the KT lines is indicated by LED signalling. The status of the KT lines is indicated by LED signalling.

Tab. 2.24: LED signalling on the line keys of a key telephone

LED signalling	Meaning
LED flashing rapidly	Call on that line
LED lit	Line is seized
LED flashing slowly	Line is parked

4.10.1 KT lines and Line Keys

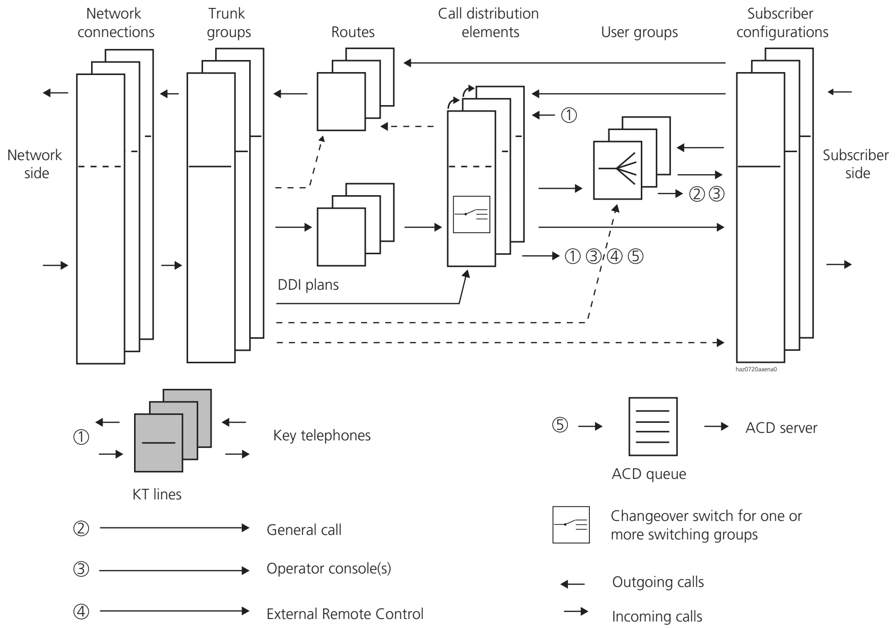


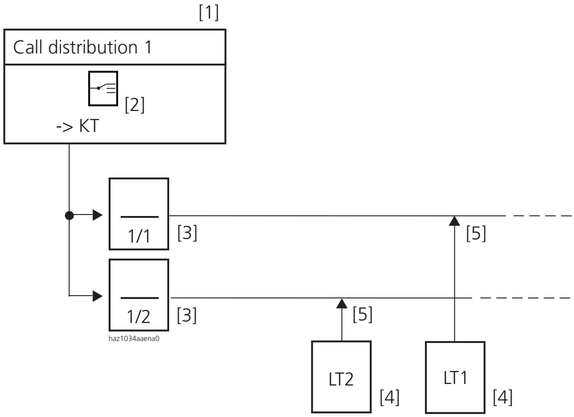
Fig. 2.50: Key telephones in relation to the other routing elements

KT Lines

Each call distribution element is allocated under its reference number one or more lines for key telephones (KT lines) if "KT" (or destination combinations with that destination) has been set as the destination (see "Call destination", page 227).

Line Keys

Each line key of a key telephone is allocated to a KT line. For example one line key is allocated to KT line "1/1", another to KT line "1/2". The first digit is the reference number of the call distribution element; the second digit is the line number. It also indicates the priority with which calls are offered on the line.



[1] Call distribution element with reference number 1
[2] Set destination: KT or combinations with KT
[3] KT lines
[4] Line keys on the same or different key telephones
[5] Allocation of the line key to a KT line

Fig. 2.51: Allocating line keys

Terminating KT Lines and through KT Lines

Any number of line keys from different key telephones can be allocated to the same KT line. If only one key telephone is allocated to one or several identical KT lines, we talk of a terminating KT line (TL). If several line keys of different key telephones are allocated to the KT line, we talk of a through KT line (THL).

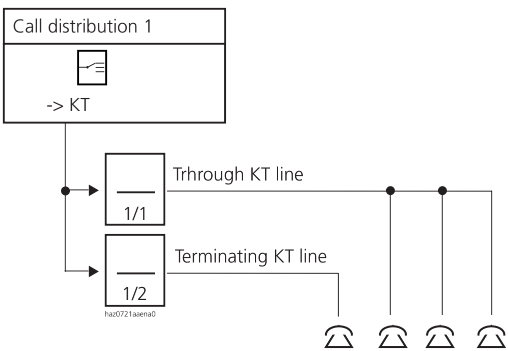


Fig. 2.52: Through and terminating KT lines

Note:
Unlike call forwarding to terminating KT lines, call forwarding to through KT lines are not carried out.

Tip:
Calls to through KT lines are normally answered by substitution by the other connected key telephones.
Switch-related destination assignment in the configuration of the call distribution element can be used to achieve an overflow for connections on a through KT line. For example Call Forwarding on No Reply to general bell or the Operator Consoles can be configured in combination with a delayed user group.

4.10.2 Incoming Calls via a KT Line

All calls can be routed via a KT line if the destination "KT" is defined in the corresponding call distribution element:

- Calls from the public ISDN network
- Calls from the public analogue network
- Calls from the private network
- Internal calls

If an incoming call reaches a busy KT line, the call is routed to the second KT line. If the second line is also busy, the call is routed to the third KT line, and so on. If there are no more KT lines available, "busy" is signalled. If "CDE if busy" is configured, the call is routed is via the call distribution element.

Note:
If a call is routed to a KT line to which no line key is connected, the call will simply idle or be routed to the alternative destination (setting "CDE if no answer").

Transferring from a Key Telephone to another Destination

Each connection on a KT line can be transferred to any internal subscriber.

Transferring to a Key Telephone

A call transferred to a key telephone is offered on the key telephone’s internal key or line key. If the call comes from the public network, it is signalled with the external ringing pattern.

Transferring to a key telephone with prior notice:

- If a call is transferred to a key telephone that is already receiving the call via a line key, it is offered on both the internal key and the line key. The call can be answered using either key.
  – If the call is answered using the internal key, the subscriber will be connected to the transferring party.
  – If the call is answered using the line key, the subscriber will be connected to the caller.
- If a call is transferred to a key telephone that is not receiving the call via a line key, it will be offered on the internal key only.
  If the call is answered, the subscriber will be connected to the transferring party.

Transferring to a key telephone without prior notice:

- If a call is transferred to a key telephone that is already receiving the call via a line key, the call will be offered on the line key only.
  If the call is answered, the subscriber will be connected with the caller.
- If a call is transferred to a key telephone that is not receiving the call via a line key, it will be offered on the internal key only.
  – If the call is answered, the subscriber will be connected with the caller.
  – If the call is not answered, it will be offered again to the transferring party once the recall time has elapsed.

Identifying a Call

System terminals with a display will indicate the name of the call distribution element if the call distribution element is configured with "DDI No. instead of CDE name = no" (initialization).
They will indicate the DDI number via which the call has been routed if "DDI No. instead of CDE name = yes".

4.10.3 Outgoing Calls via a KT Line

A KT line can be configured either as an outgoing line to the network or as a normal internal line.

KT Line as an Outgoing Line to the Network

Direct network access is enabled when a call is set up: The network dialling tone is obtained. This means the subscriber does not have to dial an exchange access prefix to be able to dial out into the public network. The route is determined by the "KT route" setting in the call distribution element.

If the call number dialled is a number with an exchange access prefix and a hyphen.

Example:
The display on the key telephone indicates CLIP number: 0-222 30 30. If an outgoing call is initiated by dialling this number, the number 222 30 30 is dialled and the call is transmitted to the public network via the configured KT route.

To enable outgoing calls to the public network, "Outgoing barring = No" must be set in the key telephone configuration. The setting "Outgoing barring = Yes" does not enable outgoing calls to be set up via this KT line.

The call charges can be logged via the "KT cost centre" setting.

KT Line as a normal Internal Line

If no KT route has been defined in the call distribution element ("KT route = -"), the KT line will respond like an ordinary internal line. This means the subscriber has to dial an exchange access prefix to be able to dial out to the public network. The route is determined by the "Route" setting in the subscriber configuration. Furthermore, the other settings in the subscriber configuration also apply.

The following number is presented as CLIP to the internal destination subscriber:

- The call number of the call distribution element, provided it has been allocated in the numbering plan.
- The internal call number of the key telephone if the call distribution element was not allocated a call number.

Note:
If a KT cost centre is entered in the call distribution element and a subscriber cost centre in the subscriber configuration, the call charges are allocated to both cost centres. This means the total sum of the call is allocated twice.

Application Examples for Key Telephones

Destination Combination KT+UG

The combined destination KT line and user group 5 has been configured in call distribution element 1 with number 200 in the numbering plan.

Two line keys are connected to the KT line 1/1 It is therefore a through KT line The first key belongs to the key telephone with subscriber number 211; the second belongs to the key telephone with subscriber number 221.

The element "Operator Console" is configured on user group 5. Internal subscriber 291 is entered as member of the subscriber group. Delay is activated for both elements (Operator Consoles and Subscriber).

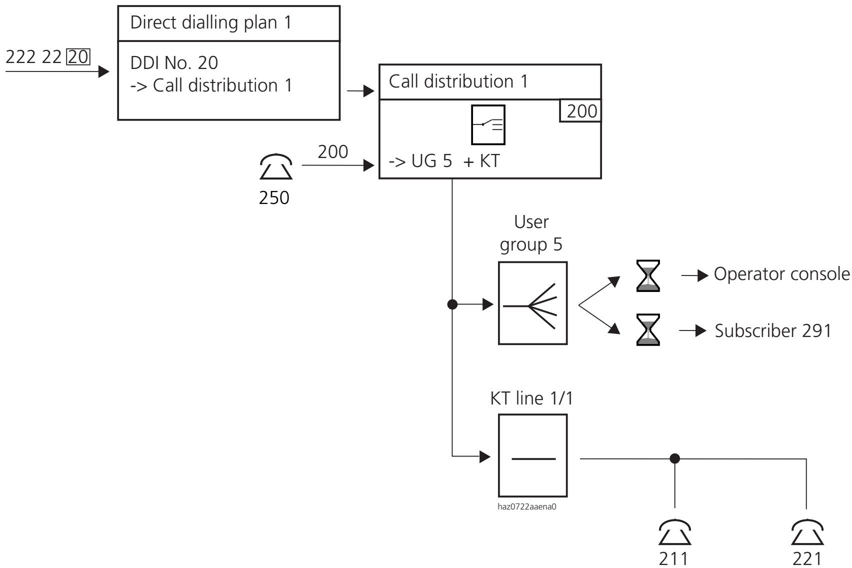


Fig. 2.53: Application for key telephones and user group

If an incoming call is not answered within the set delay time using the line keys of subscribers 211 or 221, the call will be routed on to user group 5 and signalled at the same time to the Operator Consoles and subscriber 291.

Destination KT

Travel agency Application

The number for the travel agency's Africa Desk is listed in the telephone directory under the number 222 22 20.
Calls for travel to Africa are first route to the Africa Desk At the Africa Desk the calls are answered by employees 1 to 3.

A call is offered on the line keys of KT line 1/1.
If KT line 1/1 is busy, the call is offered on the line keys of KT line 1/2, etc.

The travel agents working at the Europe Desk will only answer calls to the Africa Desk if all its three travel agents are busy. That is why they are only connected to the KT line for Africa in fourth priority (KT line 1/4).

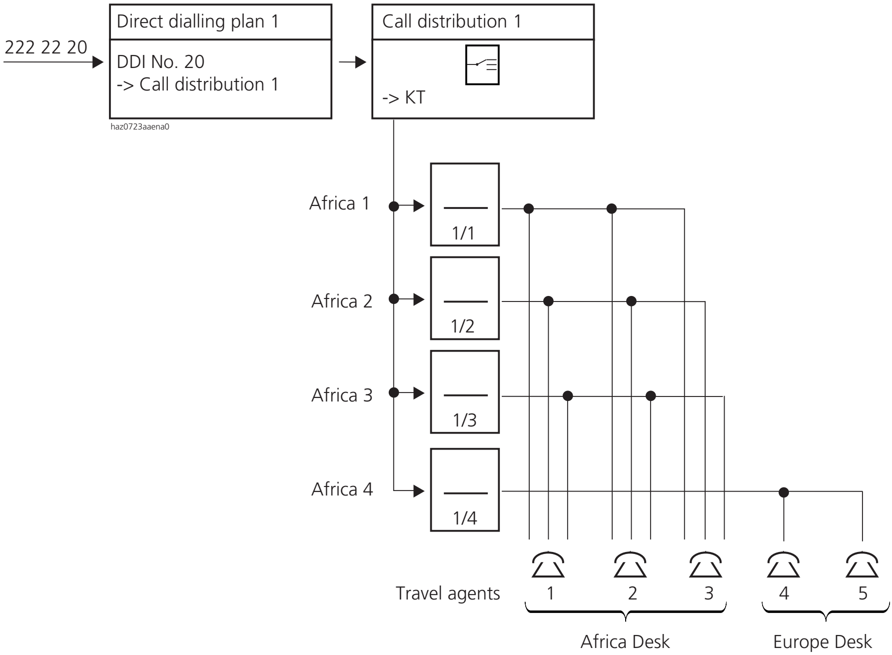


Fig. 2.54: Substitution Circuit

Substitution Circuit

The first call is answered by the manager personally; a second simultaneous call will ring on the deputy manager's set; the third call will ring in the secretary; the fourth caller will obtain "busy". The calls can be visually signalled everywhere immediately. Acoustic signalling takes place after a delay.

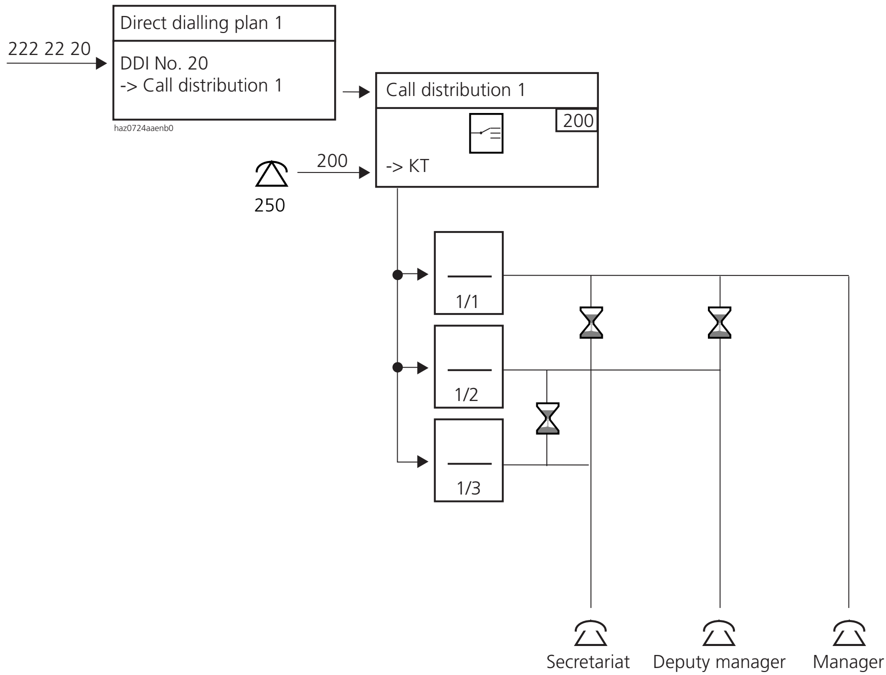


Fig. 2.55: Substitution circuit with key telephones

Other Subjects Relating to Key Telephones:

Internal traffic, Incoming traffic, Outgoing traffic, Subscriber-related features.

4.11 ACD Server

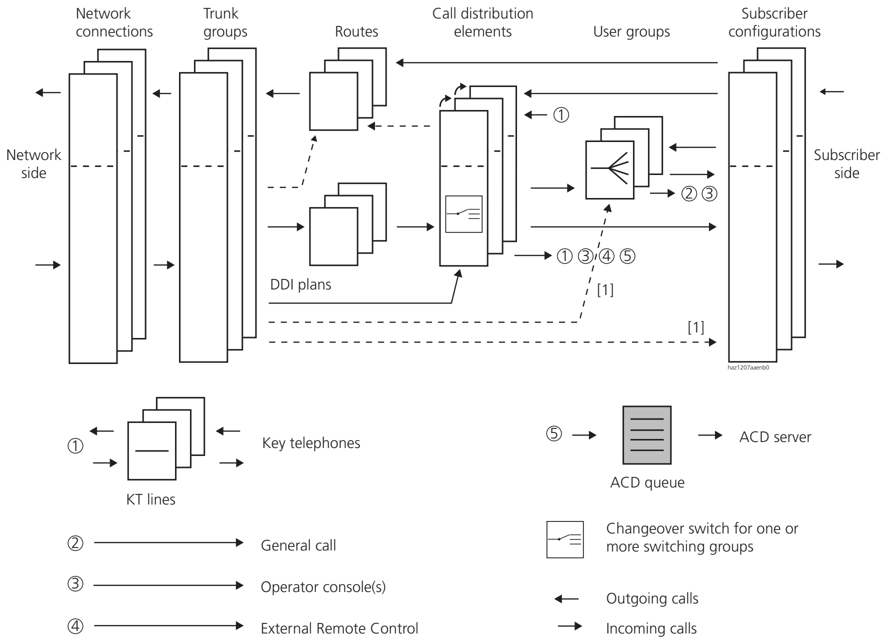


Fig. 2.56: The ACD server in relation to the other routing elements

With an ACD application on the third-party CTI interface, control of the call routing is shifted from the PBX to the external ACD server (ACD: Automatic Call Distribution). The ACD application determines the routing and the PBX routes call according to its default settings.

Calls to an ACD server are routed to the ACD queue where they are sorted ("ACD" destination in the call distribution element settings).

The PBX informs the ACD server of the calls in the ACD queue. The ACD server analyses the calls and tells the PBX where to route the calls. Potential destinations are internal subscribers and PISN subscribers (e.g. agents working from home).

Further Information
see "Ascotel CTI", page 133

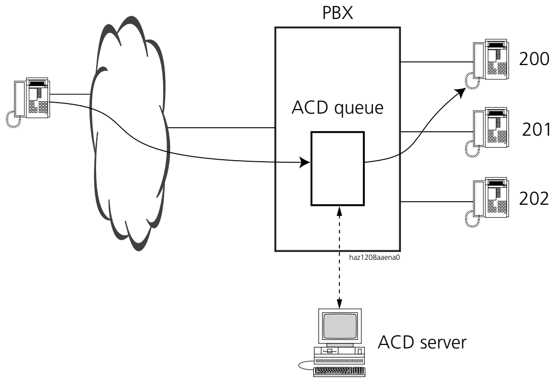


Fig. 2.57: PBX call routing controlled by the ACD server

If the call is not answered by the destination subscriber (agent) by the time the ringing time has expired or if the destination subscriber is busy, the PBX will return the call to the queue and inform the ACD server accordingly.

Utilization of the ACD queue is subject to the acquisition of a licence.

Note:
For the ACD server to analyse calls correctly, "DDI No. instead of CDE name = yes" has to be configured in the direct dialling plan.

Call Routing in the event of an ACD Server Failure

Alternative destinations have to be defined so that calls can be routed to a destination even in the event of an ACD server failure (see "Alternative Destinations", page 229).

If the ACD server fails, an event message is generated ("ACD server out of operation").

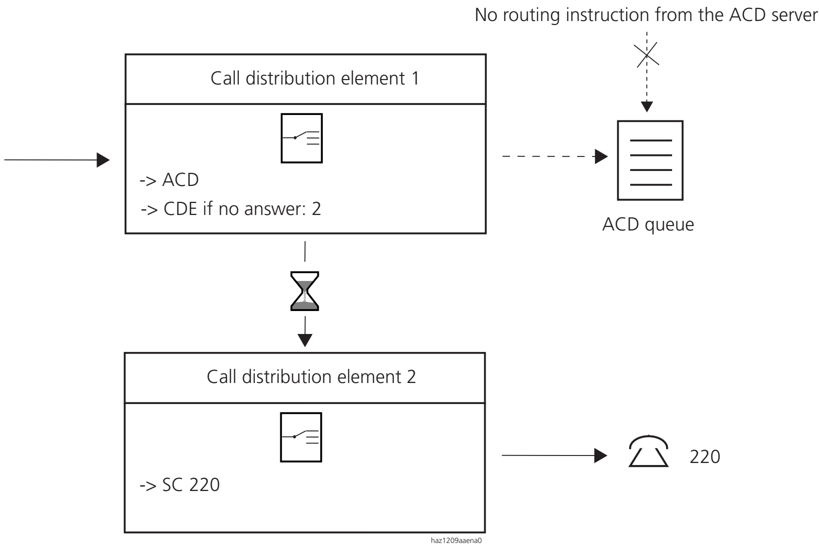


Fig. 2.58: Emergency routing in the event of an ACD server failure

If the same call routing as with the ACD server is to be achieved, the ACD server configuration has to be replicated in the PBX configuration also (for example ACD agent groups have to be replicated as user groups in the PBX configuration).

5 Call Routing

This chapter deals on the one hand with the interaction between routing elements with internal, incoming and outgoing traffic. On the other, it deals with topics that are clearly attributable to one particular type of traffic. For instances it also looks at Least Cost Routing in connection with outgoing traffic.

This chapter is divided as follows:

5.1 Internal Traffic

5.1.1 Internal Destinations

Many internal destinations are allocated numbers in the internal numbering plan. These destinations are dialled directly by dialling these numbers or the names allocated to them.

The table below shows the internal destinations, their availability and their dialling options.

Tab. 2.25: Internal destinations and their availability

Internal destinations	Remarks
Hard-wired subscribers:	In the internal numbering plan this is the "internal subscribers" category
• Subscribers on AD2 interfaces	Selectable using number and name selection
• Subscribers on S interfaces	Selectable using number and name selection
• Subscribers on a/b interfaces	Selectable using number and name selection
cordless subscribers:	In the internal numbering plan this is the category "internal subscribers"

Internal destinations	Remarks
• Ascotel DECT-Subscribers • Pager subscriber • Virtual subscribers	Selectable using number and name selection Selectable using number and name selection • Selectable using number and name selection • Respond in the same way as analogue subscribers
Internal destinations to which another destination has been permanently allocated: • Emergency number • Abbreviated dialling numbers • PISN Subscriber	 • Selectable using number dialling only • Destination No: internal, external, PISN subscribers • Selectable using number and name selection • Destination No: internal, external, PISN subscribers • Selectable using number and name selection • Destination No: PISN internal (subscribers on other PINX in the PISN)
Central destinations: • Operator Console • General Bell	 Selectable using number dialling only Selectable only indirectly via a user group or via coded ringing
Door Intercom Systems	• Selectable using number and name selection • Dialling: can only dial predefined destination
Distribution elements: • User Groups • Call Distribution Elements • KT lines on key telephones	 Selectable using number and name selection Directly selectable only via number selection • Selectable using number selection of the relevant call distribution element. • Dialling: using allocated line keys
Routing elements: • Routes	 Directly selectable only via number selection

5.1.2 Internal Digit Barring

There are several digit barring options available for internal traffic. The same rules apply as for external digit barring facilities (see "Digit Barring Facilities", page 288).

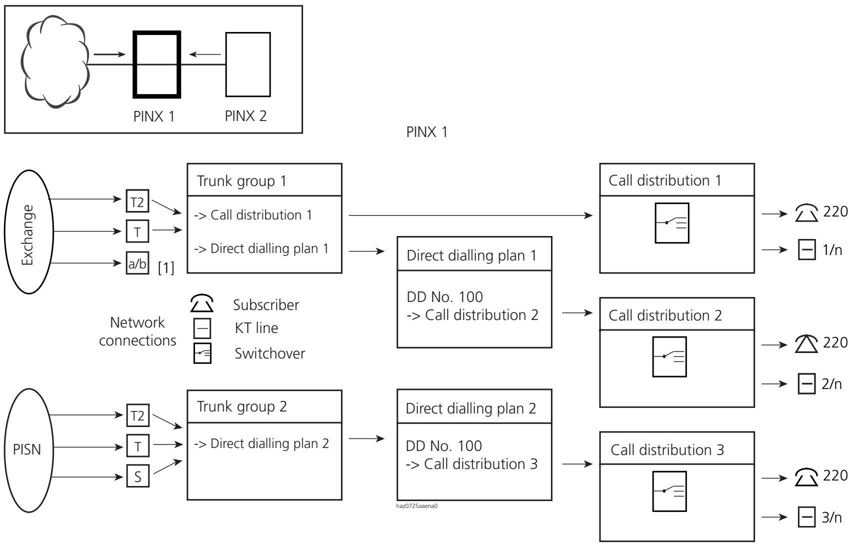
5.2 Incoming Traffic

5.2.1 Routing

Network interfaces with the same network-specific characteristics are all grouped together in a trunk group. It is for example specified whether the network interfaces allocated to a trunk group are connected to a private leased-line network or to the public network.
A call is routed via a trunk group to a direct dialling plan, a call distribution element or a destination with a number from the internal numbering plan.

Each direct dial number is allocated a call distribution element. Several direct dial numbers can be allocated to the same call distribution element.

A call distribution element is allocated destinations according to the circuit (see "Call destination", page 227).



[1] One and the same trunk group cannot contain both analogue and digital network interfaces

Fig. 2.59: Routing and destinations of an incoming call

Call routing depends in principle on whether a call originates

- from the public network or
- from the private leased-line network (QSIG) and
- whether there is a suitable direct dial number for the phone number.

In terms of call routing, calls from a virtual PISN are handled in the same way as calls from the public network.

The diagram below shows how an incoming call is routed:

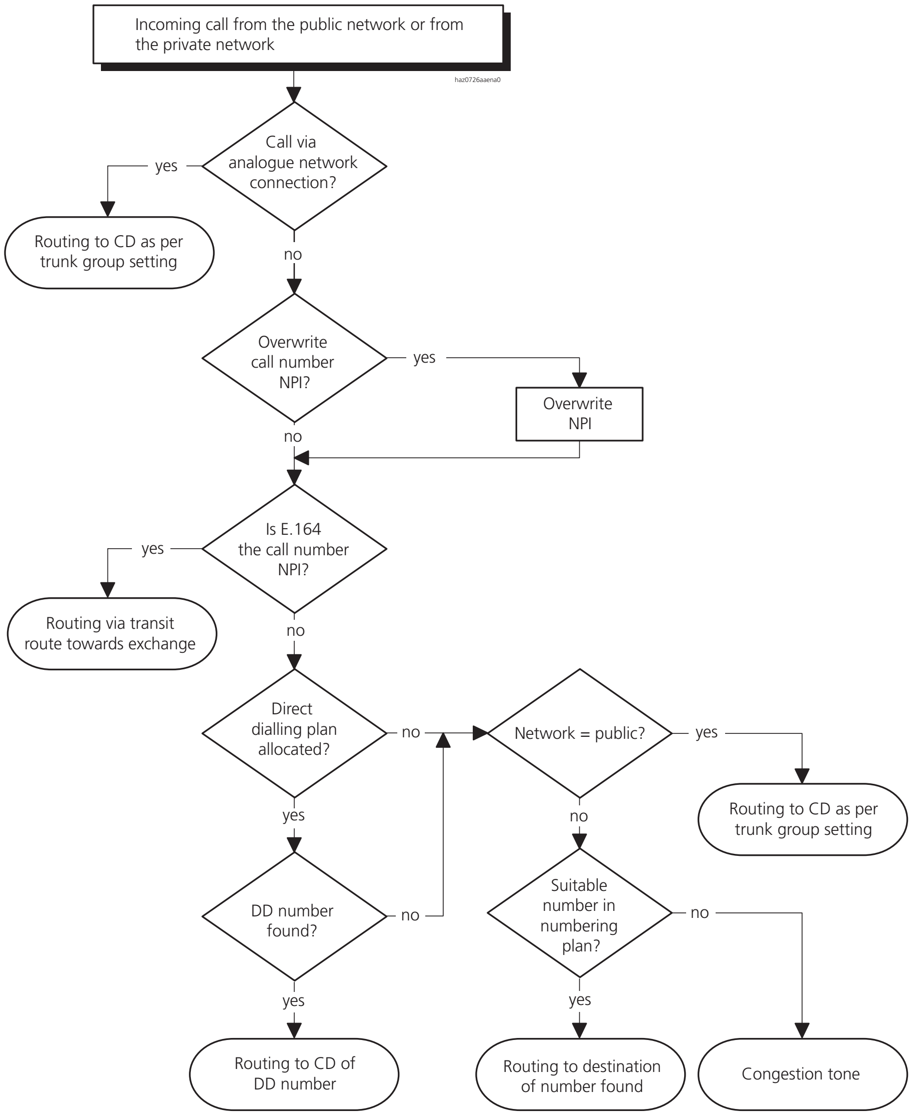


Fig. 2.60: Routing an incoming call

5.2.1.1 Call from the Public Network

A call with a suitable direct dial number is routed to the destination via the call distribution element allocated in the direct dialling plan.

If a suitable direct dial number is not found, the call is routed in the same way as a call from the public network without direct dialling (see "Call from the Public Network", page 272).

Direct dialling is not supported for calls from the analogue network.

Routing with Direct Dialling

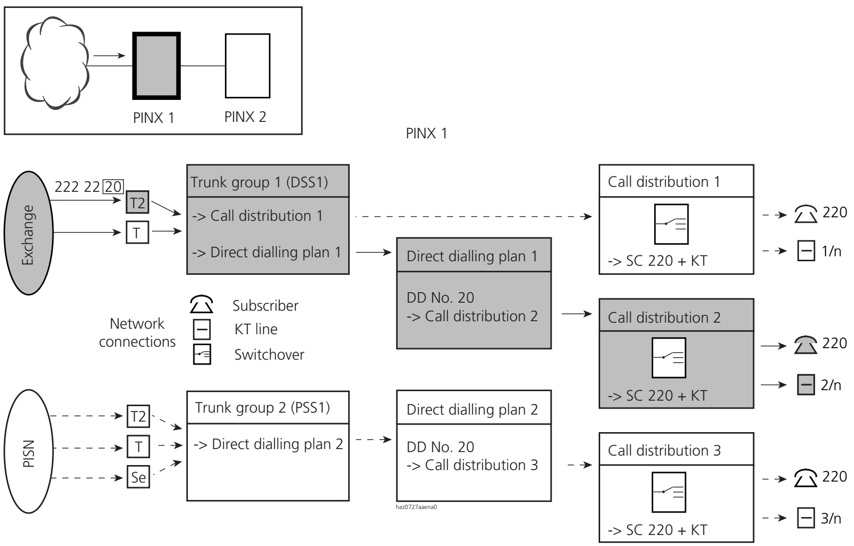


Fig. 2.61: Routing a call from the public network with direct dialling

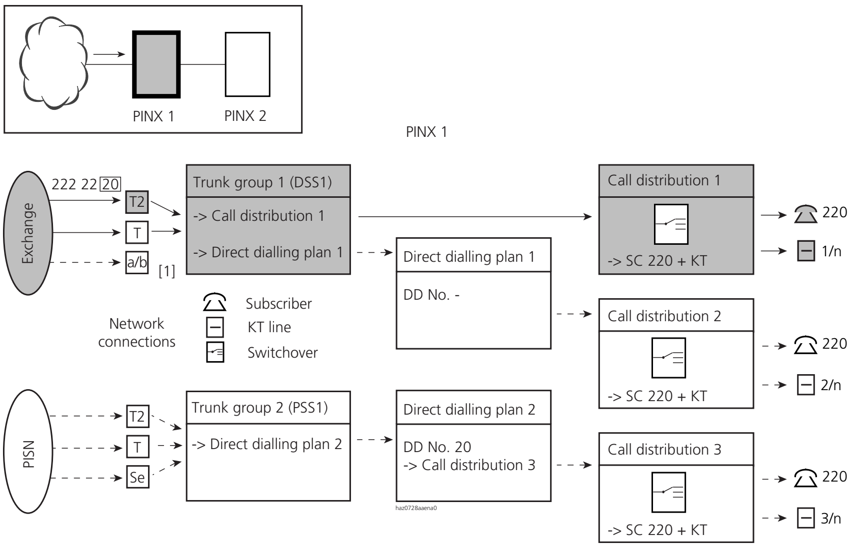
Tab. 2.26: Setting the routing parameters

Parameter	Parameter value
Trunk group 1:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Public
• Protocol	DSS1
• Overwrite NPI	No

Parameter	Parameter value
• Direct dialling plan • Call Distribution Element	1 (number of a direct dialling plan) 1 (significant only if a suitable direct dial number is not found)
Direct dialling plan 1: • DD number 20	 2 (reference number of a call distribution element)
Call distribution element 2: • Destinations • Incoming connections	 Switching position 1: SC 220 + KT Number of connections allowed simultaneously

Routing without Direct Dialling

A call without a suitable direct dial number is routed to the call destination via the call distribution element allocated in the trunk group.



[1] One and the same trunk group cannot contain both analogue and digital network interfaces.

Fig. 2.62: Routing a call from the public network without direct dialling

Tab. 2.27: Setting the routing parameters

Parameter	Parameter value
Trunk group 1: • Network interfaces • Incoming connection	 Network interfaces in this trunk group Number of connections allowed simultaneously

Parameter	Parameter value
• Network type	Public[1]
• Protocol	DSS1[1]
• Overwrite NPI	No[1]
• Direct dialling plan	1 (relevant only if a suitable DD number is found)
• Call Distribution Element	1 (reference number of a call distribution element)
Call distribution element 1:	
• Destinations	Switching position 1: SC 220 + KT
• Incoming connections	Number of connections allowed simultaneously

1) Not relevant for trunk groups with analogue network interfaces

5.2.1.2 Call from the Private Leased-Line Network

In the private leased-line network, direct dialling plans are set up only if calls are to be routed to their destinations via call distribution elements in order to benefit from the advantages of the flexible routing properties of call distribution elements (see "Call Distribution Element (CDE)", page 226).

Call distribution elements can be dialled up directly if they have been allocated a phone number in the numbering plan and if they exist as PISN subscribers in the other PINXs. However, without a direct dialling plan it is more difficult to achieve a numbering that matches.

Tab. 2.28: Flexible routing with and without direct dialling plan; difference in numbering

	PINX 2 PISN Subscriber	PINX 1 DDI number	PINX 1 Call Distribution Element	PINX 1 Destination subscriber
with direct dialling plan	250	250 → 250	1	250
without direct dial-ling plan	250	–	1, phone number 250	251

Calls from the private leased-line network do not have any DDI numbers. If you set up a separate direct dialling plan, however, these numbers can also be handled in the same way as DDI numbers.

Tip:
Only individual numbers can be organized via a direct dialling plan; the others are organized directly in a numbering plan.

Routing with Direct Dialling

A call with a suitable number in the direct dialling plan is routed to the destination via the call distribution element allocated there.

If the first few digits of the phone number match the number entered under "Own regional prefix" in the numbering plan, they will be truncated before the search for a suitable direct dial is carried out.

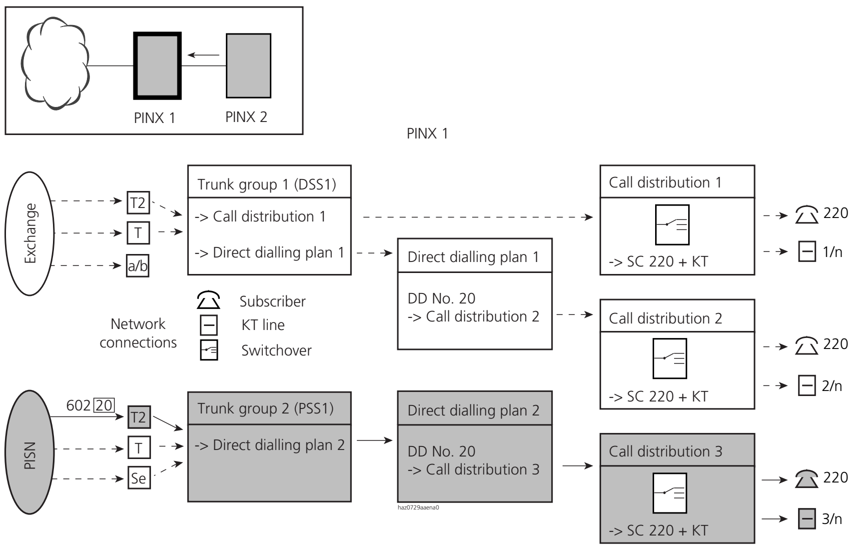


Fig. 2.63: Routing a call from the private leased-line network with direct dialling

Tab. 2.29: Setting the routing parameters

Parameter	Parameter value
Trunk group 2:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Private
• Protocol	QSIG or QSIG / PSS1 ISO
• Overwrite NPI	No
• Direct dialling plan	2 (number of a direct dialling plan)
• Call Distribution Element	Not relevant to this case
Direct dialling plan 2:	
• DD number 20	3 (reference number of a call distribution element)

Parameter	Parameter value
Call distribution element 3:	
• Destinations	Switching position 1: SC 220 + KT
• Incoming connections	Number of connections allowed simultaneously

Direct Routing

A call without direct dialling is routed directly to a destination of the internal numbering plan.

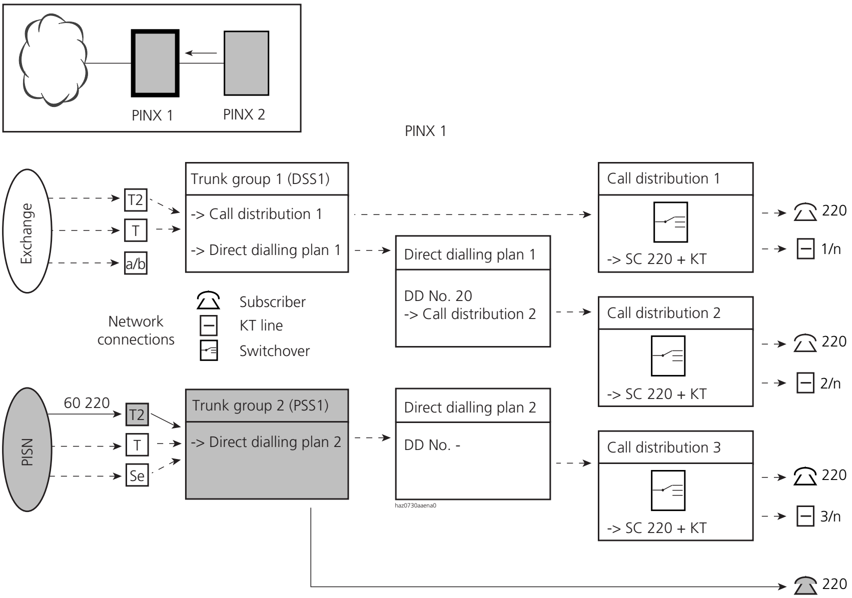


Fig. 2.64: Routing a call from the private leased-line network without direct dialling

Tab. 2.30: Setting the routing parameters

Parameter	Parameter value
Trunk group 2:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Private
• Protocol	QSIG or QSIG / PSS1 ISO

Parameter	Parameter value
• Overwrite NPI	No
• Direct dialling plan	2 (if a suitable DD number is found) or
• Call Distribution Element	Not relevant to this case

5.2.2 Call Forwarding Unconditional if no answer

Besides the CFNR redirecting function controllable by the subscriber and which forwards the call after a specific number of rings (see "Call Forwarding on No Reply (CFNR)", page 445), there are other configuration possibilities for redirecting an unanswered call.

5.2.2.1 CDE Alternative Destinations

If at the original destination the call is neither answered nor forwarded within a configurable period of time, it can be routed to a CDE alternative destination (see "Alternative Destination if no Answer", page 229).

5.2.2.2 Default Call Forwarding per Subscriber

Default response in the case of unanswered calls can be configured for each subscriber for both internal and external calls. With the appropriate configuration a Call Forwarding Unconditional to a call distribution element is effected after delays set separately in each case. This means the default response if unobtainable can vary according to the call's origin, e.g. Voice Mail for internal calls and transfer for external calls. In a PISN the internal response within the network can be handled in the same way with this function as the response of the local PINX.

The table below shows the interaction with other activated functions, configurations and situations when the Default Call Forwarding function is configured:

Tab. 2.31: Interaction of Default Call Forwarding with...

Function / Configuration / Situation	Response
Function CFU Unconditional activated	Only CFU Unconditional is executed
Function CFB activated and subscriber free	Only Default Call Forwarding is executed
Function CFB activated and subscriber busy	Only CFB is executed (Default Call Forwarding can only be used with free subscribers)
Call Deflection (CD) activated before Default Call Forwarding	Default Call Forwarding is not executed

Function / Configuration / Situation	Response
Function CFNR activated after 0, 3, 5 or 7 rings	Depends on the parameter "Priority over activated CFNR": • "No": Only CFNR is executed • "Yes": Default Call Forwarding is always executed. (If the CNFR call forwarding delay is shorter than the internal or external delay of the Default Call Forwarding, CNFR is executed first.)
DECT handset is unobtainable	Depends on the parameter "Unobtainable signal" • "Unobtainable": Default Call Forwarding is executed. • "Busy" or "Forwarded": Default Call Forwarding is not executed.
Entry under "CDE if no answer" in the CDE configuration	Depends on the times configured: If the CDE call forwarding delay in the CDE configuration is shorter than or equal to the internal or external delay of the default call forwarding, CDE call forwarding is activated; if not, the default call forwarding function is executed.
Call transfer without prior notice	Depends on the times configured: The recall is always carried out. If the internal or external delay of the default call forwarding is shorter than the recall time, the default call forwarding function is executed first.
Routing of the call to the subscriber via UG	Default Call Forwarding is not executed
Routing of the call to the line key to a key telephone	Default Call Forwarding is not executed

Other characteristics of the Default Call Forwarding function:

- The function is activated only if the called subscriber is free
- Unlike CFNR (*61) the subscriber terminal diverting the call does not carry on ringing in parallel.
- The Default Call Forwarding is still executed even if no terminal is connected.
- Before the Default Call Forwarding is executed, a check is carried out to see if the destination is free. If the destination is busy, the call is not forwarded. Each time the timer expires the delay timer is restarted for a new connection attempt.

Default Call Forwarding with calls already diverted:

Situation: Subscriber A calls subscriber B, who has redirected to subscriber C. A default call forwarding to subscriber D is configured at subscriber C.

Tab. 2.32: Default Call Forwarding response to calls already forwarded

Subscriber B has...	Default Call Forwarding is executed
CFU Unconditional activated	Yes
CFB activated	Yes
CFNR activated	No
Call Deflection (CD) activated	Yes
Follow Me activated	No
Default Call Forwarding activated	Yes

Redirect the destination of a Default Call Forwarding

Situation: Subscriber A calls subscriber B, where a Default Call Forwarding to subscriber C has been configured. Subscriber C has activated a call forwarding to D.

In this case the call forwarding from subscriber C to subscriber D is always executed.

Note:
Although chains of several default call forwarding are also possible, they do involve long ringing times.

System configuration

All the settings can be configured individually for each subscriber

Tab. 2.33: Default Call Forwarding: System configuration

Parameter	Parameter value	Remarks
Internal call delay	<10 to 300 seconds>	
CDE for internal call	<CDE No.>	
External call delay	<10 to 300 seconds>	
CDE for external call	<CDE No.>	
Priority over activated CFNR	Yes / No	

5.2.3 Response if busy[1)]

The following Chapter describes how the system responds when busy and how that response can be influenced using specific settings.

1) Does not apply to Italy

5.2.3.1 Response if the call destination is busy

If the call destination is busy, an incoming call will be handled according to the type of destination. Busy call destinations may be:

- An individual, busy subscriber
- A busy user group
- A busy KT line
- A subscriber with a stored message
- A user group with busy subscribers but without the elements Operator console and General bell.

Within the context of this Chapter a call destination is said to be busy if both he original destination and the alternative destinations, where configured, as busy (setting "CDE if busy").

Call destination: Individual, busy Subscriber

Call waiting allowed but is rejected

- In the case of an incoming call from the public ISDN network the caller obtains the busy tone.
- In the case of an incoming call from the private leased-line network call waiting is not possible.
- In the case of an incoming call from the public analogue network call waiting is repeated.

Call waiting not allowed or not possible

If no alternative destinations have been configured, the following rules apply:

- In the case of an incoming call from the public ISDN network the caller obtains the busy tone.
  If the caller has subscribed to the service "Automatic callback (CCBS)" with the network operator, he can activate that service.
- In the case of an incoming call from the private leased-line network the caller obtains the busy tone.
- In the case of an incoming call from the public analogue network the caller waits until the called party is free (polling).

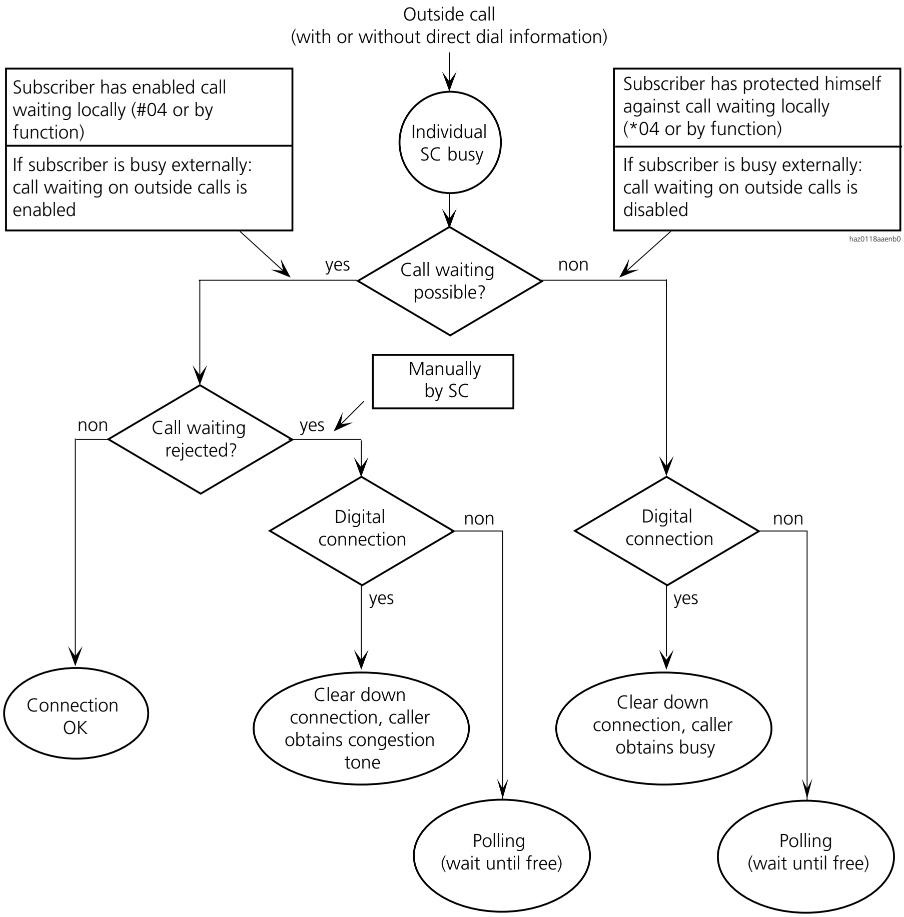


Fig. 2.65: Call distribution if subscriber busy

Call destination: Busy User Group

A user group is busy if all its members are busy, if call waiting is rejected, if call waiting is not enabled for any of the user group members and if neither the element Operator Console nor the element General Bell is activated.

If a user group is busy, an incoming call is routed to user group 16. If user group 16 is also busy,

- the caller in the public ISDN network will obtain the congestion tone after call waiting has been rejected;
- the caller in the private leased-line network will obtain the congestion tone.
- a call from the public analogue network will wait until the subscriber is free after call waiting has been rejected.

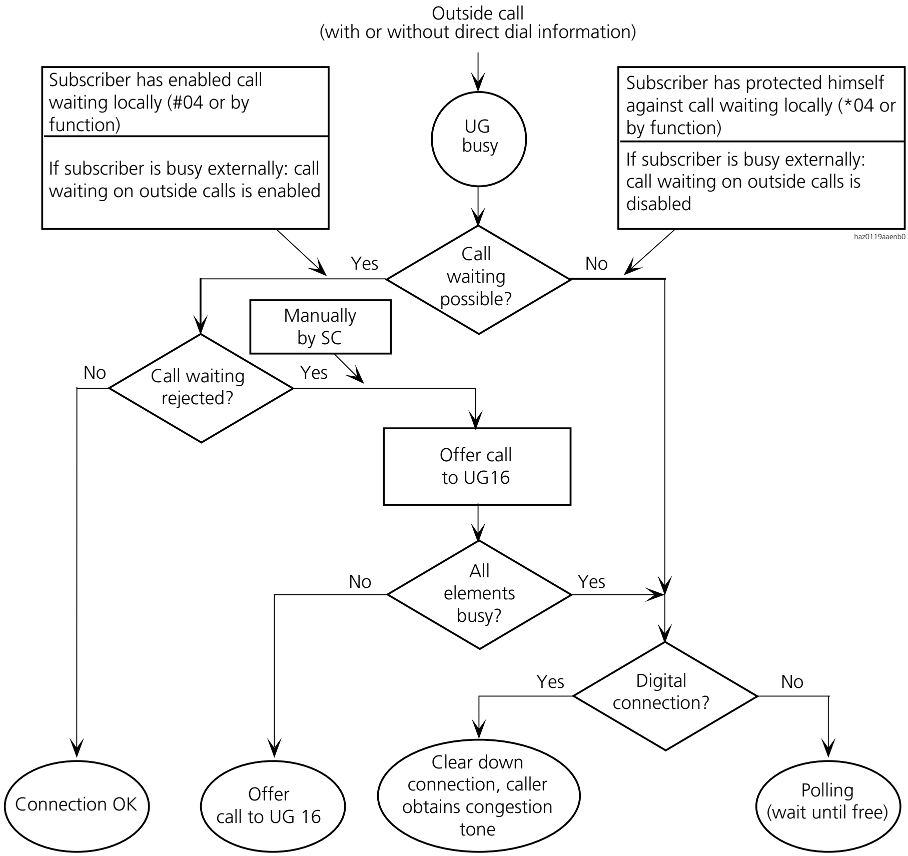


Fig. 2.66: Call distribution if user group busy

Call destination: Busy KT Line

If an incoming call is routed to a busy KT line, the call will be rejected and the caller obtains the busy tone.

Call destination: Subscriber with a Stored Message

If a subscriber has stored a message, an incoming call will be routed to the predetermined Call Forwarding Unconditional destination.
If a predetermined Call Forwarding Unconditional destination has not been defined, the subscriber will be called nonetheless.

5.2.3.2 Forwarding a call if busy

To ensure that each incoming call is answered, the following configuration recommendations must be observed:

Configuration for Subscribers and terminating KT Lines

- Configure "Call Forwarding on No Reply if busy" and "predetermined Call Forwarding on No Reply":
  The call is diverted to a predetermined call forwarding destination if the subscriber is busy.
- Configure predetermined Call Forwarding Unconditional.
  The call will be routed to a predetermined Call Forwarding Unconditional destination in the case of a stored message or Call Forwarding Unconditional to standard text.
- Activate permanent Call Forwarding on No Reply.
  If the subscriber does not answer, a delayed call is made to the CFNR destination.

Configuration of User Groups

Enter elements with call queues in the user group (Operator console or General bell).

Configuration of the call distribution elements

Configure alternative destinations if busy (setting "CDE if busy").

Configuration for through KT lines

- In the call distribution configure "KT line and user group" as the destination
- Delay the elements of the user group.

The user group is therefore an additional distributor if all the addressed through KT lines are busy..

Using a Voice Mail System

Unanswered calls can also be forwarded to a Voice Mail System where they are processed (see "User Groups for Voice Mail and Other Applications", page 243).

5.2.3.3 Not Forwarding a Call if busy

If the caller is to obtain the busy tone when the subscriber is busy, the following configuration recommendations must be observed:

- Do not configure an alternative destination if busy (leave the "CDE if busy" setting blank).
- Do not configure Call Forwarding on No Reply if busy
- Disable call waiting on exchange connections in the system configuration
- Disable local call waiting using *04

Note:
If a fax machine is connected to an internal user-network interface, disable "Call waiting".

5.2.3.4 Release Destination if Incoming Dialling is Incomplete[1)]

If the direct dial number is incompletely dialled the outside call will be routed to the call distribution element allocated to the trunk group after 8 to 15 seconds (depending on the country) and then forwarded to the destinations entered there.

1) Only in countries in which digit-by-digit DDI is implemented in their public exchanges.

5.2.4 Emergency Routing[1)]

5.2.4.1 Routing if the Call Destination is busy

If the call destination is busy, an incoming call will be handled according to the type of destination. Busy call destinations may be:

- an individual, busy subscriber
- a busy user group
- a busy KT line
- A subscriber with a stored message

Call destination: Individual, busy Subscriber

Call waiting is allowed, but is rejected

Tab. 2.34: Call waiting is allowed, but is rejected

	Response if the Capolinea destination ...	
Origin of the call	... is defined	... is not defined
Call from the public ISDN network	Call is routed to the defined Capolinea destination	Call is cleared down, caller obtains busy tone
Call from the public analogue network	Call is routed to the defined Capolinea destination	Wait until free, caller obtains ring-back tone

Call waiting is not allowed

Tab. 2.35: In the call distribution "CDE if busy" is set on Capolinea

	Response if the Capolinea destination ...	
Origin of the call	... is defined	... is not defined
Call from the public ISDN network	Call is routed to the defined Capolinea destination	Call is cleared down, caller obtains busy tone
Call from the public analogue network	Call is routed to the defined Capolinea destination	Wait until free, caller obtains ring-back tone

Note:
If a fax machine is connected to an internal user-network interface, "Call waiting" should be disabled for that particular subscriber.

1) For Italy only

Call destination: Busy User Group

A user group is busy if all its members are busy, if call waiting is rejected, if call waiting is not enabled for any of the user group members and if neither the element Operator Console nor the element General Bell is activated.

If a user group is busy, an incoming call is routed to user group 16.

If Call waiting is not enabled for any of the members of user group 16, the caller is obtains busy tone.

Call destination: Busy KT Line

If an incoming call is routed to a busy KT line, the call will be rejected and the caller obtains the busy tone.

Call destination: Subscriber with a Stored Message

If a subscriber has stored a message, an incoming call will be routed to the predetermined Call Forwarding Unconditional destination.
If a predetermined Call Forwarding Unconditional destination has not been defined, the subscriber will be called nonetheless.

5.2.4.2 Release Destination if Dialling is Incomplete

If the direct dial number is incompletely dialled, the outside call will be routed to the call destination element allocated to the trunk group after 8 seconds and then forwarded to the destinations entered there.

Scope

Valid only if the network provider transmits the digits of the direct dial numbers using the overlap receiving method. If the direct dial numbers are transmitted using the en-bloc method, an incomplete direct dial number will never be transmitted to the PBX.

5.3 Outgoing traffic

All outgoing calls are routed to a network via a route.

The authorization to make outgoing calls can be specified for each subscriber (page 299).

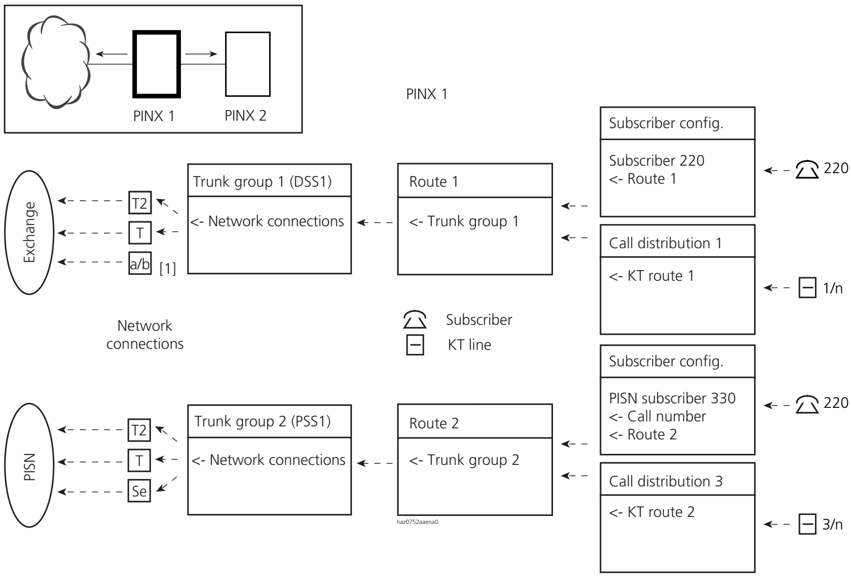
Digit barring facilities can also be used to regulate dialling access on the basis of the numbers dialled (page 288).

The feature "Priority exchange allocation" can be used to give priority to a subscriber wishing to set up an outgoing call (page 299).

The LCR (Least Cost Routing) function is used to control automatically the path (in the PBX and in the network) via which an outgoing call is to be routed (page 304).

5.3.1 Routing

All outgoing calls are routed to a trunk group via a route. They include calls routed via the Least Cost Routing function or transit calls in a PISN. Different types of call destinations have to be routed via different routes. For example calls to the private leased-line network must not be routed via the same routes as calls to the public network.



[1] One and the same trunk group cannot contain both analogue and digital network interfaces.

Fig. 2.67: Routing outgoing calls

5.3.2 Digit Barring Facilities

Digit barring facilities are user-definable filters used for regulating exchange access authorization based on the numbers dialled. Several digit barring facilities are available in each case for internal and outgoing traffic (internal and external digit barring facilities).

Difference between Internal and external Digit Barring:

- Internal digit barring filters internal phone numbers:
  Numbers that are entered in the internal numbering plan.
- External digit barring filters external phone numbers:
  Numbers that are sent into the network.

Allocating Digit Barring:

- Each subscriber can be allocated internal and external digit barring for all three switch positions.

- External remote control can be allocated an internal digit barring to restrict the features that can be remote controlled..
- The lock function on the phone lock variants activates an internal and an external digit barring.
- Digit barring facilities cannot be allocated to a PISN subscriber.

Bypassing the digit barring

Digit barring facilities are bypassed in the following cases:

- Deactivation of the external digit barring allocated to the subscriber in the route configuration
  Example:
  The digit barring is deactivated in the route configuration for route 1 and activated in the route configuration for route 2.
  If a subscriber with an allocated external digit barring sets up a call via route 1, the digit barring will not be consulted; if he sets up the call via route 2, the digit barring will be consulted.
- Calls via analogue network interfaces that are set to "Down-circuit from the PBX".
- Stored phone numbers of PISN subscribers
- Stored phone numbers of emergency and abbreviated dialling numbers, provided the emergency or abbreviated dialling number is dialled.
- Stored phone numbers of abbreviated dialling numbers, provided they are dialled using dialling-by-name.
- The digit barring for external remote control cannot be bypassed. (This applies only to internal subscribers on their own PBX but not to PISN subscribers in a QSIG network.)

Note:
If a procedure used for operating a feature is stored under an abbreviated dialling number, make sure the abbreviated dialling number is barred in the digit barring for unauthorized internal subscribers and that no name is assigned to the abbreviated dialling number. In a QSIG network this applies in particular to all PINXs that have entered the abbreviated dialling number as PISN subscriber in the numbering plan.

Setting up the digit barring

In a digit barring everything can in principle be enabled ("Basic function = enable all") or barred ("Basic function = bar all").

Exceptions to the basic setup are entered in an enabled list or in a barring list.

Digit sequences that are not on the enabled or barring list are either enabled or barred, depending on the basic setup.

A phone number is compared from left to right with the digit sequence of the allocated digit barring.

Example:

- Basic function = enable all
- Digit "6" is entered in the barring list. This digit barring restricts all phone numbers that begin with 6.
- The digit sequence "62" is entered in the barring list. This digit barring only restricts phone numbers that begin with 62.
- The digit sequence "6" is entered in the barring list and the digit sequence "63" in the enabled list. This digit barring restricts all phone numbers that begin with 6, except those that begin with 63.

Number of character strings

Up to 10 character strings can be entered per list.
A character string can consist of up to 20 characters.

Type of characters

Digits: 0, 1 to 9
Characters: *, #, A, B, C, D
Control key, Flash: R

Nesting entries in the enabled and barred lists

Exceptions to a digit sequence barred in the barring list are entered in the enabled list and vice versa. In the example on the left in Fig. 2.68 all phone numbers that begin with the digit sequence "00" are barred except those that begin with "003" or "004". This nesting depth is permitted.

The entry in the example on the right bars all phone numbers that begin with the digit sequence "00" except those that begin with "003" but not with "0031". This nesting depth is not admissible. The entry "0031" is ignored by the system.

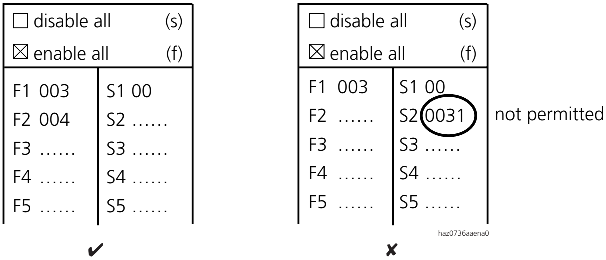


Fig. 2.68: Only one degree of nesting is permitted

Examples of digit barring facilities

A subscriber or subscriber group may only dial the following external destinations:

- Destinations within their own network group
- Destinations of network group 031 and 033
- Destinations in Germany (0049)

The following restrictions also apply:

- No external connections through cost centre selection
- No external connections through route selection

These two restrictions are regulated using the internal barred-code; the others, using the external digit barring:

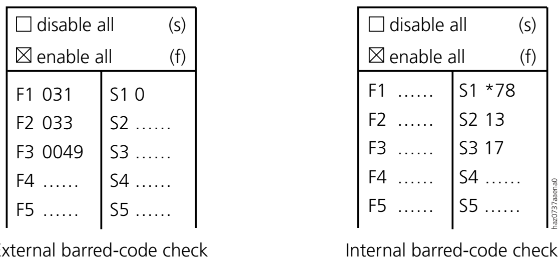


Fig. 2.69: Example of digit barring facilities

In this example the exchange access prefixes are entered as follows in the numbering plan:

- Exchange access for cost-centre selection: 13
- Exchange access for route selection: 17x

Procedure *78 is used to allocate a cost centre using suffix dialling. That is why the digit sequence *78 is also barred.

Initialization Settings

After an initialization a number of digit barring options already have pre-entered digit sequences. They can vary from country to country.

Examples of digit barring initialization values:

- External digit barring 1:
  Internal: All barred except service and emergency numbers.
- External digit barring 2:
  Local: All barred except service and emergency numbers and calls within your own network group.
- External digit barring 3:
  Only domestic calls permitted.
- External digit barring 4:
  Only calls within Europe permitted.
- External digit barring 5:
  All enabled except */# features on the exchange.
- Internal digit barring 5:
  Courtesy operation and remote control of */# procedures are barred
- Internal digit barring 8 (2025 / 2045) and 16(2065): Remote control external barred (ERC *75, *85).

5.3.3 Calls to the public Network

Access to the public network can be obtained with a variety of dialling types:

- Dialling an exchange access prefix
- Dialling an abbreviated dialling number (see page 293)
- Dialling the emergency number (see page 294)

- Dialling via a line key on a key telephone (see page 296)
- Dialling via a line key on an Operator Console (see page 297)
- Dialling the phone number of a virtual network PISN subscriber (see page 297)

Dialling an exchange access prefix

The allocation of prefixes to access types is set out in the numbering plan, where the prefixes can be configured (see "Numbering Plan Identifiers", page 161). Exchange access prefixes are used to dial the following access types:

- Exchange access, Business:
  The call is routed via the route configured for the subscriber.
  The call charges are logged (among others) under Business on the subscriber counter (for more information on call charge allocation see "Individual charge counting or ICC", page 361).
- Exchange access, Private:
  The call is routed via the route configured for the subscriber.
  The call charges are logged (among others) under Private on the subscriber counter.
- Cost centre selection:
  The call is routed via the route configured for the subscriber.
  The call charges are logged (among others) on the counter for the selected cost centre.
- Route selection:
  The call is routed via the route selected by means of a prefix.
  The call charges are logged (among others) under Business on the subscriber counter.

Dialling an abbreviated dialling Number

With an abbreviated dialling number dials the stored phone number. The phone number must have an exchange access prefix.

The digit barring facilities are bypassed. If the call destination for an abbreviated dialling is to be barred using digit barring, the abbreviated dialling number must be entered in the internal digit barring.

The call is routed via the subscriber’s route, provided the stored phone number does not already have a prefix for exchange access with route selection.

The call charges are logged in accordance with the subscriber configuration, provided the stored number does not already have an exchange access prefix that regulates call charge logging (e.g. "Exchange access, Private").

A name can be stored with each abbreviated dialling number, thereby also enabling name dialling.

Dialling the emergency Number

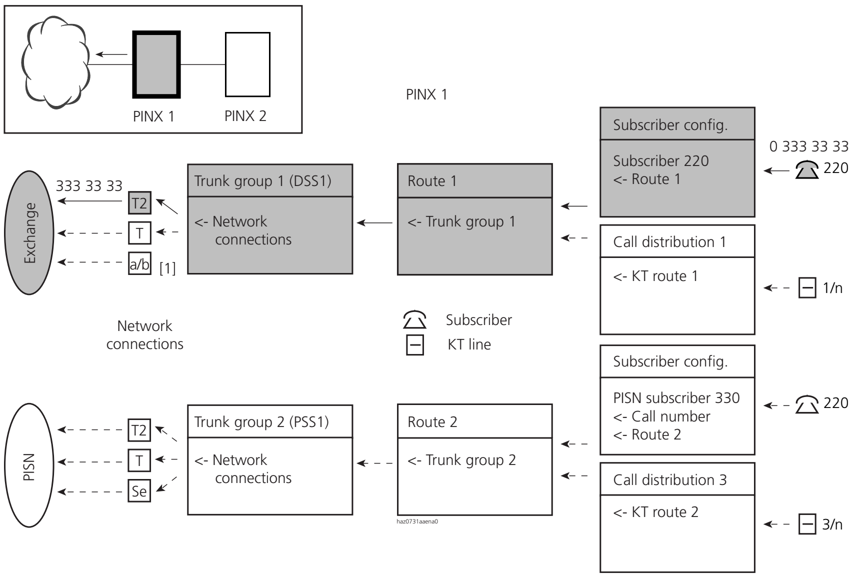
Depending on the switch setting, the emergency number dials one of the three stored phone numbers. The phone numbers must have an exchange access prefix.

The external digit barring is bypassed.

The call is routed via the subscriber’s route, provided the stored phone number does not already have a prefix for exchange access with route selection.

The call charges are logged in accordance with the subscriber configuration, provided the stored number does not already have an exchange access prefix that regulates call charge logging (e.g. "Exchange access, Private").

Routing the call



[1] One and the same trunk group cannot contain both analogue and digital network interfaces.

Fig. 2.70: Routing a call to the public network

Tab. 2.36: Setting the routing parameters

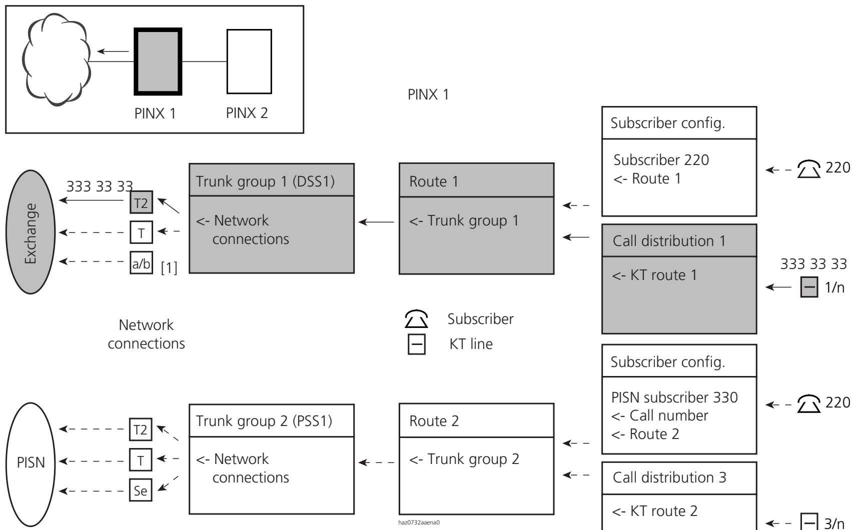
Parameter	Parameter value
Subscriber configuration SC 220: • Route • External digit barring	 1 (route reference number) One digit barring each for switching position 1, 2 and 3
Route 1: • Trunk groups • Outgoing connections • Digit barring • Numbering plan identifier NPI	 1 (reference number of one or more trunk group(s)) Number of connections allowed simultaneously Yes (poll digit barring) E.164
Trunk group 1: • Network interfaces • Network type • Protocol	 Network interfaces of this trunk group Public[1)] DSS1[1)]

1) Not relevant for trunk groups with analogue network interfaces

Call to the public Network via a Key Telephone

Dialling via a line key on a key telephone routes the call via the allocated KT route The KT route is entered in the call distribution element of the KT line.

The call charges can be logged (among others) at the KT cost centre. The KT cost centre is entered in the call distribution element of the KT line (for more information on call charge allocation see page 361).



[1] One and the same trunk group cannot contain both analogue and digital network interfaces.

Fig. 2.71: Routing a call to the public network via a line key of a key telephone

Tab. 2.37: Setting the routing parameters

Parameter	Parameter value
Call distribution element 1: • KT route	 1 (route reference number)
Route 1: • Trunk groups • Outgoing connections • Digit barring • Numbering plan identifier NPI	 1 (reference number of one or more trunk group(s)) Number of connections allowed simultaneously Yes (poll digit barring) E.164

Parameter	Parameter value
Trunk group 1:	
• Network interfaces	Network interfaces of this trunk group
• Network type	Public[1)]
• Protocol	DSS1[1)]

1) Not relevant for trunk groups with analogue network interfaces

Call to the public Network via an Operator Console

Dialling via a line key of Company A routes the call via Route 1.
Dialling via a line key of Company B routes the call via Route 2.

Call to a virtual Network PISN Subscriber

The virtual network PISN subscriber is integrated in the PISN via the public network. The call to a virtual network PISN subscriber is therefore routed via the public network.

The PISN subscriber must be created in the internal numbering plan. The caller dials the PISN subscriber number.

The routing information on the PISN subscribers is allocated to the subscriber configuration and includes the route to be used and the phone number under which the destination subscriber can actually be reached (the phone number is indicated without exchange access prefix). In the following example the PISN subscriber with phone number 440 can be reached in the public network under phone number 333 33 40.

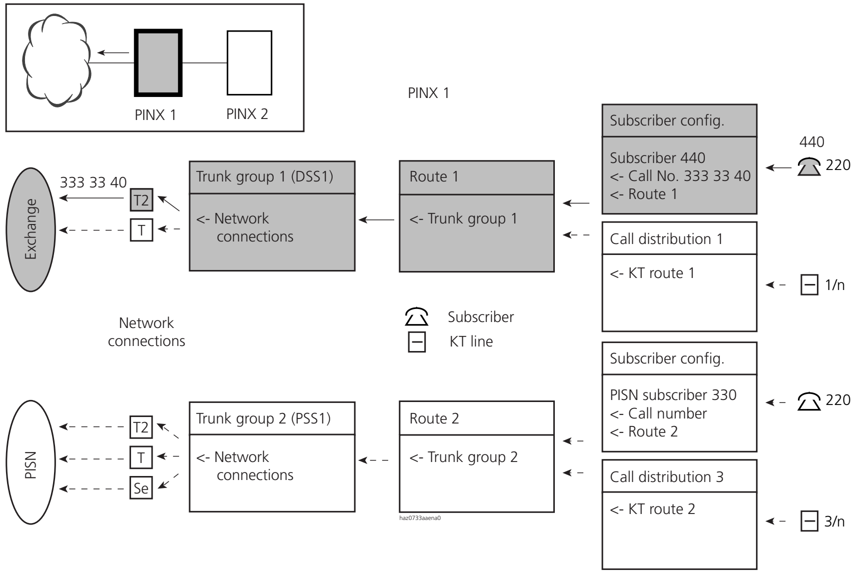


Fig. 2.72: Routing a call to a virtual network PISN subscriber via the public network

Tab. 2.38: Setting the routing parameters

Parameter	Parameter value
Subscriber configuration PISN SC 440: • Route • Number	 1 (route reference number) 333 33 40 (phone number to be dialled, without exchange access prefix)
Route 1 • Trunk groups • Digit barring • Numbering plan identifier NPI	 1 (reference number of one or more trunk group(s)) Yes (poll digit barring) E.164
Trunk group 1: • Network interfaces • Network type • Protocol	 Network interfaces of this trunk group Public DSS1

Exchange access authorization

The authorization to make outgoing calls to the public network can be granted or restricted for each subscriber ("Exchange access authorization" setting in the subscriber configuration).

This setting does not bar dialling into the public network with abbreviated dialling and emergency numbers (see "Bypassing the digit barring", page 289).

Priority exchange allocation

This feature gives individual subscribers preferential treatment when they set up outgoing connections. If a subscriber with priority exchange allocation sets up a connection and all the B channels of the selected route to the network are busy, one of the B channels will be cleared down and made available to the subscriber (subscriber configuration setting: "External priority = emergency).

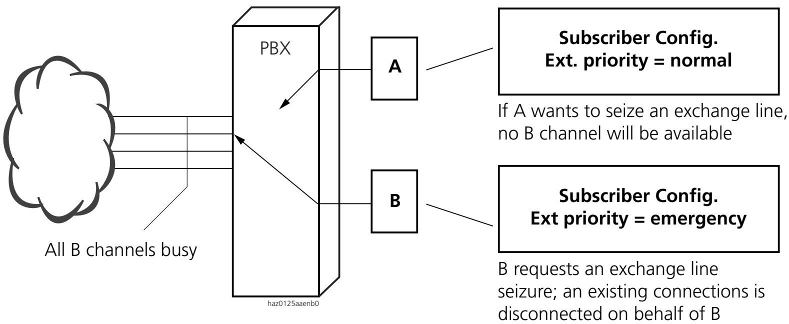


Fig. 2.73: Network access rights for subscribers with and without "Priority exchange allocation"

Example

In the event of an alarm, an alarm system independent of the PBX transmits a message to an alarm headquarters via an ISDN card on an S user-network interface (e.g. a text or a file).

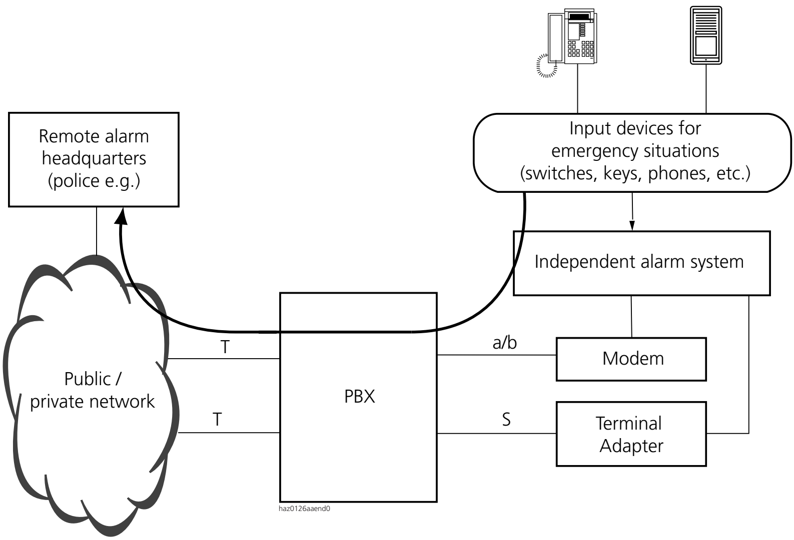


Fig. 2.74: Overview of a configuration for emergency applications

Scope

The priority setting is activated only in the case of direct dialling, not however in the case of call forwarding, CFNR etc

In a private network the prioritization of an outgoing connection is only possible on the PBX connected to the public network (gateway PINX).

In principle all internal subscribers can be defined with "External priority = emergency", even if there are fewer B channels to the public network than authorized subscribers.

Connections seized by subscribers who also have priority will not be cleared down.

Note:
Network interfaces used for external priority calls must be connected with the public network and active. It is advisable to provide a specific network interface for this purpose and to check it on a regular basis.

Initialization setting

On initialization all subscribers are defined without "external priority".

5.3.4 Call to the private Leased-Line Network

The call to a fixed network PISN subscriber is routed via the private leased-line network. The PISN subscriber must be created in the internal numbering plan. The caller dials the PISN subscriber number.

The routing information on the PISN subscribers is allocated to the subscriber configuration and includes the route to be used and the phone number under which the destination subscriber can actually be reached.

Normally a PISN subscriber in the fixed network can be reached directly under his PISN phone number, which means that no other phone number needs to be entered in the subscriber configuration.

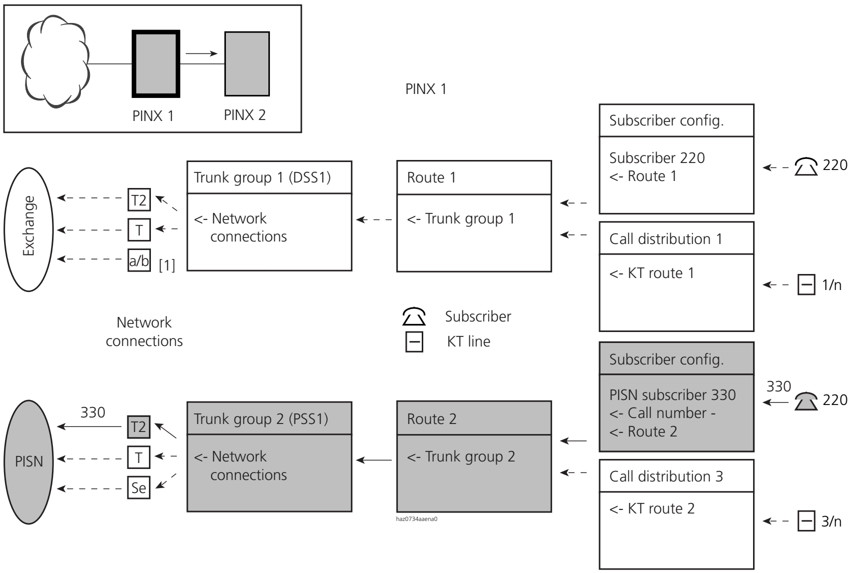


Fig. 2.75: Routing a call to the private leased-line network

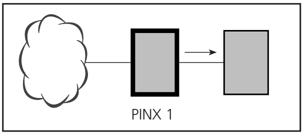
Tab. 2.39: Setting the routing parameters

Parameter	Parameter value
Subscriber configuration PISN SC 330: • Route • Number	(Subscriber PISN) 2 (route reference number) Not relevant in this case
Route 2: • Trunk groups • Digit barring • Numbering plan identifier NPI	 2 (reference number of one or more trunk group(s)) No (do not poll digit barring) PNP
Trunk group 2: • Network interfaces • Network type • Protocol	 Network interfaces of this trunk group Private QSIG or QSIG / PSS1 ISO

5.3.5 Call to a DSS1 Terminal equipment on the S Bus (DDO)

The S external interface can be used to address a terminal equipment that has its own direct dialling plan. The system dials the terminal's end destinations using DDI numbers, which is equivalent to a DDO (direct dialling out) function. A fax server is an example of one such terminal.

A PISN subscriber is created in the PBX for each outgoing direct dial number.



PINX 1

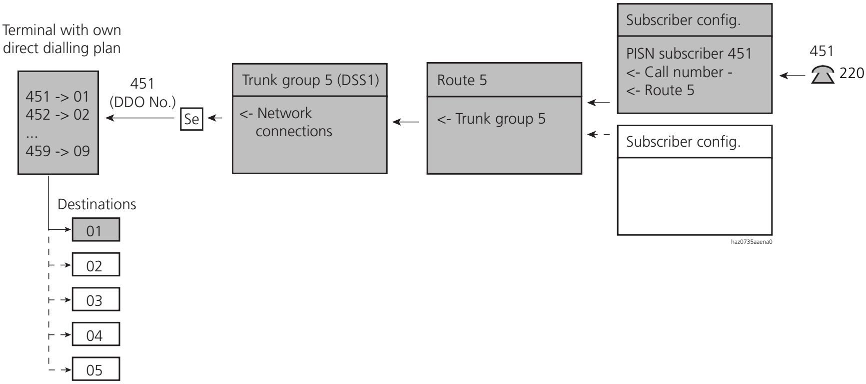


Fig. 2.76: Call to a terminal with its own direct dialling plan

The following services are supported on S external:

- Base Call
- CLIP / CNIP
- Call charge information

Tab. 2.40: Setting the routing parameters

Parameter	Parameter value
Subscriber configuration PISN SC 451: • Route • Number • Numbering plan identifier NPI	 5 (route reference number) – E.164
Route 5: • Trunk groups • Digit barring	 5 (separate trunk group with S external for DDO application) Use or do not use digit barring
Trunk group 5: • Network interfaces • Network type • Protocol	 S external interface Private DSS1

Terminals with a separate direct dialling plan down circuit from an Ascotel IntelliGate system can also be addressed from the public or private leased-line. From a routing technology viewpoint this corresponds to the situation "Routing a call from the public / private network to the PISN" (see also the descriptions as of page 335).

An S external interface (P-P or P-MP) can be used as the network interface.

The call charges are transmitted in ETSI format

5.4 Least Cost Routing (LCR)

Nowadays users usually have several service providers at their disposal to rely on for routing their calls. To ensure that calls are routed as cost effectively as possible, it make sense to select the service provider according to the call destination (e.g. to use a different service provider for long-distance calls compared with local calls).

A service provider will either have his own network or a licence agreement with another network provider. A private leased-line network as defined for the LCR function is a service provider with special characteristics.

In this chapter the term network provider will be used for both network providers and service providers.

5.4.1 Direct or indirect Selection of the Network Provider

The network provider can be selected either manually for each call or automatically using the LCR function.

The network of the required network provider can be reached directly or indirectly from the PBX.

Direct Network Access

The PBX is directly connected with several networks operated by different network providers.

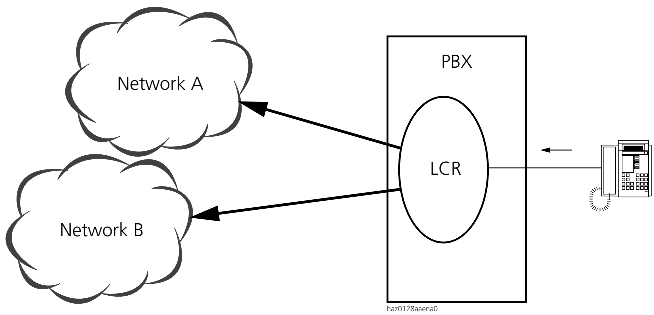


Fig. 2.77: Direct access to network A or B using LCR

Indirect Network Access

The PBX is connected to a specific network (network A). The destination network (network B) is reached indirectly via this network. This case occurs frequently.

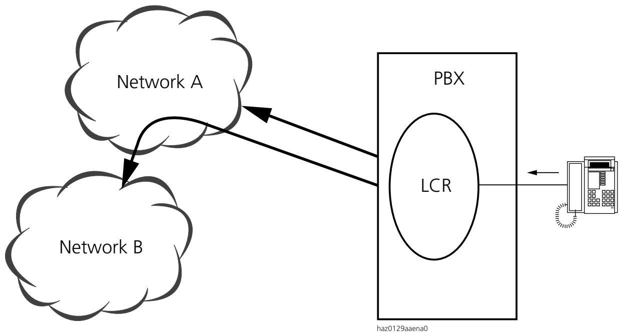


Fig. 2.78: Indirect access to network B via network A using LCR

For indirect access the phone number dialled must contain the following information:

- Phone number of the destination subscriber.
- Network provider required (in the example network provider B).
- The code information (in the example for network provider B) used by B to check whether the caller is a subscriber to his network.

Network provider A can respond to the call in the following way:

- He either routes the destination number on directly using his own numbering plan.
- He takes the call and waits for code information, such as the destination number, to be transmitted by the caller in DTMF mode.

5.4.2 LCR Function

To be able to make outgoing phone calls, a PBX subscriber normally dials an exchange access prefix first.

If the LCR function is deactivated the PBX routes the call in accordance with the exchange access prefix dialled (see "Exchange access authorization", page 299).

If the LCR function is activated and able to analyse the phone number dialled, the phone number will be routed in accordance with the configured LCR criteria. The exchange access prefix is not analysed by the LCR function.

The LCR function can be activated or deactivated throughout the system. If it is activated, it can be deactivated for individual subscribers.

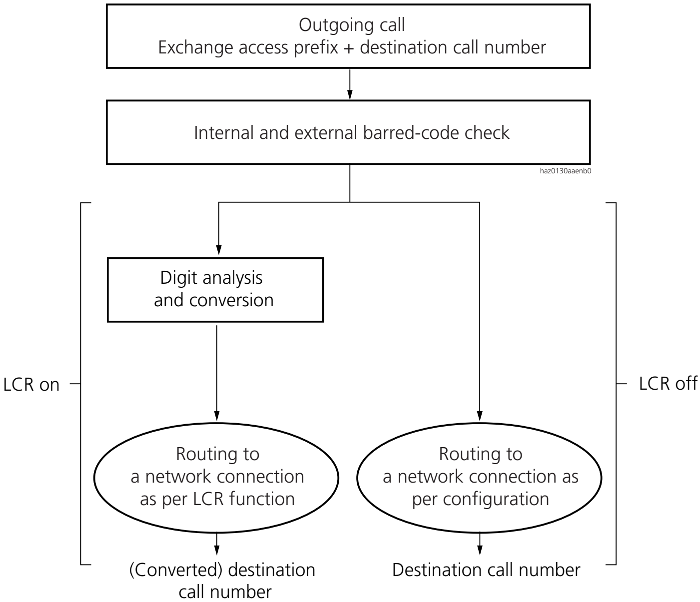
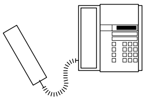


Fig. 2.79: Outgoing exchange traffic using LCR

The call is analysed and routed in three stages:

- Classification of the outgoing call on the basis of the LCR table and allocation to a particular routing table.
- Using the routing table to select a primary and alternative network provider, depending on the time of day and the weekday.
- Network provider-specific conversion of the phone number and routing of the call on the basis of the network provider table.



LCR-Tabelle

Analysable digit sequences	Pointer on routing table
E0	National
E00	International

Routing Table, National

Time zones	Day(s)	Time	Network Operator	Alt. Network Operator
Zone 1	Mon–Fri	08:00–17:30	Provider 1	Provider 2
Zone 2	Sat–Sun	08:00–17:30	Provider 2	Provider 1
Zone 3	Mon–Sun	00:00–07:59	Provider 3	Provider 1
Zone 4	Mon–Sun	17:31–23:59	Provider 3	–

Routing Table, International

Time zones	Day(s)	Time	Network Operator	Alt. Network Operator
Zone 1	Mon–Fri	09:00–18:00	Provider 2	–
Zone 2	Sat–Sun	00:00–23:59	Provider 4	Provider 2

Network Operator Table

Network Oper. Name	Access Code	Route	Barring	Conversion Rule	User ID	PIN	Call charge code length
Provider 1	0512	–	enabled	EUKSN	4321	1234	4
Provider 2	–	2	enabled	–			
Provider 3	–	3	barred	–			
Provider 4	–	4	enabled	<3–>			

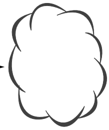


Fig. 2.80: Example of call routing using the LCR function

5.4.3 Allocating the Internal Routing Table (LCR Table)

The LCR table is used to categorize an outgoing call and allocate it to a routing table.
A call is categorized by the evaluation of the phone number digits.

The first digits of an external phone number can be evaluated in terms of the LCR function if they are entered in the LCR table (evaluatable digit sequence) and allocated to a routing table (2nd column). Up to 400 digit sequences can be entered in total in the LCR table.
An analysable digit sequence can consist of up to 10 digits.

Tab. 2.41: Example of an LCR table

Evaluatable digit sequences	Routing tables
E0	National
E00	International
E032	–
E0044	United Kingdom
E0044171938	London South West

Based on the entries in this LCR table, calls are routed as follows:

- In this example phone number 0-061 601 22 22 is routed via the "National" routing table.
- Phone number 0-0033 1 41 23 45 67 is routed via the "International" routing table.
- Phone number 0-032 631 27 17 is routed in accordance with the subscriber configuration (no LCR routing as a routing table was not specified for digit sequence 032).
- Phone number 0-0044 1425 275341 is routed via the "United Kingdom" routing table.
- Phone number 0-0044 171 938 9123 is routed via the "London South West" routing table.
- Phone number 0-631 27 17 is routed in accordance with the subscriber configuration (no LCR routing as the phone number does not contain any analysable digit sequences).

External and PISN-Internal Entries (E and I Prefix)

To indicate whether an entry in the LCR table relates to an external destination in the public network or to a destination in the private leased-line network, the prefix "E" (for external) or "I" (for PISN-internal) must be added to the digit sequence.

Tab. 2.42: Example of an LCR table with a PISN-internal entry

Evaluatable digit sequences	Routing tables
E0	National
E00	International
I62	Region 62

- The external phone number 0-624 38 27 will be routed in accordance with the subscriber configuration (no LCR routing as there is no E entry for digit sequence 62).
- The PISN phone number 62 2020 will be routed via the routing table "Region 62".

Emergency Routing (X Prefix)

If specific phone numbers (e.g. emergency numbers) are to be routed in each and every case (including forced routing) in accordance with the subscriber configuration or subscriber selection and not according to LCR criteria, they must be entered with the prefix "X" in the LCR table. In the "Routing Table" column the system enters the term "Emergency" and a routing table is not allocated.

Example:

- All national calls in Britain are to be routed via network provider A.
- All remaining calls are to be routed via network provider B, reached indirectly, except for the "999" emergency number. This number is to be routed via the settings of the subscriber configuration in all cases.

Tab. 2.43: Example of an LCR table with the prefix X

Evaluatable digit sequences	Routing tables
E0	National
E1	Network group 1
..	...
E9	Network group 9
X999	Emergency

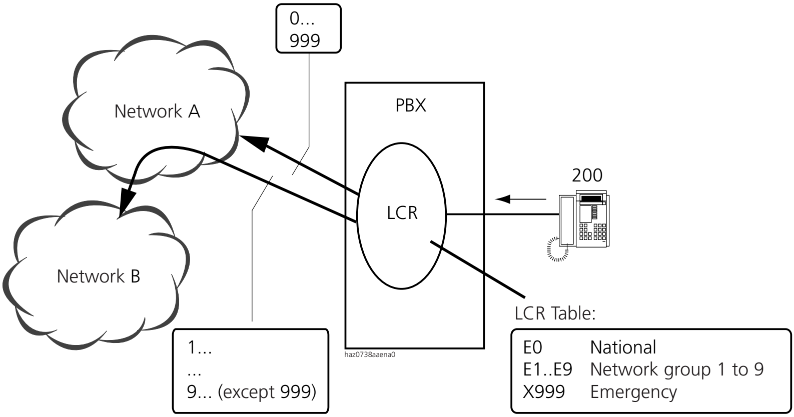


Fig. 2.81: Routing the emergency number 999

If "E999" is entered for the emergency number instead of "X999", an exceptional routing can be configured. The table below shows the routing for prefixes X and E.

Tab. 2.44: Difference in routing with the X prefix and the E prefix

Dialling "999" via the various different exchange accesses	Force network provider is enabled[1]		Force network provider is not enabled	
	X999	E999	X999	E999
Business prefix (0)	SC config	LCR config	SC config	LCR config
Private prefix (10)	SC config	LCR config	SC config	LCR config
Cost centre selection prefix (13n)	SC config	LCR config	SC config	LCR config
Route selection prefix (17x)	Route selection	Route selection	SC config	LCR config
Key telephone line key	KT route	KT route	SC config	LCR config

[1] For more on the subject of "Forcing the network provider", see page 317

Config UT: Routing via route in accordance with the subscriber configuration
Config LCR: Routing via route in accordance with the LCR configuration
Route selection: Routing via manually dialled route
KT route: Routing via the route allocated to the KT line in the call distribution element

5.4.4 Selecting the Network Provider (Routing Tables)

The routing tables are used to select a primary or an alternative network provider for a categorized call, depending on the time of day and the weekday.

A total of 10 routing tables with up to 10 time zones each can be defined.

Tab. 2.45: Example of a routing table

Time zones	Day(s)	Time	Primary Network provider	Alternative Network provider
Zone 1	Mon-Fri.	08:00–17:29	Network provider 1	Network provider 2
Zone 2	Sat-Sun	08:00–17:29	Network provider 2	–
Zone 3	Mon-Sun	00:00–07:59	Network provider 3	Network provider 1
Zone 4	Mon-Sun	17:30–23:59	–	Network provider 1

Depending on the current zone a call will be routed to one of the following network providers:

- Primary network provider
- Alternative network provider (alternative routing)
- Network provider in accordance with the subscriber-specific routing (subscriber configuration)

The criteria for selecting one of these network providers are shown in Tab. 2.46.

Tab. 2.46: Selection of the network provider depending on settings and situation

Settings in the routing table		Response of the LCR function
Primary Network provider	Alternative Network provider	
Network provider 1	-	Routing to network provider 1; if this is not possible, routing in accordance with the subscriber configuration.
Network provider 1	Network provider 2	Route to network provider 1; if this is not possible, alternative routing to network provider 2
–	Network provider 2	Routing in accordance with the subscriber configuration; if this is not possible, alternative routing to network provider 2.
–	–	Routing in accordance with the subscriber configuration

If neither the network provider selected initially nor the alternative network provider is available, the call will be cleared down. The caller will obtain the congestion tone.

Automatic alternative routing can be activated or deactivated throughout the system.

Time zones

The time zones are used to allocate network providers depending on the time of day. This means it is possible to take account of the fact that network provider 3 for example is more cost effective at night than network provider 2

If the time at which a connection is set up is outside the defined time zones the call is routed in accordance with the subscriber configuration (without LCR function).

If the time indications of several time zones overlap, the time zone placed further up in the table applies to the area of overlap:

Tab. 2.47: Example of overlapping time zones

Time zones	Day(s)	Time	Primary Network provider	Alternative Network provider
Zone 1	Mon-Fri	07:00-16:59	Network provider 1	Network provider 2
Zone 2	Mon-Sun.	00:00-23:59	Network provider 2	–

Tab. 2.48: Zone 1 applies in the overlap area

Time	12:00:00 AM to 06:59:00 AM	07:00:00 AM to 04:59:00 PM	05:00:00 AM to 11:59:00 PM
Zone 1		Network provider 1	
Zone 2	Network provider 2		Network provider 2

Alternative Routing (Fallback Routing)

If the LCR function realizes that access to the network provider initially selected is not possible, a call is routed to the alternative network provider and an event message is generated ("LCR via alternative network provider").

The LCR function recognizes that access to a network provider is not possible if,

- all the B channels in the selected route are busy,
- routing via the primary network provider is barred in the network provider table,
- the network signals to the PBX that the primary network provider is not available (e.g. due to overloading).

Manual alternative Routing

In some situations the LCR function cannot recognize that the primary network provider is not available (for example if the network provider answers the call with a voice message). The user then has the possibility to dial via the alternative network provider manually. To do so he interrupts the connection and dials *90. The number is then redialled in the same way as with a last-number redial but this time via the alternative network provider.

If the user routes a call via the alternative network provider manually, no event message will be generated.

If users are not to be authorized to dial the alternative network provider themselves, *90 should be barred in the internal digit barring.

Manual alternative routing also works if automatic alternative routing is not activated.

Restricted scope of performance by a Network Provider

Not all network providers offers every service (voice, fax, data traffic, etc.) If for example the network provider table contains network providers that can only transfer voice service, subscribers will have to manually force the data service-compatible network provider they want when setting up data connections (see "Bypassing LCR manually (Forced Routing)", page 317).

5.4.5 Conversion and Routing (Network Provider Table)

The phone numbers are converted specifically for each network provider based on the network provider table; the call routing is then determined. 10 network providers can be entered in the table.

Tab. 2.49: Network operator table

Network provider	Access code	Route	Barring	Conversion rule	User ID	PIN	Charge code length
Network provider 1	0512	–	unassigned	EUKSN	4321	1234	3
Network provider 2	–	2	unavailable	–			
Network provider 3	–	3	unassigned				

Settings of the network provider table:

- Access code:
  Used for indirect access to a network provider. For direct access to a network provider, indicating a route is sufficient.
  Maximum access code length: 10 digits.
- Barring:
  Enable or bar call routing to the corresponding network provider.
- User-ID / PIN:
  syntax and length depend on the network operator.
- Call charge code length (single digit: "1" – "9"):
  reduces the call charge code called up in the conversion rules to the specified length, starting from the end. Example:
  – In the conversion rule the subscriber number is called up as a call charge code.
  – The call charge code length is set to "3".
  – Subscriber number 3426 is transmitted as call charge code 426.

Conversion Rules

The conversion rules specify how a dialled phone number is to be converted to enable automatic access to a network provider.

Tab. 2.50: Conversion rule parameters

Parameter	Meaning
E	Add access code
"0"–"9", "*", "#"	Add specified characters
N	Add dialled phone number
<x–y>	Add digit x to digit y to the phone number
Z	Switch over to DTMF
Pn	Pause (n = 1-9 [seconds])
U	Add user ID
K	Add PIN (Personal Identification Number)
S	Add subscriber as call charge code ("S" or "C" only)
C	Add cost centre as call charge code ("S" or "C" only)

x- defines the start position for creating the substring;
if x is not specified, 1 is considered as the start position.
-y defines the end position for creating the substring;
if y is not specified, the last digit of the number is considered as the end position.
x / y If x or y only is specified without separator, the designated position applies.

Tab. 2.51: Examples for parameter <x–y>

Parameter	Meaning
<2–4>	3 digits from the second position of the number dialled
<3–>	All the digits from the third position to the end (corresponds to <3–.>)
<–5>	The first 5 digits (corresponds to <1–5>)
<3>	The third digit only (corresponds to <3–3>)
<.>	The last digit only
<1–>	The entire number (corresponds to <1–.> and N)

A conversion rule can have up to 20 characters in total. The result string generated from the conversion rule must not exceed a maximum of 40 characters.

Examples Relating to the conversion Rules

Access code for network B via network A: 132
Subscriber dials: 0-0 1222 774518
User-ID: 26013
PIN: 7725

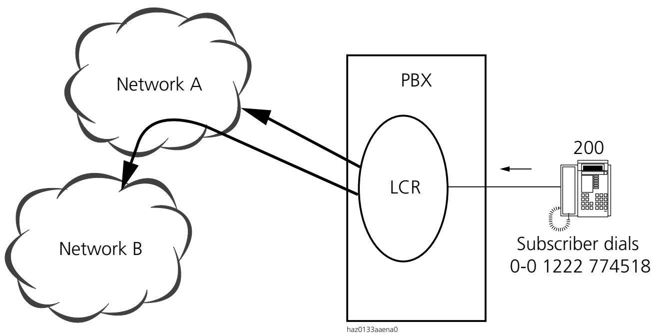


Fig. 2.82: Reference illustration for the following examples

Tab. 2.52: Table with examples of conversion rules and phone numbers converted accordingly

Rule	Conversions	Result string
EN	Access code + number dialled	13201222774518
E<3–>	Access code + all the digits of the dialled number from the third position onwards	132222774518
<1>E<2–>	1. First dialled digit + access code + second to last dialled digits	01321222774518
00EN	00 + access code + number dialled	0013201222774518

Rule	Conversions	Result string
EZP2<3–>#	Access code, DTMF dialling, 2 s pause + third to last dialled digit + #	132222774518#
EZUP2N	Access code, DTMF dialling, User ID, 2 s pause, phone number	1322601301222774518
EZUKSN	Access code, DTMF dialling, User ID, PIN, SC No. as call charge code, phone number	13226013772520001222774518

Digits dialled in DTMF mode are underlined (e.g. 22277458#).

5.4.6 Bypassing LCR manually (Forced Routing)

A user may be authorized through the subscriber configuration to determine the network provider himself by bypassing the LCR settings ("Force route = yes").

Depending on whether the network provider he wants is connected directly or indirectly, the user will add to the phone number either a route prefix or the prefix of the network provider he wants.

Directly connected Network Provider

With route selection the user can dial into the network of a directly connected network provider (direct access).

Calls with other exchange access prefixes will be routed via the LCR function even if authorization is enabled (Tab. 2.53).

Tab. 2.53: Call routing to a directly connected network provider

Exchange access prefix	Force network provider is enabled	
	No	yes
Business (0)	LCR routing	LCR routing
Private (10)	LCR routing	LCR routing
Cost centre selection (13n)	LCR routing	LCR routing
Route selection (17x)	LCR routing	Routing in accordance with route selection

Indirectly connected Network Provider

If the network provider the user wants is not connected directly (indirect access), the user dials as a prefix the number required or the necessary code.

Tab. 2.54: Call routing to an indirectly connected network provider

	Force network provider is enabled	
	No	yes
User dials network provider number or code	LCR routing	Routing as per user's choice

5.4.7 LCR with Key Telephones

LCR routing when dialling via the line keys depends on the "Force route" authorization.

- "Force route" enabled:
  Routing is effected via the KT route as with a deactivated LCR function.
- "Force route" not enabled
  Routing is effected via the LCR function.

5.4.8 LCR in the private Leased-line Network

Where the LCR function is concerned, a private leased-line network (PISN) is a special network provider, characterized as follows:

- A PISN is usually reached directly (see "Direct Network Access", page 304).
- Digit sequences of PISN-internal phone numbers must be entered with the I-prefix in the LCR table (see "External and PISN-Internal Entries (E and I Prefix)", page 309).
- Overflow routing from the PISN to the public network is implemented with the LCR function by entering the PISN as the primary network provider and the public network provider as the alternative network provider. Unlike fallback routing, routing to the alternative network provider does not generate an event message (see also "Alternative Routing (Fallback Routing)", page 313).

5.4.9 Call logging and Data Protection

In connection with the LCR function, the OCL output format PC5 (recommended) or PC4 must be used (see "Output formats", page 385).

When the data protection function is activated, the following data will not be output or output only in part, in OCL output format PC5 and PC4:

- The last four digits of the phone number dialled by the subscriber will be truncated.
- The last four digits of the phone number dialled by the LCR function will be truncated.
- User IDs and PIN codes will not be output.
- User IDs and PIN codes will also be suppressed when the LCR tables are printed out.

5.4.10 Examples of LCR

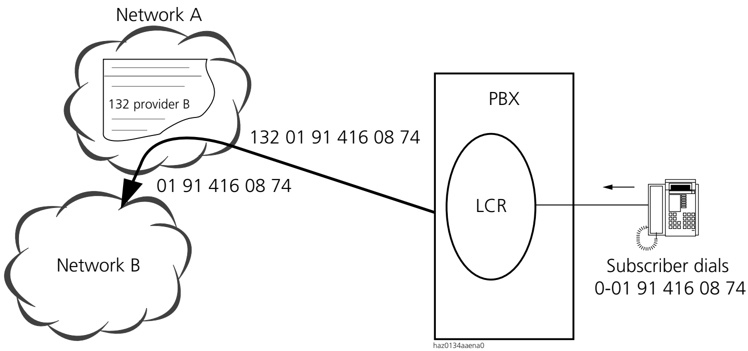


Fig. 2.83: 1. Example: Network provider B is integrated in the numbering plan of network provider A

Tab. 2.55: 1. Example: Entry in the network provider table

Network Provider	Access code	Route	Barring	Conversion rule	User ID	PIN	Charge code length
Network provider B	132	–	–	EN	–	–	–

1. Stage:

- The system reaches network provider B via network provider A
- Network provider B seizes and the connection provider B – PBX is set up

2. Stage:

The system transmits the phone number in DTMF mode in accordance with the configured conversion rule.

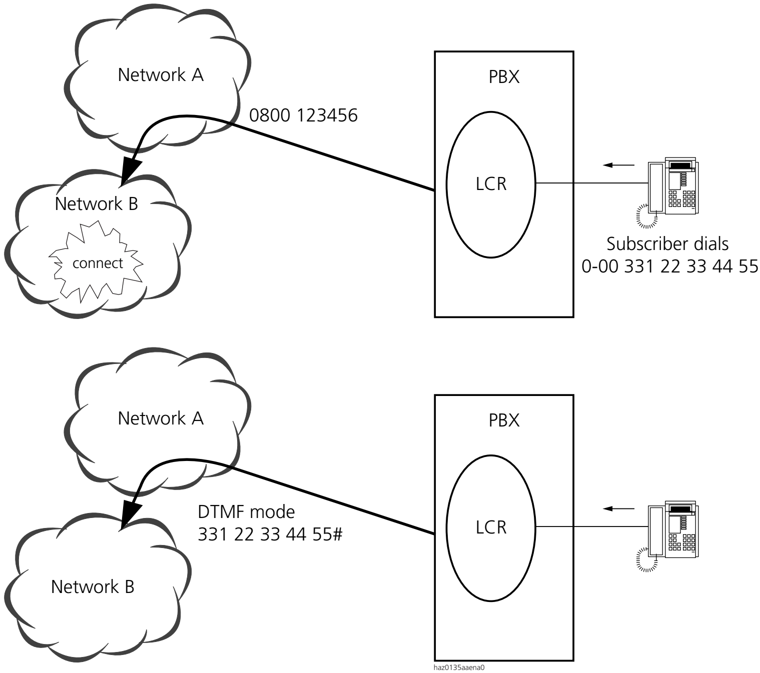


Fig. 2.84: 2. Example: Network provider B is not integrated in the numbering plan of network provider A

Tab. 2.56: 2. Example: Entry in the network provider table

Network Provider	Access code	Route	Barring	Conversion rule	User ID	PIN	Charge code length
Network provider B	0800123456	–	–	EZ<3->#	–	–	–

5.4.11 Higher-Level LCR Settings

The table below summarizes once again the higher-level LCR settings.

Tab. 2.57: LCR settings

Parameter	Parameter value	Remarks
Least Cost Routing (Account Manager):		
• LCR	On / Off	Activate / deactivate LCR function throughout the system (see page 306)
• Alternative Routing	On / Off	Activate / deactivate alternative routing throughout the system (see page 313)
Subscriber configuration:		
• LCR	On / Off	Activate / deactivate LCR function for a specific subscriber (see page 306)
• Force routing (LCR)	Yes / No	Bypass LCR manually (see page 317)
• Internal digit barring	Bar *90	Bar manual alternative routing (see page 313)

Initialization Settings

After initialization the LCR function is deactivated.
When activating the LCR function after initialization, automatic alternative routing is activated.

5.5 Exchange-to-Exchange Traffic

Exchange-to-exchange traffic covers all interactions involving at least 2 subscribers in the public network and at least 1 local subscriber in the PBX (internal subscriber).

5.5.1 Exchange-to-Exchange Connections

In an exchange-to-exchange connection two seized exchange lines to the public network are connected with each other locally in the PBX.

Restrictions applicable throughout the system

Exchange-to-exchange traffic can be restricted or barred throughout the system. The setting is not effective for inter-network connections to the public network or to on one side only, e.g. PISN-PISN or PISN-exchange.

The system supports exchange-to-exchange traffic on both digital and analogue network interfaces. The following settings are possible:

- "Not allowed": No exchange-to-exchange connections allowed
- "Only digital-digital": Both network interfaces must be digital
- "Also digital-analogue": At least one network interface must be digital
- "Also analogue-analogue": Both network interfaces can be analogue

If sections of exchange-to-exchange connections are analogue, the transmission quality will decrease.

If a user tries to set up an inadmissible exchange-to-exchange connection (e.g. by initiating an exchange enquiry call and then hanging up), the second connection is disconnected and subscriber B obtains long ringing after hanging up, to be able to answer the first connection on hold. This is the case for example if one or both network interfaces are analogue and the parameter is "Exchange-to-exchange connection" = "Only digital-digital".

Tip:
In some countries private PBX operators are not authorized to transfer an outside call back to the public network. Explain the situation regarding operating rights to the PBX operator during the negotiation stage.

Subscriber-specific configuration

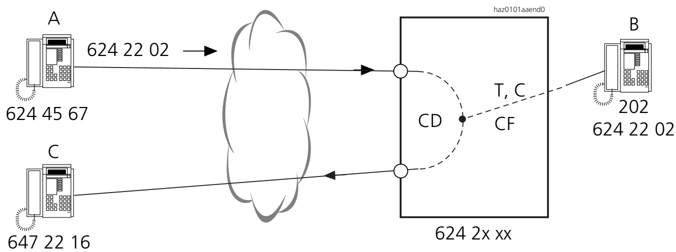
The settings described in the last section can also be configured individually for each subscriber. The subscriber-specific configuration takes priority over the setting for the system as a whole. If a subscriber's settings are not to deviate from the settings made for the system as a whole, the parameter has to be configured to "According to exchange settings" (initialization value).

Specially configured abbreviated dialling numbers

Exchange-to-exchange traffic can be enabled in general for specially configured direct dialling numbers ("Exchange-to-Exchange connection = Yes"). This allows all types of exchange-to-exchange connections, and is also valid in cases where exchange-to-exchange traffic is barred in the system configuration **and** the subscriber-specific configuration. The abbreviated dialling number stored does not have to be complete, which means digits can be suffix-dialled manually. This allows for example exchange-to-exchange traffic to be enabled for an entire office branch using a single abbreviated dialling number.

5.5.1.1 Setting up Exchange-to-Exchange Connections

An exchange-to-exchange connection can be set up using Call Forwarding Unconditional, Conference, Call Forwarding on No Reply, Call Deflection and Transfer with or without prior notice.



V Transfer
C Conference
CFU Call Forwarding Unconditional
CFNR Call Forwarding on No Reply
CD Call Deflection

Fig. 2.85: Exchange-to-Exchange Traffic

5.5.1.2 Clearing down Exchange-to-Exchange Connections

Digital-Digital (D-D):

The public network sends the PBX a release signal once the external call partners of an exchange-to-exchange connection have finished the call. The connection is then cleared down by the PBX.

Without a release signal, the PBX cannot clear down an exchange-to-exchange connection.

The amount of time between completion of the call and the sending of the release signal depends on whether the exchange-to-exchange connection is set up end-to-end within the ISDN network (end-to-end ISDN connection) or whether sections of it are analogue (non-end-to-end ISDN connection).

At transitions to other networks (for example from leased-line-network to cellular network) it is possible, due to the lack of correct signalling, that an end-to-end ISDN connection is signalled as a non-end-to-end connection.

End-to-End ISDN Connection

The release signal is sent as soon as the call is completed.

Non-End-to-End ISDN Connection

With non-end-to-end ISDN connections the amount of time between the completion of the call and the release depends on who set up the connection:

- If the PBX subscriber set up the connection (i.e. from the PBX's viewpoint an outgoing call), and the external partner (subscriber C in Fig. 2.85) hangs up, it can take a few minutes for the release signal to be sent.
- If one of the external partners set up the connection (i.e. from the PBX's viewpoint an incoming call) and the external partner (subscriber B in Fig. 2.85) hangs up, the release signal is sent immediately.

Note:
If two announcement services such as sports and weather information are connected with each other, this exchange-to-exchange connection will not be cleared down automatically This can lead to high call charges.

Each exchange-to-exchange connection will be cleared down by the PBX after two hours.

Note:
If an exchange-to-exchange connection is transferred to the exchange using Partial Rerouting or Call Deflection, the PBX no longer has any control over the connection and therefore cannot disconnect it.

Analogue-Analogue (A-A) or Digital-Analogue (D-A)

- Release on the analogue interface cannot be guaranteed with these connection types. On analogue network interfaces the PBX only recognises loop interruptions but not polarity reversals, busy tones, etc., as a release criterion. Detection of the loop interruptions can be activated or deactivated throughout the system.
- Any exchange-to-exchange connection is disconnected at the latest after 2 hours.
- The maximum duration of an analogue exchange-to-exchange connection can be further restricted for the A-A connection type (1...120 minutes).

Note:
As release cannot be guaranteed for connection types D-A and A-A, unintentional high costs can occur. What's more the national guidelines and regulations should be observed before enabling these connection types.

System configuration

Tab. 2.58: Exchange-to-exchange connections: System configuration

Parameter	Parameter value	Remarks
Exchange settings: • Exchange-to-exchange connection • Disconnect timeout • Wait for connection	 Not allowed / Only digital-digital / Also digital-analogue / Also analog-analog <1...120 minutes> Yes / No	 Throughout the system applies only to connection type "Analogue-Analogue" For description see page 442
Subscriber settings: • Exchange-to-exchange connection	 Not allowed / Only digital-digital / Also digital-analogue / Also analog-analog / According to exchange settings	 Priority over the setting for the system as a whole
Abbreviated dialling settings: • Exchange-to-exchange connection	 Yes / No	 Priority over the exchange setting for the system as a whole **and** the subscriber-specific setting

5.5.1.3 Possible Exchange-to-Exchange Connections

The following PBX features can be used to set up exchange-to-exchange connections:

- Call Forwarding Unconditional
- Call Forwarding on No Reply
- Call Deflection
- Switching calls
- Conference circuit

The following tables and examples illustrate which features are available in which situations.

Connecting an incoming call with an outgoing call

An incoming call is diverted to the public network, forwarded on or connected in a conference.

Tab. 2.59: Features supported

Subscriber A	→	Call Forwarding Unconditional Call Forwarding on No Reply Call Deflection Switching calls Conference circuit	→	Subscriber C

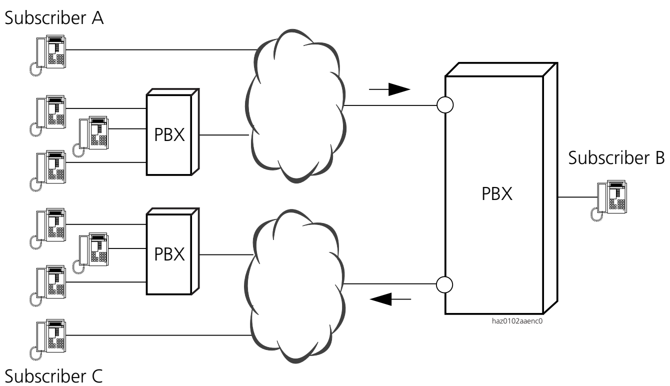


Fig. 2.86: Connecting an incoming call with an outgoing call

See also:
" "Wait for connection" setting", page 442.

Connecting two outgoing calls

This situation occurs for example

- when setting up a conference when both conference parties are called.
- when the attendant sets up a connection for a member of staff, then calls him back and transfers the call.

Tab. 2.60: Features supported

Subscriber A	◄──	Switching calls Conference circuit	──►	Subscriber C

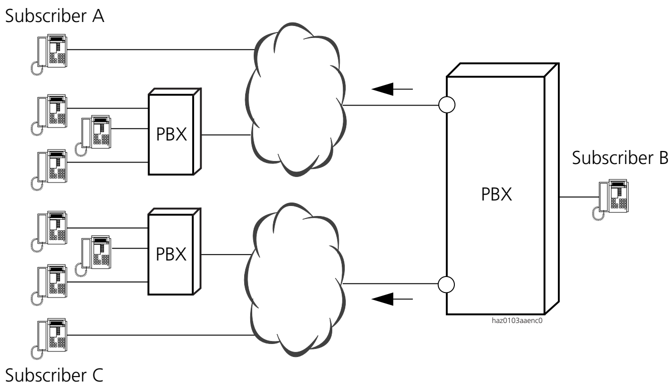


Fig. 2.87: Connecting two outgoing calls

Two incoming calls

The B channels of two incoming calls can be connected with each other via a conference circuit or by a normal call handover by going on-hook (transfer).

Tab. 2.61: Features supported

Subscriber A	──►	Conference circuit Switching calls	◄──	Subscriber C

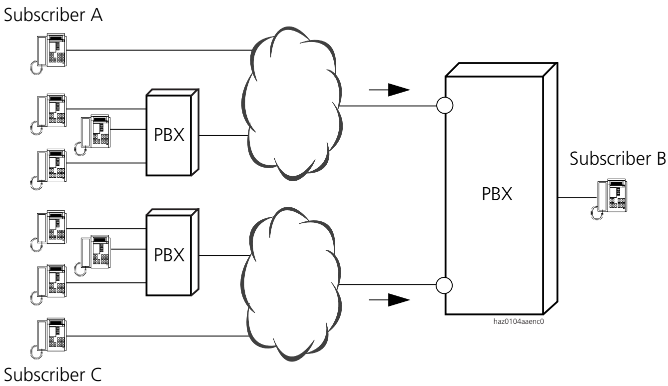


Fig. 2.88: Connecting two incoming calls

Preventing pointless Exchange-to-Exchange Connections

To prevent exchange-to-exchange connections being set up with announcement services or with special numbers (e.g. info boxes), the numbers concerned should be barred in the digit barring.

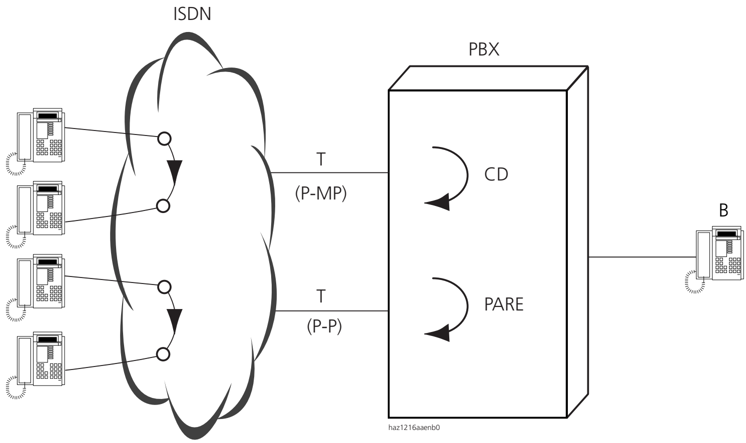
5.5.2 Transferring Call Forwarding Unconditional to the Exchange

Internal subscribers can have their terminal diverted to external destinations When an external subscriber calls the destination diverted externally, the exchange-to-exchange connection created occupies two B channels.

The system can be configured so that such call forwarding are transferred from the PBX to the public network, thus freeing the two B channels. To do so, the system automatically activates the supplementary services partial rerouting (in point-to-point operation) and call deflection (in point-to-multipoint operation).

The subscribers themselves are not aware of this process.

The called subscriber in the public network is presented with the caller's CLIP as well as information on who redirected the call.



PARE Partial Rerouting
CD Call Deflection
P-P Point-to-point operation
P-MP Point-to-multipoint operation

Fig. 2.89: Transferring Call Forwarding Unconditional to the Exchange

Call Deflection

Call deflection (CD) is a supplementary service for ISDN subscribers and is available only on a point-to-multipoint connection. Call deflection can be used to reroute a call during the ringing phase. The feature is also provided at the user-network interface (see "Deflecting a call during the ringing phase (CD)", page 448).

Partial Rerouting

Partial rerouting (PARE) is a supplementary service for PBX operators and is available only on a point-to-point connection (basic and primary rate access).

Call Forwarding Procedure

Call Forwarding Unconditional is transferred to the exchange as follows (Fig. 2.89):

- Subscriber B activates a Call Forwarding Unconditional to subscriber C.
- Subscriber A calls subscriber B.
- The PBX carries out the Call Forwarding Unconditional locally in the PBX. Two B channels are busy. 2 channels B are busy

- The PBX activates PARE or CD at the public network provider.
- The network provider takes charge of the Call Forwarding Unconditional; the 2 B channels are freed.
- Subscriber C is called. He is presented with subscriber A's phone number and the redirecting information by way of CLIP. At the same time the redirecting information is also transmitted back to subscriber A (see "Display for Call Forwarding Unconditional", page 191).

Call charges:

- Subscriber pays the call charges up to the call forwarding location in the network.
- Subscriber B pays the call charges from the call forwarding location to subscriber C.

Call Forwarding Functions Supported

The system routes the following call forwarding to the exchange:

- Call Forwarding Unconditional (CFU)
- Call Forwarding Busy (CFB)
- Call Forwarding on No Reply (CFNR)
- Call Deflection (CD) by a subscriber (forwarding a call during the ringing phase)

With all call forwarding functions the call only continues ringing at subscriber C once it has been forwarded to the exchange.

Requirements

Call forwarding to the exchange is subject to the following requirements:

- ISDN network interfaces T/T2 (QSIG and analogue are not supported).
- In point-to-point operation the supplementary service partial rerouting must be available (subscription may be required).
- In point-to-multipoint operation the supplementary service call deflection must be available (subscription may be required).
- Subscriber B must be defined as a "Subscriber"-type individual destination in the call distribution element used by subscriber A to make his call.
- The relevant authorizations must be enabled.
- The call number of the external call forwarding destination must not be entered as an analysable digit sequence in an LCR table if LCR is active.

System configuration

Tab. 2.62: Transferring Call Forwarding Unconditional to the exchange: Settings

Parameter	Parameter value
Subscriber configuration: • Exchange access authorization • Rerouting in the Exchange	 yes yes
Trunk group configuration: • Rerouting in the Exchange • PSTN supports "Identity of charge" • Network type • Protocol	 yes yes[1)] Public DSS1
Call distribution element: • Destination	 Subscriber

[1)] If the parameter configuration is "Yes", the PBX also sends the call charge identity when call forwarding to the exchange is transferred. This ensures that call charge information is correctly logged in the PBX. The parameter setting depends on whether or not the network operator supports "Charge Identity".

5.5.3 Three-Party Connections in the Exchange

A locally implemented three-party connection with two external subscribers takes up two B channels.
In point-to-multipoint operation the system can be configured so that the node of such a three-party connection is transferred from the PBX to the public network, thereby freeing up at least one B channel and PBX resources. To do so the system accesses the supplementary services of the network provider.

The subscribers themselves are not aware of this process.

The following PBX features can be transferred to the exchange:

Tab. 2.63: Supplementary services take charge of features transferred to the exchange

PBX feature	Supplementary service	Feature description
Hold	Hold	see page 460
Enquiry call	Inquiry Call	see page 460
Brokering	Brokering	see page 462
Call transfer (with or without prior notice)	Explicit Call Transfer	see page 467
Call back (only after call transfer with prior notice)	Recall	see page 504
Three-party conference	Three-Party Conference	see page 464

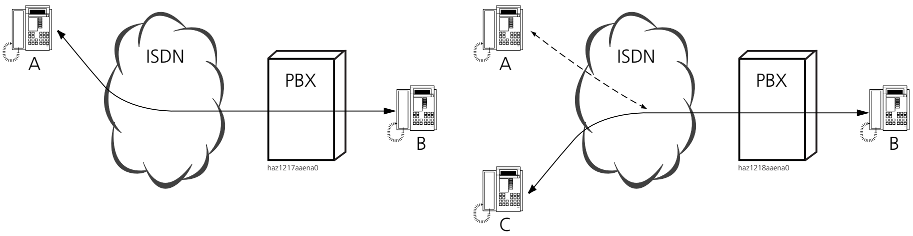


Fig. 2.90: External connection followed by hold and enquiry calls

Description of the Procedure

Calls on hold in the exchange (Fig. 2.90):

- Subscriber is through to subscriber B.
- Subscriber B puts subscriber A on hold: The call is put on hold locally in the PBX.
- Subscriber B calls subscriber C: As soon as subscriber B dials the external phone number; the PBX transfers the locally held call to the exchange by activating the hold supplementary service with the network provider.

All the other three-party connections can be set up from this situation. Example with brokering:

- Subscriber A is on hold in the exchange
- Subscriber B is through to subscriber C
- Subscriber B brokers to subscriber A:
  As subscriber A in the exchange is on hold, the PBX itself does not broker; instead it requests the network provider to do so (by sending "hold" for subscriber B and "retrieve" for subscriber A).

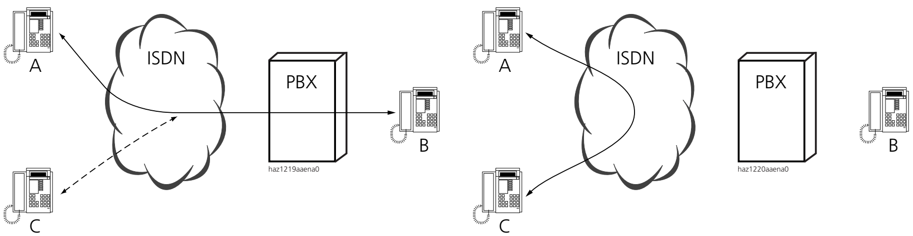


Fig. 2.91: Brokering followed by call transfer

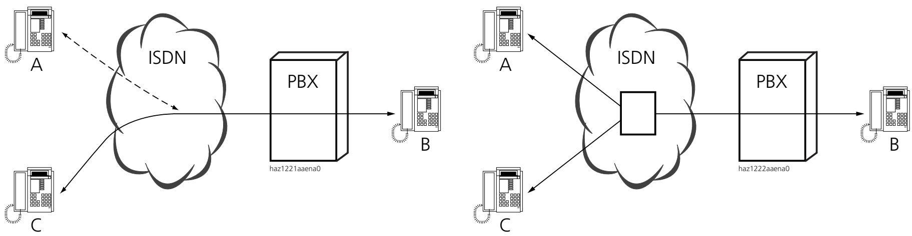


Fig. 2.92: Enquiry call and brokering, followed by three-party conference

Requirements

The following requirements have to be met for three-party connections in the exchange to be activated:

- Basic accesses in point-to-multipoint operation (DSS1 only; QSIG and analogue not supported).
- For Italy only: Basic accesses in point-to-point operation (DSS1 only; QSIG and analogue not supported).
- The supplementary services required must be available at every basic access at which the function is to be supported (subscription may be required).
- The enquiry call connection must be set up as an outgoing call by the internal subscriber. It has to be routed via the same basic access as the first connection.
- Authorisations must be enabled (see "System Configuration", page 334).

PBX Response if the Procedure Fails in the Exchange:

Hold cannot be transferred to the exchange:

- The call is put on hold in the PBX.
- Any subsequently initiated three-party connections are carried out locally in the PBX.

Three-party conference / call transfer in the exchange not carried out:

- The PBX cannot carry out the function locally as the call is on hold in the exchange. The function cannot be carried out.

Tab. 2.64: Transferring three-party connections to the exchange: Settings

Parameter	Parameter value
Subscriber configuration: • Exchange access authorization	 yes
Network interface: • TEI management	 P-MP
Trunk group configuration: • Transfer hold to the exchange • Three-party conference in the exchange • Call transfer in the exchange • Network type • Protocol • Trunk connections	 yes yes yes Public DSS1 Group in the same trunk group all the basic accesses that are to support the function

5.6 Transit Routing in the Private Leased-Line Network

When a PINX forwards a call on the network side, it is a transit routing.

If a PINX routes a call from the public network to the private leased-line network or vice versa, it assumes a gateway function. It therefore acts as the gateway PINX for the call.

If a PINX routes a call from a PINX in the private leased-line network to another PINX in the private leased-line network, it assumes a transit function. It therefore acts as the transit PINX for the call.

In this chapter you will find out how Ascotel IntelliGate resolves the gateway and transit function , and the settings required.

Note:
A transit call must never be routed from network to network via the same trunk group; otherwise this may lead to endless loops and block all available B channels.

5.6.1 From the Public Network to the Private Leased-Line Network

Routing with Direct Dialling

It is advisable to create direct dial numbers at the gateway PINX for all PISN subscribers. An incoming call from the public network will then be routed on into the private leased-line network in accordance with the information relating to the dialled PISN subscriber.

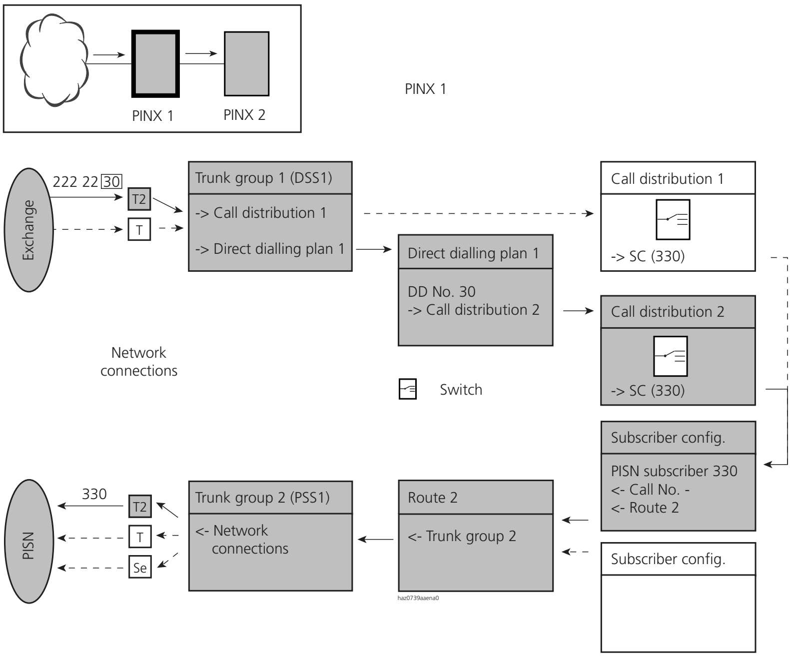


Fig. 2.93: Transit routing from the public network into the private leased-line network with direct dialling

Tab. 2.65: Routing parameter settings

Parameter	Parameter value
Trunk group 1: • Network interfaces • Incoming connection • Network type • Protocol • Direct dialling plan • Call Distribution Element	 Network interfaces in this trunk group Number of connections allowed simultaneously Public DSS1 1 (number of a direct dialling plan) 1 (relevant only if no suitable DD number is found)
Direct dialling plan 1: • DD number 30	 2 (reference number of a call distribution element)
Call distribution element 2: • Destinations • Incoming connections	 Switching position 1: 330 (PISN Subscriber) Number of connections allowed simultaneously
Subscriber configuration PISN SC 330: • Route • Number	 2 (route reference number) Not relevant in this case
Route 2: • Trunk groups • Digit barring • Outgoing connections • Numbering plan identifier NPI • Type of number TON	 2 (reference number of one or more trunk group(s)) Use or do not use digit barring Number of connections allowed simultaneously PNP Unknown
Trunk group 2: • Network interfaces • Network type • Protocol	 Network interfaces of this trunk group Private QSIG or QSIG / PSS1 ISO

Routing without direct dialling

An incoming call from the public network is routed on to the private leased-line network in accordance with the information relating to the PISN subscriber allocated via the call distribution element.

This is useful in only a few instances since all the calls are routed via the same call distribution element.

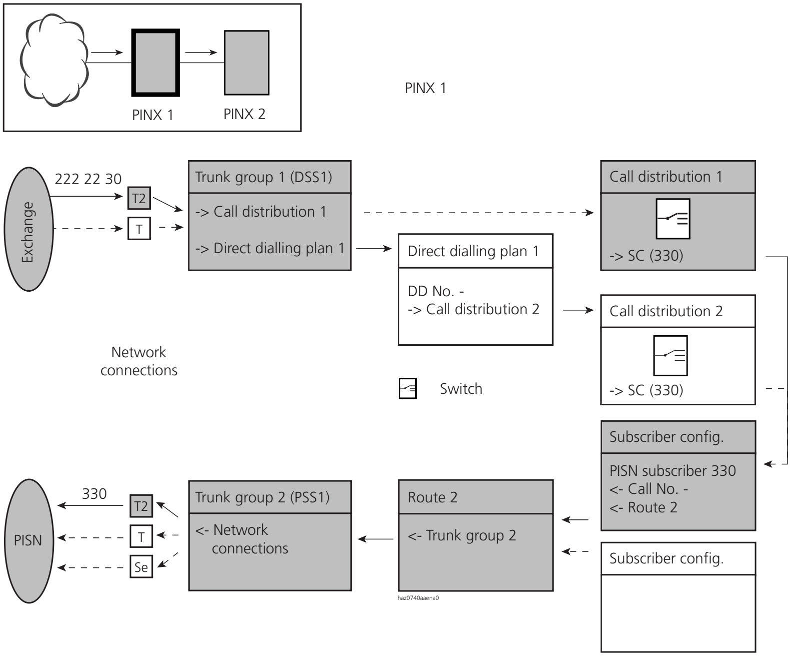


Fig. 2.94: Transit routing from the public network into the private leased-line network without direct dialling

Tab. 2.66: Routing parameter settings

Parameter	Parameter value
Trunk group 1:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Public
• Protocol	DSS1
• Direct dialling plan	1 (relevant only if a suitable DD number is found)
• Call Distribution Element	1 (reference number of a call distribution element)
Call distribution element 1:	
• Destinations	Switching position 1: 330 (PISN Subscriber)
• Incoming connections	Number of connections allowed simultaneously
Subscriber configuration PISN SC 330:	
• Route	2 (route reference number)
• Number	Not relevant in this case

Parameter	Parameter value
Route 2:	
• Trunk groups	2 (reference number of one or more trunk group(s))
• Digit barring	Use or do not use digit barring
• Outgoing connections	Number of connections allowed simultaneously
• Numbering plan identifier NPI	PNP
• Type of number TON	Unknown
Trunk group 2:	
• Network interfaces	Network interfaces of this trunk group
• Network type	Private
• Protocol	QSIG or QSIG / PSS1 ISO

5.6.2 From the private leased-line network into the public network

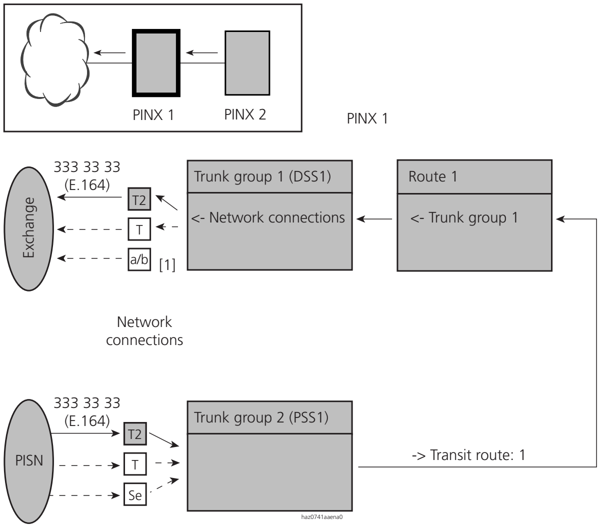
A PINX will route an incoming call from the private leased-line network on towards the public network if the incoming call has a phone number

- with NPI = E.164 or
- with an exchange access prefix.

Phone number with NPI = E.164

If the numbering plan identifier of an incoming call's phone number corresponds to type E.164, the call will be routed directly to the route set under "Transit route" by the incoming trunk group at a gateway or transit PINX.

The numbering plan identifier is set under "NPI" in the route configuration of the source PINX.



[1] One and the same trunk group cannot contain both analogue and digital network interfaces.

Fig. 2.95: Transit routing from private leased-line network → public network with NPI = E.164

Tab. 2.67: Settings for PINX 2 routing parameters

Parameter	Parameter value
Route 1:	
• Trunk groups	1 (reference number of one or more trunk group(s))
• Numbering plan identifier NPI	E.164
• Type of number TON	Unknown
• Send access code	–
Trunk group 1:	
• Network interfaces	Network interfaces of this trunk group
• Network type	Private
• Protocol	PSS1 (QSIG)

Tab. 2.68: Settings for PINX 1 routing parameters

Parameter	Parameter value
Basic setup PISN: • Transit route:	 1 (route reference number for transit calls to the public network)
Route 1: • Trunk groups • Digit barring • Outgoing connections • Numbering plan identifier NPI • Type of number TON • Send access code	 3 (reference number of one or more trunk group(s)) Use or do not use digit barring Number of connections allowed simultaneously E.164 Unknown –
Trunk group 1: • Network interfaces • Network type • Protocol	 Network interfaces of this trunk group Public DSS1

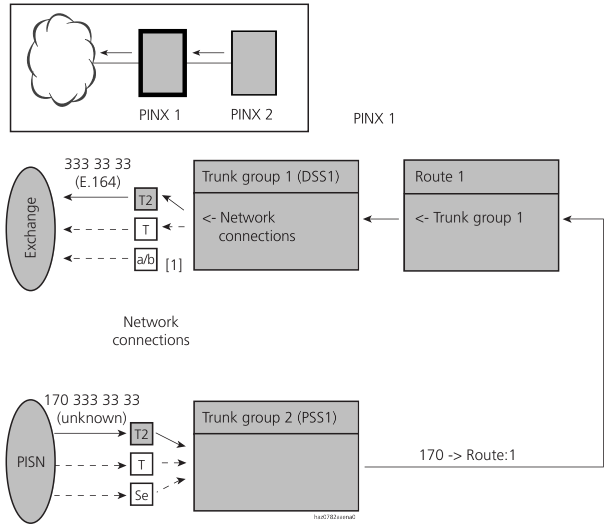
Phone number with an exchange access prefix

If the phone number has an exchange access prefix without route information (exchange access Business, Private, cost centre selection), the call will be routed on via the transit route.

If the phone number has a route selection prefix, the call will be routed via the corresponding route.

Note:
If a number has a route selection prefix and if NPI is E.164, the call will be routed via the transit route without truncating the prefix.

The exchange access prefix is set under "Send access code" in the route configuration of the source PINX.



[1] One and the same trunk group cannot contain both analogue and digital network interfaces.

Fig. 2.96: Transit routing for private leased-line network → public network with exchange access prefix

Tab. 2.69: Settings for PINX 2 routing parameters

Parameter	Parameter value
Route 1:	
• Trunk groups	1 (reference number of one or more trunk group(s))
• Numbering plan identifier NPI	Unknown
• Type of number TON	Unknown
• Send access code	170
Trunk group 1:	
• Network interfaces	Network interfaces of this trunk group
• Network type	Private
• Protocol	PSS1 (QSIG)

PINX 1 routing parameters as in Tab. 2.68.

5.6.3 From the private leased-line network into the private leased-line network

A call from the private leased-line network will be routed on at the transit PINX in accordance with the information of the PISN destination subscriber.

If the transit PINX is located in the same region as the destination subscriber, the phone number's regional prefix will be truncated.

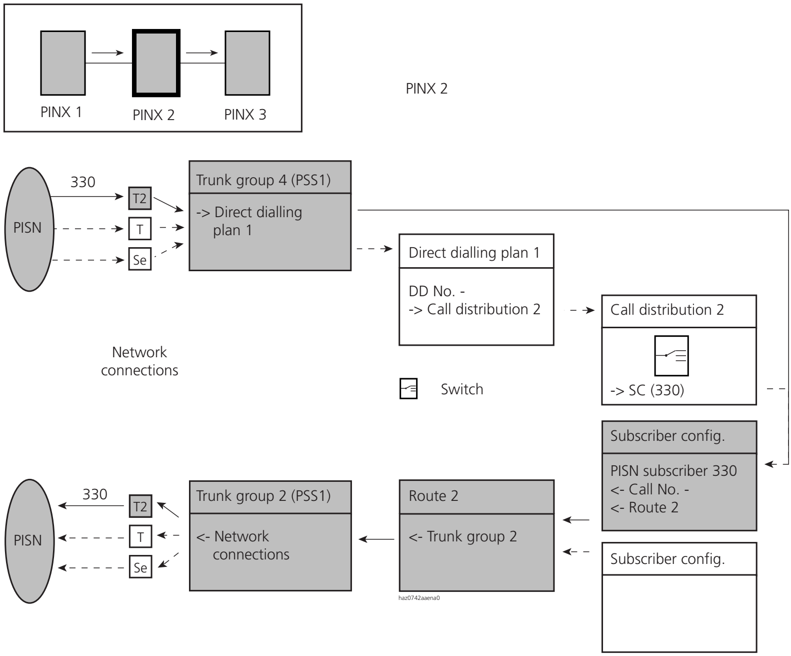


Fig. 2.97: Transit routing from the private leased-line network to another PISN subscriber

Tab. 2.70: Routing parameter settings

Parameter	Parameter value
Trunk group 4:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Private
• Protocol	QSIG or QSIG / PSS1 ISO
• Direct dialling plan	1 (relevant only if a suitable DD number is found)
• Call Distribution Element	Not relevant to this case
Subscriber configuration PISN SC 330:	
• Route	2 (route reference number)
• Number	Phone number to be dialled without exchange access prefix
Route 2:	
• Trunk group 2:	2 (reference number of one or more trunk group(s))
• Digit barring	Use or do not use digit barring
• Outgoing connections	Number of connections allowed simultaneously
• Numbering plan identifier NPI	PNP
• Type of number TON	Unknown
Trunk group 2:	
• Network interfaces	Network interfaces of this trunk group
• Network type	Private
• Protocol	QSIG or QSIG / PSS1 ISO

5.7 Testing overflow routing in the PISN

When a connection is setup the system checks the availability of the selected path. If it is not available due to overloading or due to a defect, an attempt will be made to set up the connection via an alternative route, depending on the configuration. There are two types of overflow routing:

- Overflow routing within the private leased-line network:
  Both the initial and the alternative connection path run via dedicated lines of the private leased-line network.
- Overflow routing via the public network:
  The initial connection path runs via dedicated lines of the private leased-line network while the alternative connection path runs via the public network.

Transmission of the CLIP number depends on the CLIP settings. See also the overflow situations illustrated in the example on page 199.

5.7.1 Overflow routing within the private leased-line network

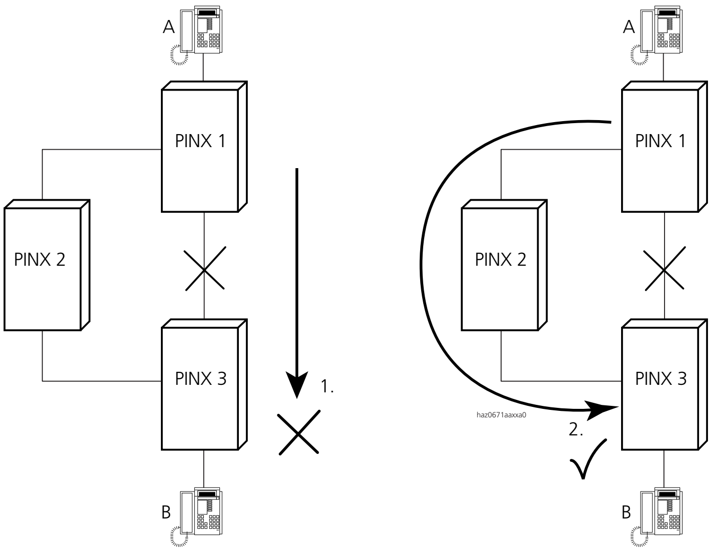


Fig. 2.98: Overflow routing in the private leased-line network via dedicated lines

Overflow routing in the private network can be resolved with the appropriate route configuration:

Configuration example

In PINX 1 let route 6 be provided for outgoing calls to PINX 3. If trunk groups 2 and 4 are allocated to this route, the first attempt will be to route the call via trunk group 2. If trunk group 2 is not available, the call will be routed via trunk group 4.

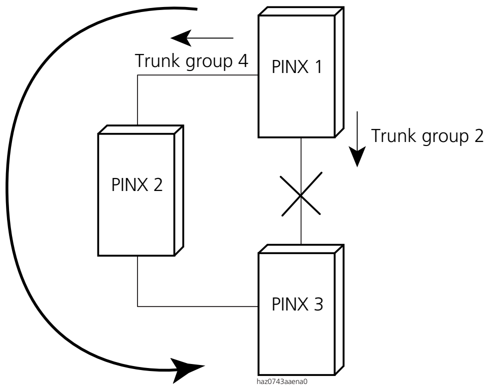


Fig. 2.99: Overflow routing in the private leased-line network using a sensible trunk group allocation in the route configuration

5.7.2 Overflow routing via the public network

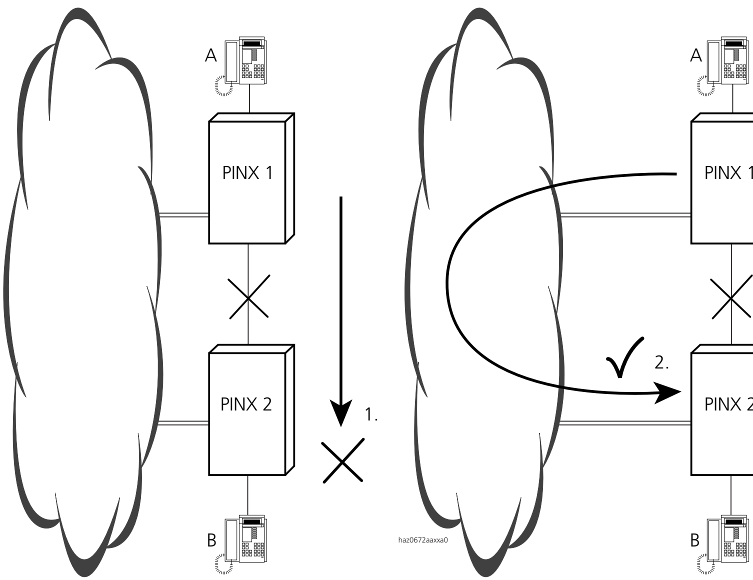


Fig. 2.100: Overflow via the public network -- the LCR function is used for this purpose

Overflow routing via the public network is resolved using Least Cost Routing.

Configuration example

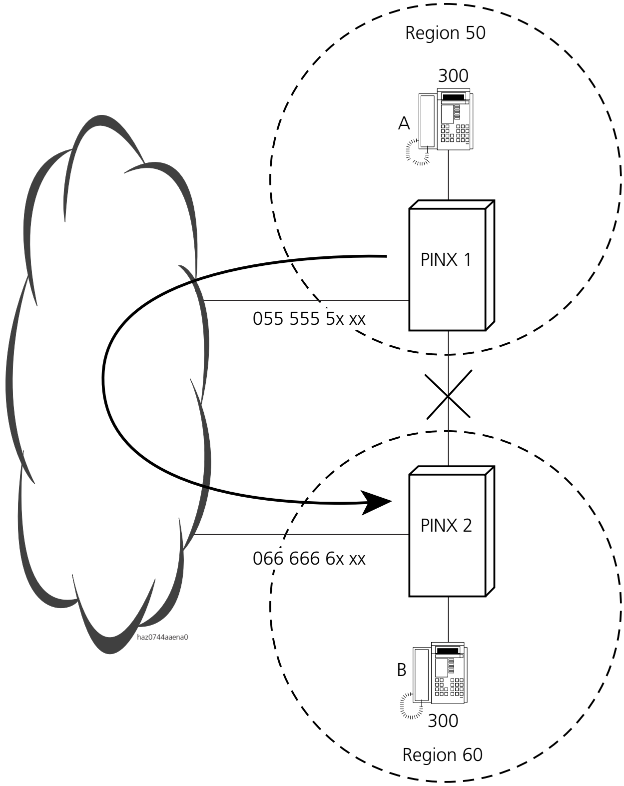


Fig. 2.101: Configuration example of overflow routing via the public network

In PINX 1's numbering plan, the PISN subscribers of PINX 2 are entered according to the principle 60xxx.

The numbers of the internal subscribers match with their direct dial numbers (subscriber B has internal number 300 and direct dial number 300).

Setting LCR to PINX 1:

- The digit sequence "60" is entered in the LCR table.
  – All outgoing, PISN-internal calls whose phone number begins with "60" will be analysed by LCR.
- In the routing table the entry for the first network provider stays blank. However, an alternative network provider is entered.
- Under normal conditions, calls whose phone number begin with "60" will be routed in accordance with the subscriber configuration. If the normal path is not available, the calls will be routed via the alternative network operator.
- The network operator table determines the route via which the alternatively routed calls are to be routed.
- In the network operator table the PISN phone number must be converted into an external direct dial number. The master number of PINX 2 is used for this purpose without its direct dial portion. The direct dial portion is formed by using the PISN subscriber number without regional prefix.
  This means that all the subscribers on PINX 2 need only one entry in the LCR configuration. This can only be achieved if the DDI numbers match the internal subscriber numbers.

Tab. 2.71: Settings for overflow routing on PINX 1

Parameter	Parameter value
LCR table: • I60 (regional prefix for PINX 2)	 O-flow PINX 2 (allocate to routing table "O-flow PINX 2")
Routing table "O-flow PINX 2": • Time zone x	 • Service provider: - • Alternative service provider: PINX 2 • Times: Allocate the times for "PINX 2"
Network operator table: • Network provider "PINX 2" • Conversion rule	 Route 6 0666666<3-.> (master number of PINX 2 without three-digit direct dial portion and the last three digits of the dialled phone number. If, for example, SC A dials 60300, the number 0666666300 is used, which corresponds to the direct dial number of SC B).
Route 6: • Name • Trunk groups • Digit barring • Outgoing connections • Numbering plan identifier NPI	 PINX 2, subscriber 2 No (do not consult digit barring) Number of connections allowed simultaneously E.164

Parameter	Parameter value
• Type of number TON	National
Trunk group 2: • Name • Network interfaces • Network type • Protocol	 ISDN exchange Network interfaces of this trunk group Public DSS1

5.8 Break-Out

An outgoing, external call is to be routed into the public ISDN only at the PINX that is closest to the call destination. If the source PINX and gateway PINX are a long way apart and connected with each other via dedicated lines, break-out can help to achieve considerable call charge savings.

For the caller always to be available under the same number regardless of the path via which his calls are routed to the public network, the called party must always be presented with a CLIP with that same number.

If the call is transmitted to the public network via a gateway PINX, the CLIP number will be outside the registered number range. If the network operator is to forward the CLIP number, the service "Special Arrangement" will have to be utilized subject to its availability from the network operator (see also page 181).

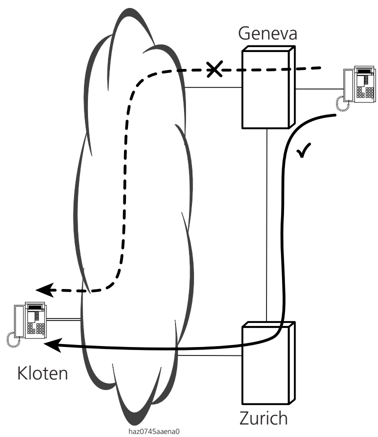


Fig. 2.102: Break-out

Configuration example

The PINXs of a company with branch offices in Zurich and Geneva respectively are connected with each other via a dedicated line. Outgoing calls made from Geneva to the local rate zone in Zurich are always to be routed into the public network at Zurich.

Incoming calls for the branch office in Geneva are always to be routed from the public network to PINX 1 in Geneva.

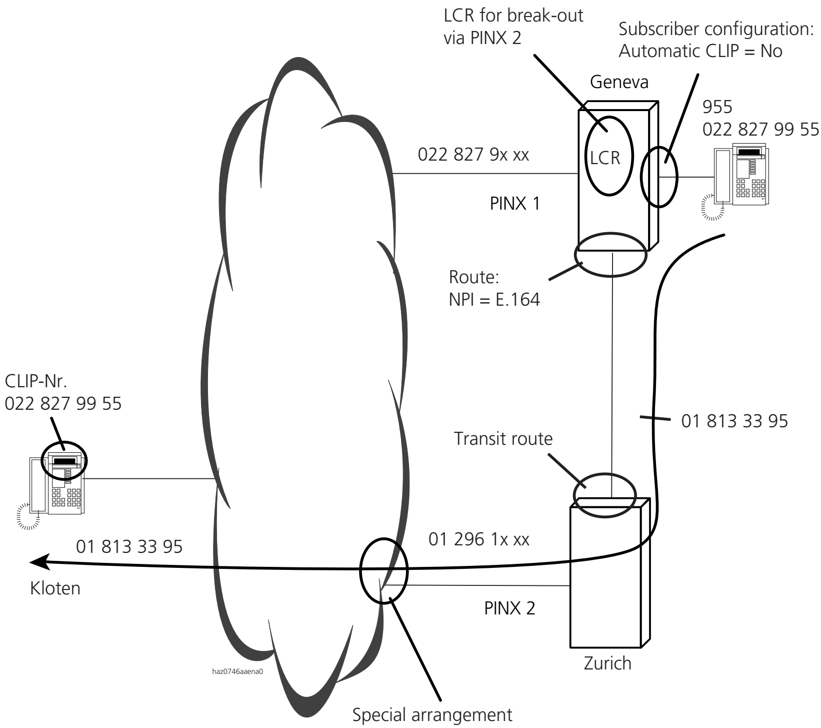


Fig. 2.103: Topology with important points

Planning the routes and trunk groups

To keep a network configuration as transparent as possible, it is a good idea always to use the same trunk group and the same route for the same function on all the PINXs. It makes sense, for example, to use trunk group 1 in each PINX for connections to the public ISDN network as trunk group 1 has this initialization value.

Settings on the source PINX (PINX 1):

- Subscriber configuration:
  A permanent CLIP number is configured for internal subscribers in Geneva, which is transmitted unchanged along with each outgoing call to the public network.
- Least Cost Routing:
  The initial digits of the numbers within the Zurich local rate zone are entered in the LCR table and allocated a route via the routing and network operator table
  (see also "Least Cost Routing (LCR)", page 304).
- Setting up routes:
  – All calls sent to the public network via Zurich are routed via a separate route. Its configuration must
    include the setting NPI = E.164 so that PINX 2 recognizes a call as external and routes it accordingly.
  – All calls addressed to PINX 2 subscribers in Zurich are routed via another route whose configuration contains the setting NPI = PNP.
  – Both routes can be allocated to the same trunk group.
- Trunk group settings:
  – Network = Private
  – Protocol = PSS1
  – Automatic CLIP = Yes

Tab. 2.72: Settings for break-out routing at the source PINX (PINX 1 in Geneva)

Parameter	Parameter value
Subscriber configuration:	
• Automatic CLIP	No (permanent CLIP number entry is used)
• NPI	E.164
• TON	National
• CLIP number	022 827 9x xx (x stands for the subscriber’s DD number)
LCR table:	
• ...	...
• 01 810	Zurich (allocate to the "Zurich" routing table)
• 01 811	Zurich (allocate to the "Zurich" routing table)
• 01 813	Zurich (allocate to the "Zurich" routing table)
• ...	...
"Zurich" routing table:	
• Time zone x	• Network provider: BreakOutZH • Times: Allocate the times for "BreakOutZH"
Network operator table:	
• Network provider "BreakOutZH"	Route 5
• Conversion rule	N (add dialled phone number)
Route 5:	
• Name	Zurich, ISDN exchange
• Trunk groups	2
• Digit barring	No (do not consult digit barring)
• Outgoing connections	Number of connections allowed simultaneously
• Numbering plan identifier NPI	E.164
• Type of number TON	Unknown
Trunk group 2:	
• Name	Zurich, PINX 2
• Network interfaces	Network interfaces of this trunk group
• Network type	Private
• Protocol	QSIG or QSIG / PSS1 ISO
• Automatic CLIP	yes

Settings at the gateway PINX (PINX 2)

Specifying the transit route
The transit route is specified using the "Transit route" setting. If an incoming call has a phone number with numbering plan identifier NPI = E.164, it will be forwarded via the defined route. This route leads to the public network (see also page 338).

Tab. 2.73: Settings for the break-out routing at the gateway PINX (PINX 2 in Zurich)

Parameter	Parameter value
Transit route:	
• Route	4 (this route is used for the transit routing)
Route 4:	
• Name	Zurich, exchange
• Trunk groups	1
• Numbering plan identifier NPI	E.164
• Type of number TON	Unknown
Trunk group 1:	
• Name	Zurich, ISDN exchange
• Network interfaces	Network interfaces of this trunk group
• Network type	Public
• Protocol	DSS1
• Automatic CLIP	yes

6 Data services

Outgoing data-service connections are set up and routed in a similar way to call connections. This also applies in a private leased-line network.

Incoming data-service connections are routed via data-service destination tables.

To route a call at a gateway or transit PINX on into the private leased-line network, a PISN subscriber is entered as the data service destination (see "Routing in the private leased-line network", page 357).

Internal data-service connections are also routed via the data-service destination tables (see "Routing to a destination in the data-service destination table", page 354).

Other subjects in this chapter include:

- "User-to-user signalling (UUS)", page 358
- "X.25 in the D channel", page 359

6.1 Data-service connections and destination tables

Data-service connections are routed via the call distribution element to a data-service destination table. In the data-service destination table each data-service type is allocated internal or PISN-internal destinations. There are several data-service destination tables; their number depends on the system type (see "System Limits", page 592).

The system analyses the data-service type involved and then routes the call to the configured destination.

Destinations include:

- Internal subscribers (including the remote maintenance access)
- User Groups
- PISN Subscriber
- Data-service individual destination

If the data-service type cannot be unequivocally allocated, it will be routed to the destination "Unknown data service".

If no destination is found, the call is cleared down.

Tab. 2.74: Data-service destination table

Data-service type	Interface of the destination terminal
Modem a/b	• Analogue user-network interface • Terminal Adapter on an S user-network interface
FAX 2/3	Analogue user-network interface
FAX 4 / CL1, 2, 3	• S user-network interfaces • Analogue user-network interface
TA V.110	Terminal Adapter on an S user-network interface
TA V.120	Terminal Adapter on an S user-network interface
B channel transparent	• S user-network interfaces • Remote maintenance access PPP
Telepac (X.25)	Terminal Adapter on an S user-network interface
Teletex	Terminal Adapter on an S user-network interface
Telex	Terminal Adapter on an S user-network interface
Videotex	Terminal Adapter on an S user-network interface
Unknown data service	Any destination

Routing to a destination in the data-service destination table

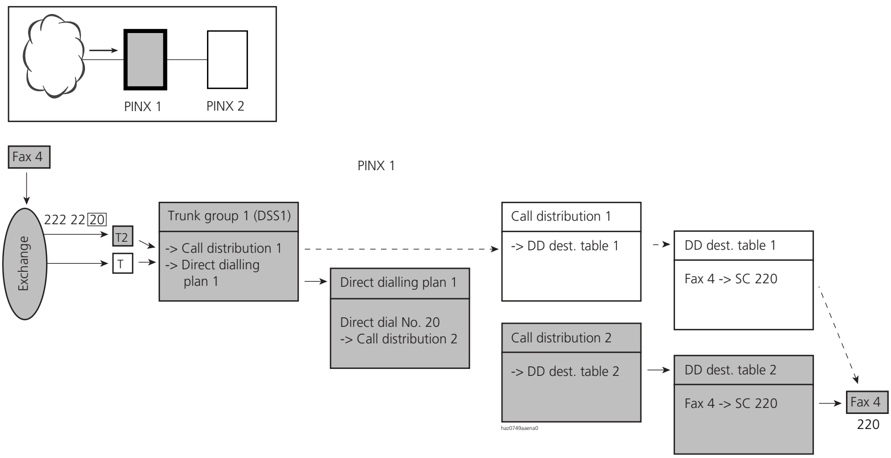


Fig. 2.104: Incoming data-service routing from the public network with direct dialling to a destination in the data-service destination table

Tab. 2.75: Routing parameter settings

Parameter	Parameter value
Trunk group 1:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Public
• Protocol	DSS1
• Direct dialling plan	1
• Call Distribution Element	1 (relevant only if there is no suitable DD number)
Direct dialling plan 1:	
• DD number 20	2 (reference number of a call distribution element)
Call distribution element 2:	
• Data-service destination table	2 (reference number of the data-service destination table)
Data-service destination table 2:	
• Data-service type "Fax 4"	220 (phone number of the data-service destination, Fax 4 in the example)

Routing to a data-service individual destination

If in the data-service destination table "DD individual destination" is entered as the destination for a data service type, the call is routed to the destination entered under "DD individual destination" in the call distribution element.

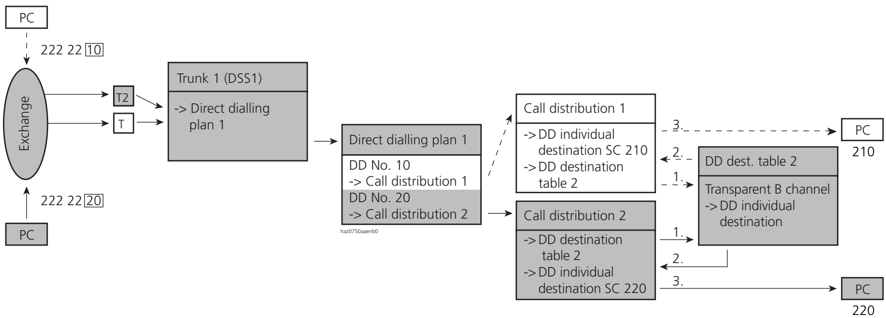


Fig. 2.105: Incoming data-service routing from the public network with direct dialling to a data-service individual destination

Tab. 2.76: Routing parameter settings

Parameter	Parameter value
Trunk group 1: • Network interfaces • Incoming connection • Network type • Protocol • Direct dialling plan	 Network interfaces in this trunk group Number of connections allowed simultaneously Public DSS1 1
Direct dialling plan 1: • DD number 10 • DD number 20	 1 (reference number of a call distribution element) 2 (reference number of a call distribution element)
Call distribution element 1: • Data-service destination table • Data-service individual destination	 2 (reference number of the data-service destination table) 210 (phone number of the data-service individual destination, in this instance PC 210)
Call distribution element 2: • Data-service destination table • Data-service individual destination	 2 (reference number of the data-service destination table) 220 (phone number of the data-service individual destination, in this instance PC 220)
Data-service destination table 2: • Data service type "B channel transparent"	 Data-service individual destination (of the call distribution elements)

The call is also routed to this destination if no data-service destination table is allocated in the call distribution element:

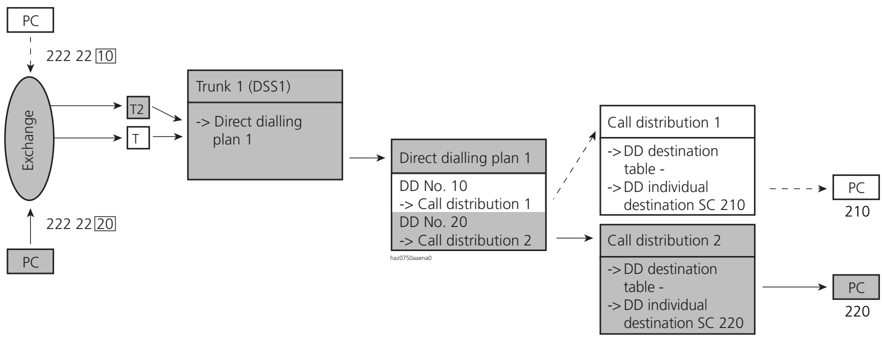


Fig. 2.106: Incoming data-service routing from the public network with direct dialling to a data-service individual destination but without entry in a data-service destination table

6.2 Routing in the private leased-line network

Data services are also available in the private leased-line network. To route a call at a gateway or transit PINX on into the private leased-line network, a PISN subscriber is entered as the data service destination.

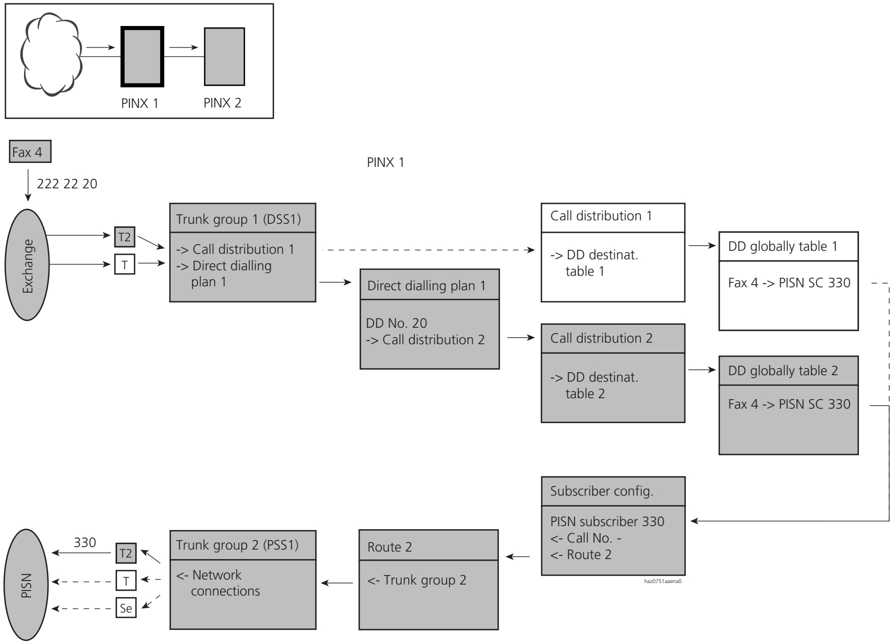


Fig. 2.107: Data-service routing transit from the public network with direct dialling to another PINX in the private leased-line network.

Tab. 2.77: Routing parameter settings

Parameter	Parameter value
Trunk group 1:	
• Network interfaces	Network interfaces in this trunk group
• Incoming connection	Number of connections allowed simultaneously
• Network type	Public
• Protocol	DSS1
• Direct dialling plan	1
• Call Distribution Element	1 (relevant only if there is no suitable DD number)

Parameter	Parameter value
Direct dialling plan 1: • DD number 20	 2 (reference number of a call distribution element)
Call distribution element 2: • Data-service destination table	 2 (reference number of the data-service destination table)
Data-service destination table 2: • Data-service type "Fax 4"	 PISN SC 330
Subscriber configuration PISN SC 330: • Route • Number	 2 (route reference number) Phone number to be dialled without exchange access prefix
Route 2: • Trunk group 2: • Digit barring • Outgoing connections • Numbering plan identifier NPI • Type of number TON	 2 (reference number of one or more trunk group(s)) Use or do not use digit barring Number of connections allowed simultaneously PNP Unknown
Trunk group 2: • Network interfaces • Network type • Protocol	 Network interfaces of this trunk group Private QSIG or QSIG / PSS1 ISO

6.3 User-to-user signalling (UUS)

The service "user-to-user signalling" allows subscribers to exchange a limited volume of data (128 bytes per subscriber) among themselves over the signalling channel (D channel) during the phase of connection set-up and clear-down. The exchange of data takes place even if a call is not answered.

Requirements:

- Both subscribers must have subscribed to the service with the network operator.
- The ISDN terminals or CTI applications used must support the service. System terminals do not support the service.

Scope

The PBX supports the service in variants 1 and 3 as per ETS 300 286, UUS1.

UUS is not supported in the private leased-line network and is only available at the PINX which is connected to the public network.

Application examples:

- Message to all callers, stating that the subscriber will only be available again later: SC A ← SC B
- Reference to a required callback: SC A → SC B
- Appointment transmission: SC A ↔ SC B
- Advance transmission of a code word or ID for logging into a system (SC B) from a CTI application: SC A → SC B

6.4 X.25 in the D channel

The public ISDN network provides transitions from the D channel of an ISDN access to X.25 data networks (X.31 Case B)[1)]. It provides the capability of transmitting packet-orientated data over the D channel at a max. of 9.600 bit/s (with low priority) in addition to protocol messages (example: automatic teller machines, credit card terminals).

The advantage of data transmission over the D channel is that voice / data calls can be made simultaneously over the allocated B channels.

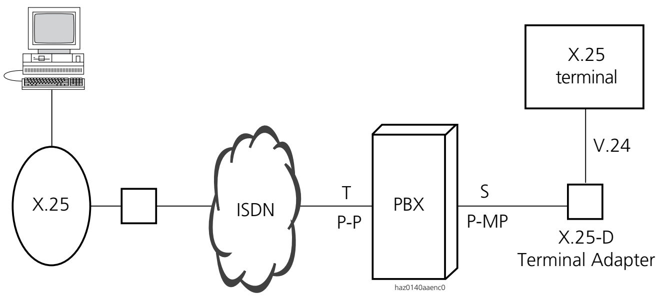


Fig. 2.108: X.25 transmission via the D channel

For information concerning the wiring and the availability of the service at the interfaces, refer to "Part 4 Installation".

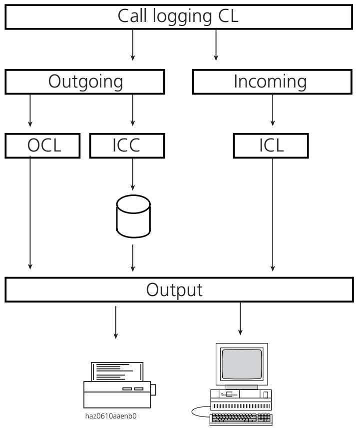
1) Only available in certain countries.

Tips on using X.25

- The digital switching of the public exchange must be capable of forwarding packet-switched data in the D channel to the X.25 network.
- Avoid excessive loads such as those that can occur through continuous data transmission or extensive backups.
- Errors can be prevented by reducing the data rate or by distributing the load over several X.25 channels.
- The S bus must be set to ETSI bus in the System Configuration.
- X.25 data can be transmitted simultaneously over the D channel and two B channels each of the S/T ports.

7 Call logging (CL)

Call logging consists of incoming call logging (ICL), outgoing call logging (OCL) and individual charge counting (ICC).



CL Call Logging
OCL Outgoing Call Logging (previously charge data acquisition CDA)
ICL Incoming Call Logging
ICC Individual Charge Counting

Fig. 2.109: Call logging at a glance

Individual charge counting or ICC

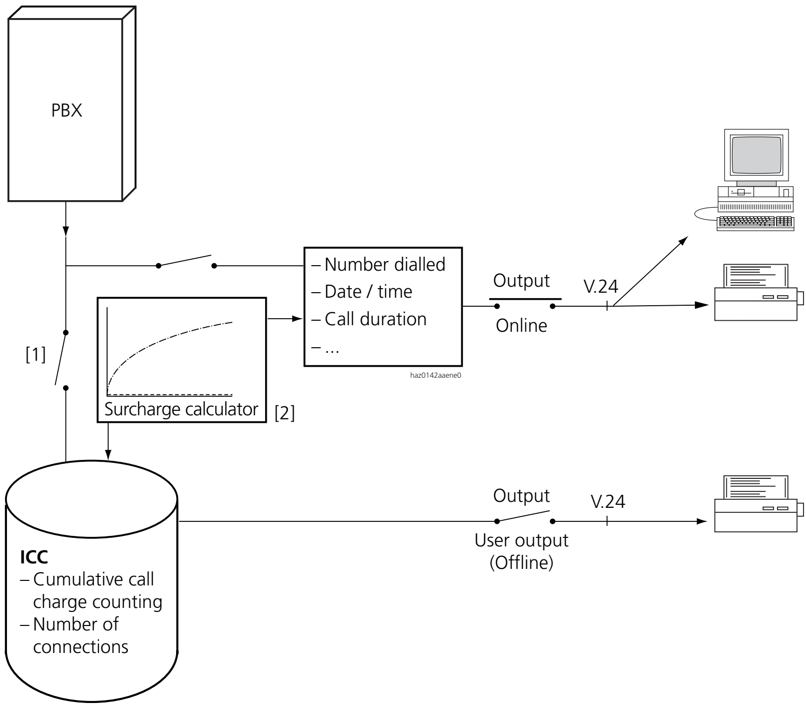
At the end of a call individual charge counting (ICC) assigns call charges to individually allocated cumulative counters. The data is stored in the PBX, can be viewed via the System Configuration and output in different forms via a V.24 interface to a Pocket Adapter, a V.24 interface of the PBX or via the Ethernet interface.

See also:
"Individual charge counting or ICC", page 364

OCL and ICL call logging

A multitude of call data from outgoing and incoming calls is logged and output directly via the corresponding interface. The data actually output in each individual case depends on the selected output format (see "Output formats", page 385).

The complete logging of OCL and ICL data for all call, transit, transfer and talk states allows a statistical evaluation of a system's capacity utilisation (OCL as of page 371, ICL as of page 379).



[1] Both OCL and ICC can be activated or deactivated throughout the system
[2] The surcharge calculator can be used for OCL and ICL

Fig. 2.110: Call logging and charge acquisition for outgoing traffic

Call logging in the PISN

In a PISN, call data is logged for each PINX. PISN-wide evaluation is carried out using PC-based applications for the acquisition and evaluation of call data.

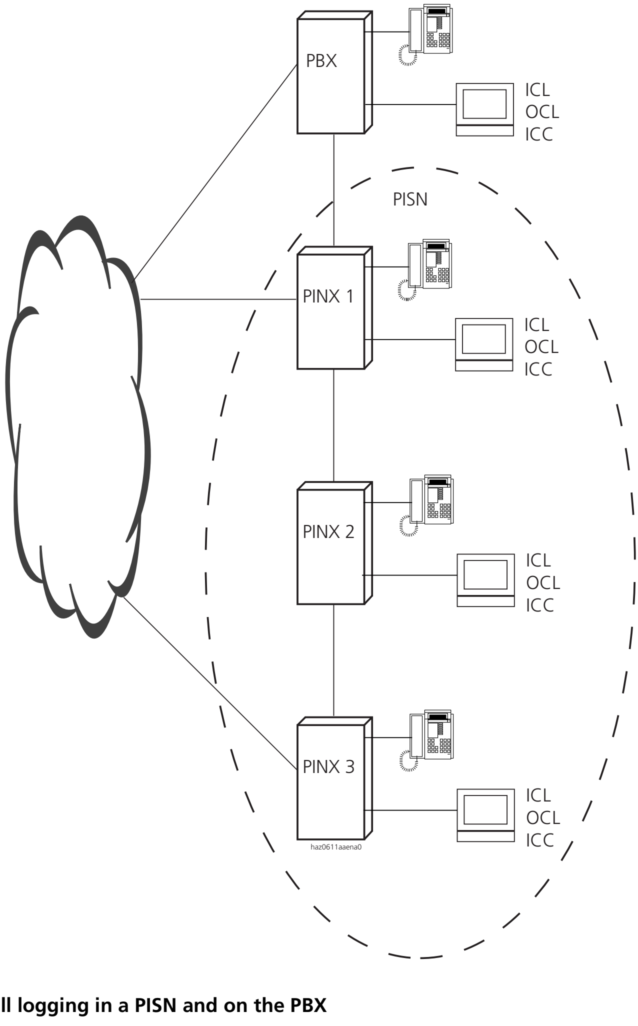


Fig. 2.111: Call logging in a PISN and on the PBX

7.1 Individual charge counting or ICC

Individual charge counting (ICC) automatically assigns call charges to cumulative counters at the end of a call; these call charges can be viewed in the System Configuration, output at the corresponding interface as individual or complete reports, or deleted.

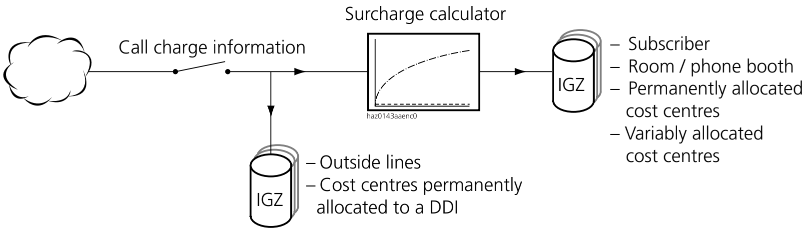


Fig. 2.112: Call charge allocation

7.1.1 Cumulative counter

In each case there is 1 counter:

- Per network interface
- Per user
- Per room
- per cost centre 00 to 100 (for more on cost centres, see "Cost centres", page 376)

There is also 1 drain counter per PBX.

In the case of subscriber counters, the ICC differentiates between 3 categories of call charges:

- Call connections, Business:
  Here call charges are added up for calls to the public network via the "Business Phone" exchange access prefix.
- Call connections, Private:
  Here call charges are added up for private calls or data connections to the public network via the "Private Phone and Data" exchange access prefix.

- Data connections, Business:
  Here call charges are added up for data connections to the public network via the "Business Data" exchange access prefix.

Counter readings

Each counter indicates the following values:

- Total amount of the call connections
- Charges for the last call connection
- Number of connections
- Logging period for the call data

Call charge allocation

- Network interface counters add up all the call charges incurred via their network interface.
- If call charges are permanently allocated to a cost centre, they are also counted on the subscriber counter.
- If call data is allocated variably to a cost centre using cost centre selection or the function *78, the data will not be counted at subscriber level.
- If subscriber B has rerouted to the network, subscriber B → subscriber C call charges will be charged to subscriber B.
- When using partial rerouting, the subscriber pays the call charges from the rerouting subscriber to the destination subscriber. The call charges are logged in the PBX.
- If a subscriber initiates a transfer call, the call charges incurred will be charged to the subscriber.

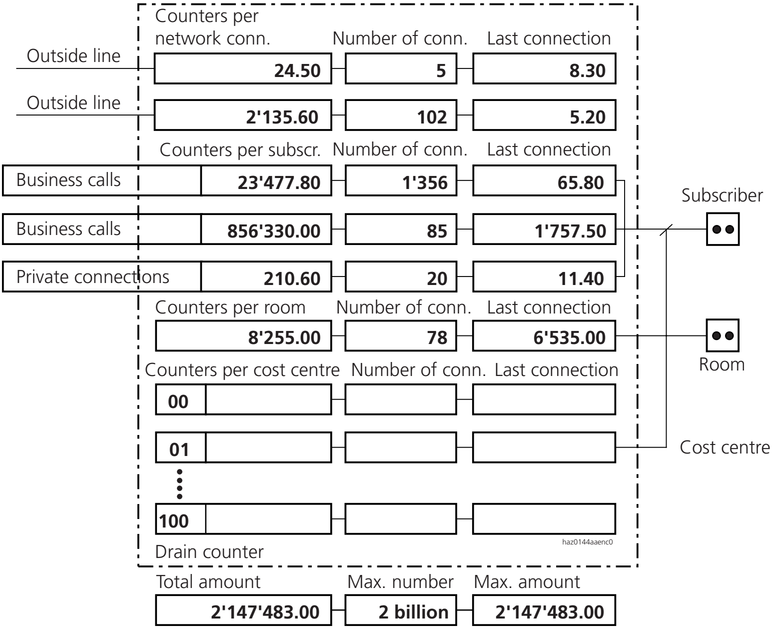


Fig. 2.113: Example of ICC cumulative counter

Currency

The amounts on the cumulative counters can be displayed in the local currency. The amount per metering pulse and the local currency depend on the parameter settings in the OCL/ICC menu.

drain counter

All call charges that cannot be unambiguously allocated will be added up by the system in a drain counter (cost centre 100). Example: Call charges for a call that was active when emergency operation was released ("Business"/"Private" allocation not possible).

Application Example

A company has the following departments: Sales, Buying, Development, Production and Logistics. To ensure that the call charges incurred can be allocated to the individual departments, a cost centre is created for each department. This cost centre is permanently assigned to each individual subscriber within each particular corresponding department. This enables the company to determine the call charges for both the department as a whole and the call charges of each individual subscriber.

7.1.2 Surcharge calculator

- The surcharge calculator is activated only if a surcharge curve has been configured and the subscriber has been allocated his business and private calls. No surcharge curves are configured after an initialisation.
- Network interface charge counters and cost centres that are charged via a call distribution element are never subject to the surcharge calculator.
- Call charges are indicated on each system terminal with a display while the call is in progress. If the subscriber has been allocated a surcharge calculator, the charges displayed include surcharges.

See also:
"Surcharge calculator", page 373.

7.1.3 ICC reports

ICC reports list all call charges over a user-definable period of time. The reports are output on the printer or PC set up for ICC.

There are two different kinds of ICC reports:

- Individual reports
- Complete reports

Individual reports

Individual reports indicate the call charges of a particular cumulative counter.

```
     ***** any text (max. 68 characters configurables) *****

CALL FEES                                                 0032
FROM 21.06.97 14:02 TO 30.06.97 16:00         OFFICE TELEPHONY
NUMBER 20            51 CALLS                     DM    123.80
```
haz0145aaena0

Fig. 2.114: Individual report for business telephony calls

```
+--------------------------------------------------------------------------+
:    ***** any text (max. 68 characters configurables) *****               :
:                                                                          :
:CALL FEES                                                             0032:
:FROM 21.06.97 14:02 TO 30.06.97 16:00       OFFICE DATA SERVICE:
:NUMBER 20             51 CALLS                          DM     123.80:
+--------------------------------------------------------------------------+
                                                              haz0146aaena0
```

Fig. 2.115: Individual report for business data service calls

```
+--------------------------------------------------------------------------+
:    ***** any text (max. 68 characters configurables) *****               :
:                                                                          :
:CALL FEES SERVICE INCLUDED                                            0033:
:FROM 21.06.97 14:02 TO 30.06.97 16:00    PRIVATE TELEPHONE+DATA:
:NUMBER 20             12 CALLS                          DM      15.20:
+--------------------------------------------------------------------------+
                                                              haz0147aaenb0
```

Fig. 2.116: Individual report for private calls (telephony and data service)

```
+--------------------------------------------------------------------------+
:    ***** any text (max. 68 characters configurables) *****               :
:                                                                          :
:CALL FEES                                                             0033:
:FROM 21.06.97 14:02 TO 30.06.97 16:00                   COST CENTRE:
:NUMBER 02             23 CALLS                          DM      23.50:
+--------------------------------------------------------------------------+
                                                              haz0148aaenb0
```

Fig. 2.117: Individual report for a cost centre

```
+--------------------------------------------------------------------------+
:    ***** any text (max. 68 characters configurables) *****               :
:                                                                          :
:CALL FEES                                                             0035:
:FROM 21.06.97 14:02    TO 30.06.97 16:00                                  :
:EXC. 2.2/1             78 CALLS                         DM     124.30:
+--------------------------------------------------------------------------+
                                                              haz0149aaenb0
```

Fig. 2.118: Individual report for a network interface

```
+--------------------------------------------------------------------------+
:    ***** any text (max. 68 characters configurables) *****               :
:                                                                          :
:CALL FEESSERVICE INCLUDED                                             0036:
:FROM 21.06.97 14:02 TO 30.06.97 16:00                              ROOM:
:NUMBER 34             4 CALLS                           DM      18.20:
+--------------------------------------------------------------------------+
                                                              haz0150aaenb0
```

Fig. 2.119: Individual report for all calls made by Room 34

Individual reports or individual receipts can also specify the following status information:

Tab. 2.78: Additional information between "Numbers" and "Connections"

Symbol	Meaning
*	If a cumulative counter has been printed out but not cleared (interim report), the cumulative counter is automatically marked with an "*".
B	If a subscriber happens to be making an external call when his cumulative counter is printed out, this fact is indicated by a "B" (BUSY). This information is not displayed in the case of cost centres and network interfaces.

Tab. 2.79: Additional information after the cumulative counter

Symbol	Meaning
+	The printed cumulative counter has overflowed during operation. The maximum value of 2,147,483 was exceeded and cumulative counting resumed at zero. (If the cumulative counter overflowed only once, the effective final amount can still be calculated by adding the value 2,147,483 to the amount displayed).
!	An individual call of more than 65,535 charge units was logged during operation.

Complete reports

All cumulative counters are printed out continuously, with a new page for each partial area. The entire header is printed out and a serial number added. If an A4 page is insufficient to hold all the related data of an area, a new page is started, with only the headers repeated to explain the columns. The total for the connections and amounts is printed out only on the last page.

If all the complete reports are printed out at the same time, the printout is made in the following order:

- Private subscribers
- Business subscribers
- Room extensions (Hotel system only)
- Cost centres
- Network interfaces

```
   ***** any text (max. 68 characters configurables) *****

CALL FEES FROM 30.07.97 18:00      SERVICE INCLUDED       1822
SUBSCRIBER     PRIVATE TELEPHONE + DATA

NUMBER   STATE   RECORD      SINCE      CALLS       FEE IN DM
20               01.07.97    18:05       104        521.10
21       B       03.07.97    21:50        27        278.30
.                .           .             .        .
43       *       02.07.97    16:25        23        204.20
```

haz0151aaena0

Fig. 2.120: Complete report for private calls made by all subscribers

```
NUMBER   STATE   RECORD      SINCE      CALLS      FEE IN DM
44                01.07.97   14:45        83        405.00
.                 .          .            .              .
691      B*       14.07.97   22:10         2          8.90
                             TOTAL       763       3216.30
```

haz0152aaena0

Fig. 2.121: New page (appears after a page break)

```
   ***** any text (max. 68 characters configurables) *****

CALL FEES FROM 27.06.97 18:00                             0040
SUBSCRIBER     OFFICE DATA SERVICE

NUMBER   STATE   RECORD      SINCE     CALLS         FEE IN DM
20                27.05.97   13:00         4         12.20
21                27.05.97   13:00         2          4.20
.                 .          .             .           .
29                27.05.97   13:00       123        213.80

                             TOTAL       412        529.40
```

haz0153aaenb0

Fig. 2.122: Complete report for business data connections

```
  ***** any text (max. 68 characters configurables) *****

CALL FEES FROM 30.07.97 18:00                            1822
EXCH. LINES

EXC      STATE   RECORD      SINCE      CALLS        FEE IN DM
 2.1     *       01.07.97    18:05          4          21.10
 2.2             03.07.97    21:50         27          78.30
 3.1             11.06.97    11:46         68         278.30
   .             .           .              .              .
10.2             14.07.97    22:10        824         848.90

                             TOTAL       2763        4213.30
```

haz0154aaenb0

Fig. 2.123: Complete report for all network interfaces

7.2 Call logging for outgoing calls (OCL)

OCL is used to log the outgoing connection data of individual calls and output the data via the system's corresponding interface at the end of the call. OCL can be activated and deactivated throughout the system and for each subscriber.

Output formats

The output formats PC1...PC5 are available for output on a PC.

For the output on a printer there is a choice of a list output (protocol) or an individual receipt output (for each call one multi-line receipt with additional text).

Only the output formats protocol and individual receipt are subject to the surcharge calculator allocated to the subscriber.

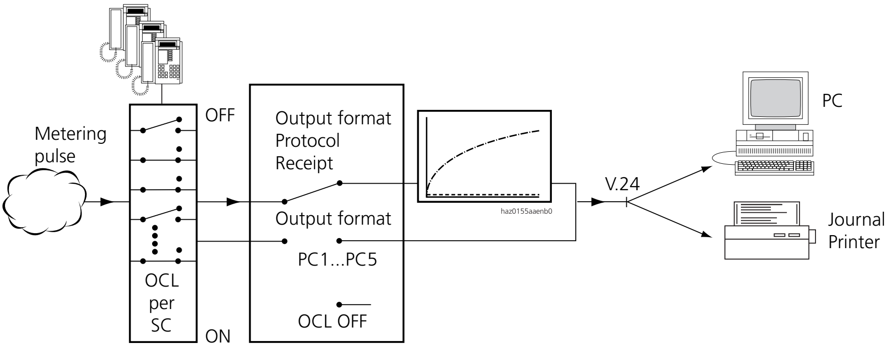


Fig. 2.124: Schematic sequence

See also:
"Output formats", page 385.)

7.2.1 General OCL settings

Activating OCL throughout the system

Allocating the required output format in the system configuration automatically activates OCL.

Tab. 2.80: Subscriber-related settings

OCL	Online output can be switched on and off for each subscriber.
Surcharge calcula-tor	One of four possible surcharge calculators can be allocated in each case for business and private calls.

Tab. 2.81: Printout as of a specific charge value

Output	As of
Business	...5.00...
Private	...0.10...
Cost centres	...0.10...
Room	...0.10...

The call charges are printed out only once the set values are exceeded.

The ICC, however, logs all the call charges and allocates them to the cumulative counters.

Digit barring if output is blocked

If for whatever reason the printer cannot print or the PC cannot receive data (see "Printer faults", page 384), the next 300 calls (System 2025 / 2045) or 1000 (System 2065) are stored internally in the PBX. After that, the selected digit barring (e.g. 1) becomes active. Then only the numbers enabled by the digit barring can be dialled.

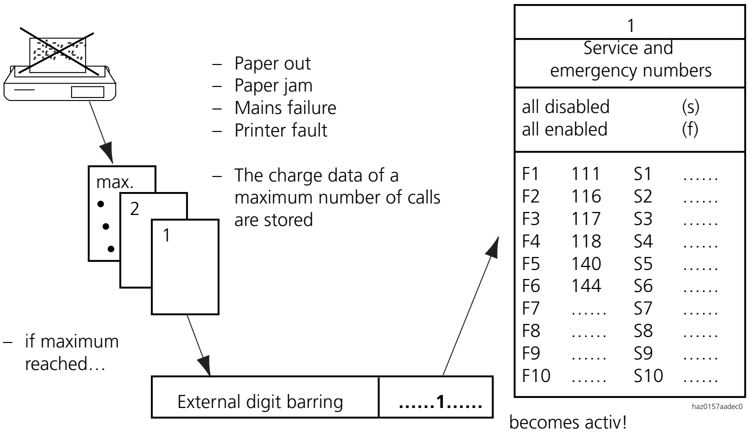


Fig. 2.125: Situation if output is blocked

7.2.2 Surcharge calculator

The surcharge calculator is used to assess surcharges on top of the official call charges.

Four independent surcharge calculators can be configured and allocated to the cumulative counters of the subscribers or rooms. Call charges are indicated to each subscriber (only on system terminals with a display) while the call is in progress. If the subscriber has been allocated a surcharge calculator, the call charges displayed include surcharges.

The cost curve of a surcharge calculator is defined by the basic surcharge and 4 cost ranges.

For each of the 4 ranges the user can specify a factor with which the call charges in the corresponding range limits are multiplied.

The basic surcharge is added to every chargeable call. If the basic surcharge is to be applied only as of, say, –.20, the following settings are necessary:

Range 1: surcharge factor 0; start of range 2: –.20.

This means, for example, that a hotel guest will only be charged for a call as of the second metering pulse.

Call charges on cost centres allocated to network interfaces or call distribution elements are never adapted via the surcharge calculator.

No surcharge calculators are configured after an initialization.

Application Example

Tab. 2.82: Example: A subscriber incurs 30.- in call charges. He pays 61.50.

Surcharge ranges	Network call charges			Surcharge		Call charge invoiced
	from	to	Amount	Factor	Charge per range	Display Charge counter
Basic surcharge	–	–	–	–	2.–	2.–
Range 1	0	10.–	10.–	X 3	= 30.–	32.–
Range 2	10.–	15.–	5.–	X 2	= 10.–	42.–
Range 3	15.–	20.–	5.–	X 1,5	= 7.50	49.50
Range 4	20.–	End value (here 30.–)	10.–	X 1.2	= 12.–	61.50

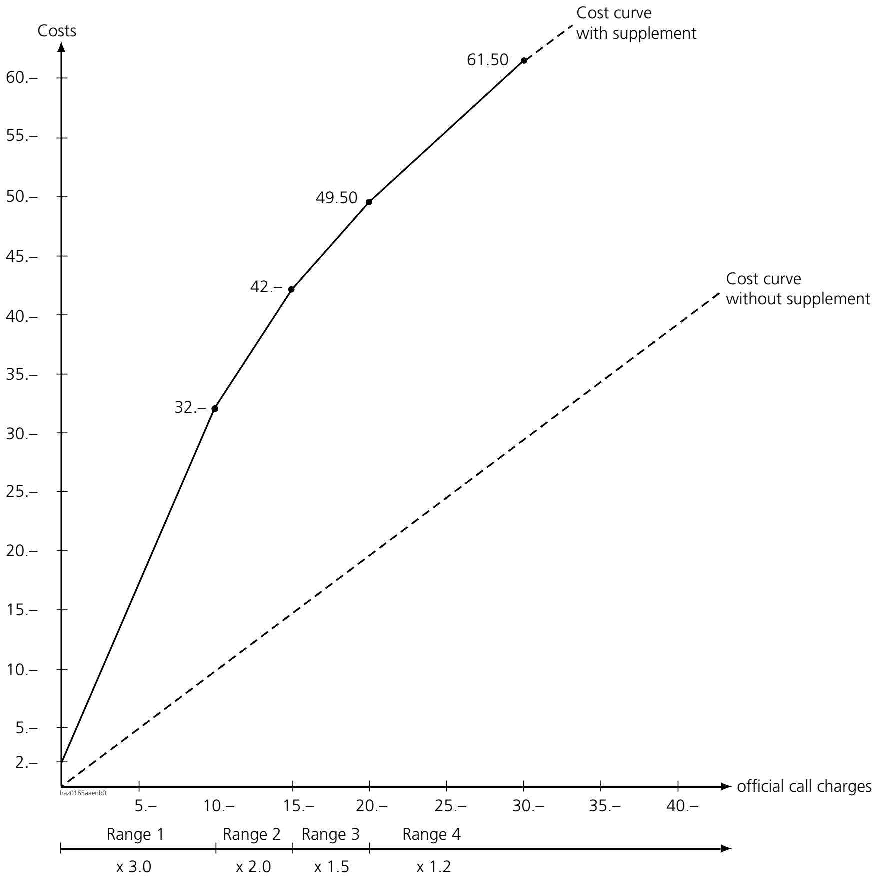


Fig. 2.126: Cost curve for the application example

See also:

System configuration:
- OCL/ICC/ICL
- OCL; SC configuration
- Metering pulse; SC configuration
- OCL; only on connection
- Report; OCL/ICC/ICL
- Surcharge calculator; OCL/ICC/ICL parameters

7.2.3 Data protection

The system offers the option to activate data protection, i. e. to blank out on the printout the last 4 digits of the number dialled (Fig. 2.127).

Data protection can be activated separately for business and private calls.

7.2.4 Cost centres

There are 100 cost centres (00 – 99) available. A cost centre can be allocated either permanently or for individual calls only (variable).

Permanent allocation

A cost centre can be permanently allocated to each subscriber and to each call distribution element. Any given cost centre can also be allocated to several subscribers or call distribution elements.

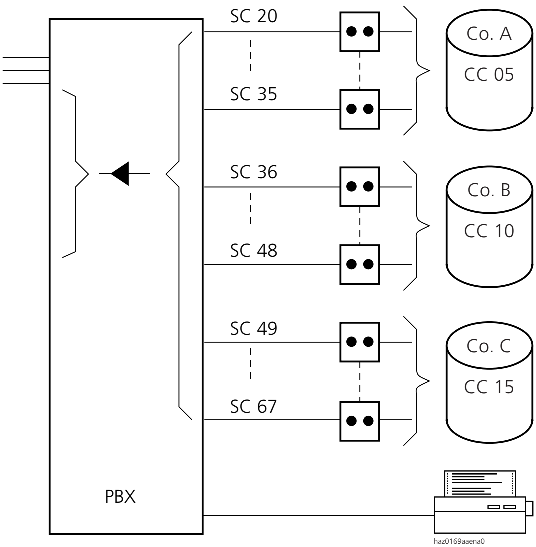


Fig. 2.127: Permanent cost centre allocation

Note:
Permanently allocated cost centres are not processed / logged in OCL (ICC only).

Variable allocation

Individual calls can be assigned to a cost centre either before the call by dialling the exchange access prefix code for cost centre selection or during the call using a */# procedure. With line keys, variable cost centre allocation is possible only using a */# procedure.

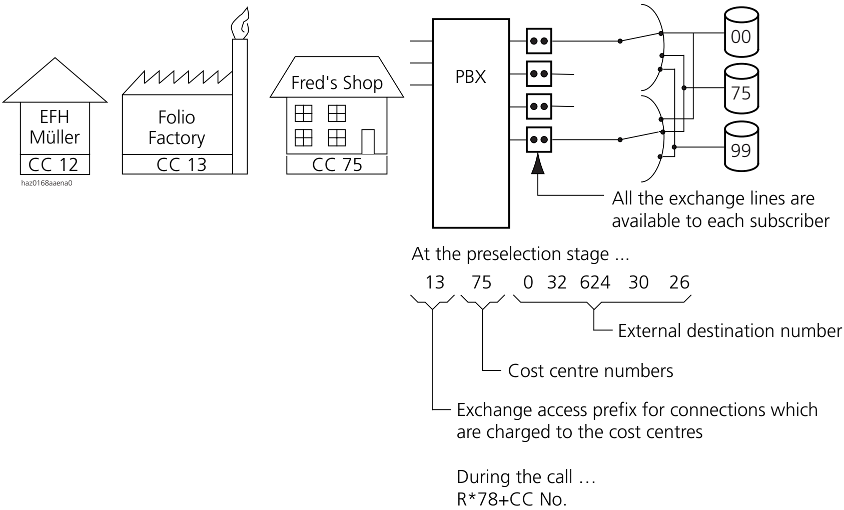


Fig. 2.128: Variable cost centre allocation

Surcharge calculator

If a subscriber has been allocated a surcharge calculator, the call charges are first adjusted with the surcharge calculator before being charged to the allocated cost centre.
The call charges logged on a call distribution element are always charged directly, without changes, to the allocated cost centre.

External cost centres

The call charges for individual calls can also be charged to external cost centres (variable allocation). External cost centres must have a two to nine-digit number. They are entered in a data field of an output format and can be analysed using a call data application.

7.2.5 Charge management

If an external call is forwarded internally, the charges incurred can be passed on to the next subscriber. This feature can be activated and deactivated throughout the system and applies only locally in the PINX. Subscriber A is making an outside call. After a while he hands the call over to subscriber B.

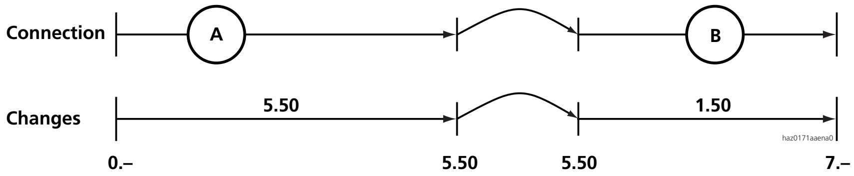


Fig. 2.129: Handing over the call charges from subscriber A to subscriber B

If charge management is switched on, the charges incurred by subscriber A during the call are passed on to subscriber B when the call is handed over. Subscriber A therefore does not incur any charges.

The total amount of 7.– is charged to subscriber B on the ICC and the OCL.

If charge management is switched off, an intermediate statement is drawn up for subscriber A when the call is handed over. It contains the charges incurred by subscriber A up to the point at which the call is passed on (5.50). This means that subscriber B incurs only those charges levied from the point at which the call is handed over to him (1.50).

On the Operator Console, call charges are always passed on to the next subscriber irrespective of whether or not charge management has been configured.

See also:
System configuration:
– Charge management; OCL/ICC/ICL parameters.

7.3 Call logging for incoming calls (ICL)

ICL deals with the logging of incoming call data. The ICL data can be used for example to analyse how quickly calls are handled, how many calls are lost because they are not answered quickly enough or not transferred successfully, or at what times a particularly large number of outside calls are received.
The data actually output in each individual case depends on the selected output format (see "Output formats", page 385).

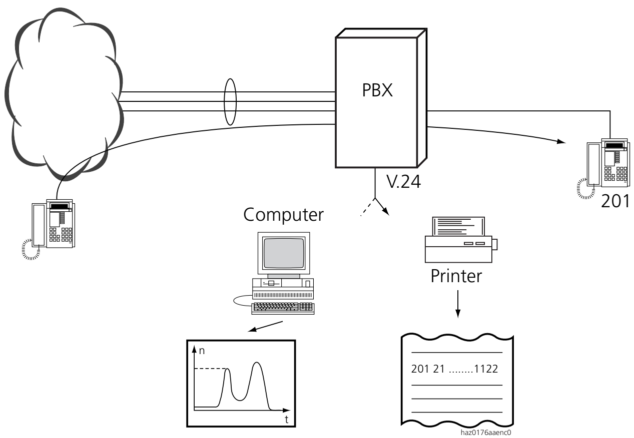


Fig. 2.130: Incoming call logging

ICL can be switched on or off for each call distribution element.

Sort characters are used to differentiate between data and voice calls and between answered, transferred and unanswered calls.

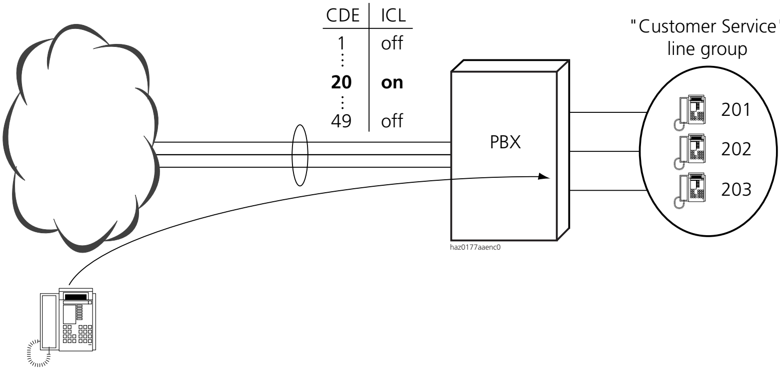


Fig. 2.131: ICL can be switched on or off in each call distribution element

Application Example

- Customer service: (032) 624 24 44
- "ICL ON", for customer service calls only (see Fig. 2.131).

Analysis is used to determine the quality of the call handling. One possible result of the analysis is that customer service is constantly busy between 10 a.m. and 11 a.m., and that an extra employee might be required during that period.

Cost centre allocation

It is possible to allocate a cost centre to an incoming call using the *78 + CC No. procedure. Businesses such as lawyers, physicians, consultants, etc., like to invoice their consultancy fees on the basis of the duration of the calls made with their clients. In such cases, ICL is combined with cost centre allocation.

Response if output is blocked

(See "Printer faults", page 384.)

ICL and OCL: Areas of conflict

ICL can lead to conflicts with OCL as the same resources are used in part. Critical points are:

- Same output channel:
  A certain amount of ambiguity can arise between OCL and ICL if clear sorting is not carried out. Under certain circumstances the equipment used for charge acquisition may have to be reconfigured.
- Separate protocols:
  ICL and OCL protocols can be configured independently of each other.
- Memory overflow:
- Ambivalence with transfer traffic:
  If external calls are transferred or rerouted to an external destination and then answered there, 2 protocol lines will be generated (if both OCL and ICL are enabled).
- Two-company system:
  ICL does not support separate logging according to company.

7.4 Call data output

ICL, OCL and ICC data is output to printers or other output devices via a V.24 interface of a Pocket Adapter, a V.24 interface of the PBX or the Ethernet interface. It is possible to configure which data is output on which of the available interfaces. Up to 4 output devices can be connected at the same time.

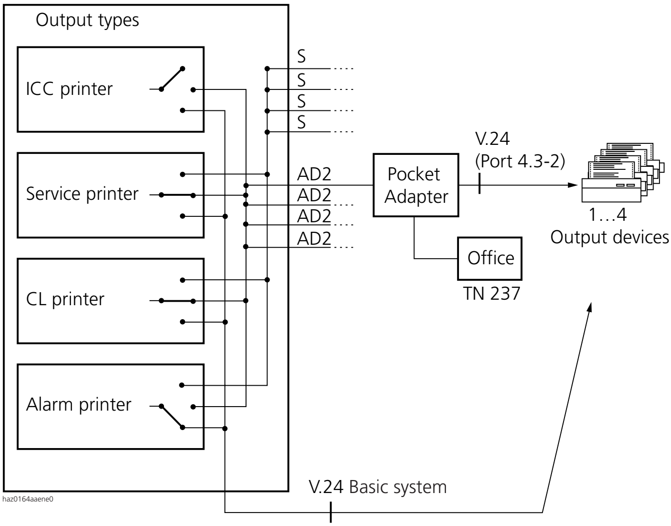


Fig. 2.132: Output concept

7.4.1 Interface configuration

One Ethernet interface and two V.24 interfaces are available on each basic system. One V.24 interface is also additionally available on each Office Pocket Adapter:

- The V.24 interfaces and the Ethernet interface on the basic system are set in the System Configuration.
- The V.24 interface on an Pocket Adapter is set at the back of the Pocket Adapter using a DIP switch.

7.4.2 Output types

The output type depends on who triggered the output. The output types are as follows:

Output type "ICC printer"

- Output at user’s request, e.g. using a command on the Operator Console
- ICC counter readings and reports (e.g. hotel check-out)

Output type "Service printer"

- Output at user's request, e.g. using a command on the Operator Console
- System configuration data
- Event message list
- Plus, for hotel:
- Wake-up lists
- Room status list

Output type "CL printer"

- Output triggered by the system (e.g. when call charges are incurred)
- OCL journal printouts (online)
- ICL journal printouts (online)

Output type "Event printer"

- Output triggered by the system
- System events such as:
  - Synchronization loss
  - External message destination unobtainable

Number of output devices

Up to 4 printers or output devices can be connected to the system. Various combinations are possible if more than one printer is connected to the system, for example:

- 4 output devices on Pocket Adapters
- 2 printers on Pocket Adapters, 2 printers on the basic system

If only 1 output device is connected, it will carry out all the output jobs. Under normal circumstances it handles the "CL printer" jobs for ICL and OCL output. If the output is triggered from somewhere else the output type is altered at short notice. If a "CL printer" job is followed by an "Event printer" job, the new job will be separated by a line of asterisks *. If the printout of the new job is to begin on a new page, a manual printer formfeed is to be performed beforehand.

Output when used in a hotel

When determining the output concept, remember to take note of the comments in "Menu and operating concept with AIMS", page 555.

Setting the page length

In principle, the page length can be set individually for all output types. If, however, only one output device is connected, the page length set for "Service printer" output type will apply.

7.5 Printer faults

If it is not possible to print on the CL printer for at least one minute (e.g. paper out), an event message will be triggered in the PBX. If the interruption can be remedied immediately, there are no further repercussions, as the call data is stored temporarily in a buffer. After a maximum number of calls (2025 / 2045 system: 300, (System 2065) 1000) the emergency digit barring is activated. The emergency digit barring affects all the subscribers throughout the system, with the exception of the Operator Console. This feature restricts the dialling options in the event of a printer jam. Once the fault is remedied, normal digit barring is activated once again.

Tab. 2.83: Buffering when output is blocked

Call	Call data
1	A corresponding event message is generated
.	ICL data is buffered
.	OCL data is buffered
.	
50%	
.	OCL data is buffered
.	ICL data is no longer buffered
.	
max.	
max. +1	Emergency digit barring is activated
.	
.	
.	

Note:
The PBX can only detect printer faults if the printer is operated with RTS/ CTS DSR/DTR flow control (hardware handshake mode).

7.6 Output formats

An output format defines which call data is to be output in which format. The following output formats are available:

Formats PC1 to PC5

Used for output on a PC. The PC5 format is the most comprehensive PC format and is recommended for all systems upgraded with a new PC application for the acquisition and evaluation of call data. The PC5 format contains ICL and OCL data (see page 385).

Formats PC1 to PC4 are still supported for PC applications that are already in operation. However, these formats are not suitable for PINX in a private network. There is a separate ICL and OCL variant for each of the formats PC1 to PC4 (see page 412).

Protocol format

This format is used for output on a printer. It does not contain all the data of the PC formats. There is a separate ICL and OCL variant for the Protocol format (see page 408).

Individual receipt format

This format is used to print out individual call charges as a receipt. The individual receipt format is available for OCL only (see page 411).

7.6.1 Structure of the PC5 output format

The PC5 format is used to output incoming and outgoing call data (ICL and OCL) on

- stand-alone PBXs and
- PINX in private networks.

It is the most comprehensive PC format and is generally recommended when upgrading with a new PC application for the acquisition and evaluation of call data.

The data is output in ASCII format in data fields. The data fields have a fixed field length. All the data fields together form a data record. The data record begins with a Tab and ends with a Carriage Return and Line Feed. These control characters are output with hexadecimal values as per Tab. 2.84.

Tab. 2.84: Control characters for separating data fields and data record

Designation	Meaning	Hexadecimal value	Usage
HT	Horizontal tab	09	Start of data record
CR	Carriage Return	0D	Together at the end of a data record (CR plus LF)
LF	Line Feed	0A	

A data field contains the following information:

- Data field name
- Data format
- Data field formatting
- The data field length

A data field can be identified by its position in the data record (Tab. 2.87).

Data field name

In PC5 format the data field name is not output.

Data format

A data field consists of a certain number of characters and a specific data format. Tab. 2.85 shows the symbols used for describing the data fields in Tab. 2.87.

Tab. 2.85: Symbols used to describe the data format

Symbol	Meaning	Number of characters
i	Integers	see "Length" in Tab. 2.87
d	Decimal figures	see "Length" in Tab. 2.87
yymmdd	yy = year, mm = month, dd = day	3 x 2 characters
hh:mm	hh = hours, mm = minutes	2 x 2 characters
hhHmmMss	hh = hours, mm = minutes, ss = seconds, H = "H", M = "M"	3 x 2 characters
cbbpp	c = primary channel group, bb = trunk card number, pp = network interface number	1+2+2 characters

Data field formatting

A data field can be formatted to be right or left justified and padded with leading numbers or blank spaces. Tab. 2.86 shows the symbols used to describe the data fields in Tab. 2.87.

Tab. 2.86: Symbols used for describing the data field formatting

Symbol	Meaning
I-	Left justified
-I	Right justified
0	Padded with "0" up to the permanently defined data field length
SP	Padded with spaces up to the permanently defined data field length

Data field length

The length of a data field can be permanently defined or remain variable up to a maximum length.

7.6.2 Data fields of the PC format

Tab. 2.87 shows the complete data record of a PC5 output. The data fields are listed in their task sequence.

Tab. 2.87: PC5 format

Data field	Name	Data format	Formatting	Length	Offset
Start of data record:					
Horizontal tab (HT)				1	0
Subscriber number	NO	i	SP -I	12	1
Cost centre number	CC	i	SP -I	9	14
Sort character	SC	i	0 -I	3	24
Date of start of connection	DATE	yymmdd	0 -I	6	28
Time of start of connection	TIME	hh:mm	0 -I	5	35
Duration of connection	DURATION	hhHmmMss	0 -I	8	41
Call charges	CHARGES	ddddddd.dd	SP -I	10	50
Number of metering pulses	METPUL	i	0 -I	5	61
Channel group / trunk card / network interface number	EXCH	cbbpp	0 -I	5	67
Caller identification 1	ID1	i	SP -I	20	73
Caller identification 2	ID2	i	SP -I	20	94
Destination number 1	DEST1	i	SP -I	40	115

Data field	Name	Data format	Formatting	Length	Offset
Destination number 2	DEST2	i	SP -l	40	156
Time-To-Answer	TTA	i	0 -l	3	197
Sequence number	SEQ.NO.	i	0 -l	3	201
Serial number	SERIAL NO.	i	0 -l	4	205
Carriage Return (CR)				1	209
Line Feed (LF)				1	210

Explanation of the data fields

Subscriber number

Outgoing:

- Entry for the caller's subscriber number.
- Entry for source PINX and stand-alone PBX; otherwise the data field remains empty.

Incoming:

- Contains an entry for source PINX and stand-alone PBX; otherwise the data field remains empty.
- Unanswered call:
  The number for the internal destination address is entered here. It can be a user group (UG), a key telephone (KT), a subscriber (SC) or a combination of these addresses.
  The SC number is entered here for SC and the combinations SC+UG or SC+KT. The UG number is entered here for UG and the combination UG+KT, where configured. If not, the configured ICL initialization number is entered, as with the KT setting.
- Answered call:
  Enters the number of the caller who took the external call or rerouted it externally.
- Transferred call:
  If the call was transferred internally or externally, the transferred subscriber is entered.

Cost centre number

- Entry for the variable cost centre (see "Cost centres", page 376).
- In the PISN the cost centre is logged only in the PINX in which the variable cost centre selection was carried out.

Sort character

The three-digit sort character xyz is used for identifying a data record. It is used to make the following distinctions:

Tab. 2.88: Meaning of the digits used in the sort character

Digit	Meaning
x	Destination/source network and connection direction
y	Type of network access/exchange-to-exchange connections
z	Call handling

Tab. 2.89: Value and meaning of the digit x

Value	Meaning
0	Outgoing to the public network
1	Outgoing to the PISN
3	Incoming from the public network
4	Incoming from the PISN

Tab. 2.90: Value and meaning of the digit y

Value	Meaning
0	Business network access, transferred
1	Business network access, subscriber-dialled
2	Incoming (appears only at the destination PINX)
3	Incoming to ACD destination (placed in ACD queue)
4	PISN transit
6	Network access with cost centre selection, transferred
7	Network access with cost centre selection, subscriber-dialled
8	Private network access, transferred
9	Private network access, subscriber-dialled

Tab. 2.91: Value and meaning of the digit z

Value	ICL	OCL
0	Incoming call, transferred	Normal call
1	Incoming call, answered directly	–
2	Unanswered call	–
3	Answered call. Appears only if 0 or 1 does not apply.	–
4	Incoming call connection, transferred to the network	Transfer call, set up through CFU / CFNR / CD into the network
5	–	Transfer call, transferred by internal subscriber
6	Incoming data service connection	Outgoing data service connections
7	–	Outgoing connections on phone booth extensions
8	–	Outgoing connections on room extensions
9	Rejected connection with destination • ERC (external remote control) • ACD (ACD queue)	

Tab. 2.92: Examples of sort characters

Sort character	Meaning
010	Outgoing connection to the public network, business network access, subscriber-dialled
160	Outgoing connection to the PISN, network access with cost centre selection, transferred
170	Outgoing connection to the PISN, network access with cost centre selection, subscriber-dialled
176	Outgoing data service connection to the PISN, network access with cost centre selection, subscriber-dialled
140	Outgoing connection to the PISN, transit
322	Incoming connection from the public network to the destination PINX, unanswered
324	Incoming connection from the public network to the destination PINX, transferred to the public network
443	Incoming connection from the PISN, transit, answered
420	Incoming connection from the PISN, transferred
421	Incoming connection from the PISN, answered directly

Tab. 2.93: Example of the output type "CL printer" in PC5 format

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
50250		321	180598	14:56	00H01m12			00101
		343	180598	14:57	00H02m05			00102
		140	180598	15:05	00H10m35			00103
50001		321	180598	15:20	00H01m12			00201

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0222222200	022222222		50	0023	014	1236
0333330000	033333333		54	0012	015	1237
0333330000	0333330000	50301	54			1238
0333330000	0333330000		50301	0012	007	1239

Date and time of start of connection

- Entry for the time of the start of connection at the logging PBX/PISN.
- In the case of forwarded calls the time logged is the time as of which the transferred call begins.

Duration of connection

- Entry for the duration of a connection by the logging PBX/PINX.
- The entry for unanswered calls is 0.

Call charges

- In the case of an ISDN connection, the call charge information supplied with the call is entered here.
- In the case of an analogue connection, the metering pulses are converted and entered.

Metering pulses

- In the case of an ISDN connection, the call charge information supplied with the call is converted and entered.
- In the case of an analogue connection the metering pulses are entered.

Network interface number

The primary channel group "0" is output in position "c", the trunk card number in position "bb" and the network interface number in position "pp".

Example:

00201 Trunk card in system slot 2. Network interface 1.

00504 Trunk card in system slot 5. Network interface 4.

Caller identification 1 and caller identification 2

These fields have a different meaning depending on the direction (incoming or outgoing calls).

- Caller identification 1, incoming:
  The number which the calling subscriber wants to present to the called subscriber is entered here. This number is displayed as CLIP on system terminals.
- Caller identification 2, incoming:
  A call number from the calling subscriber that has been verified by the network provider and found to be valid is entered here.

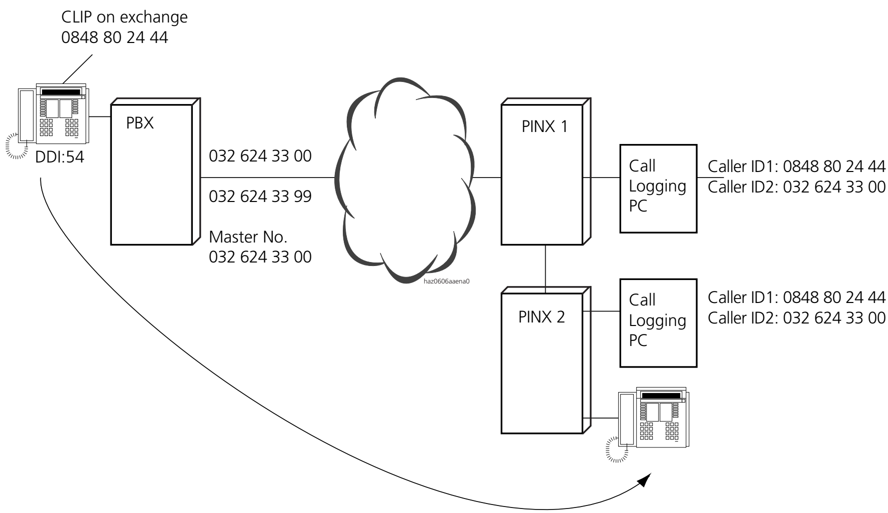


Fig. 2.133: Caller identification incoming

- Caller identification 1, outgoing:
  On the OCL report at the gateway/transit PINX: The subscriber call number valid within the network is entered here.
  On the OCL report at the source PINX no number is entered in this field.
- Caller identification 2, outgoing:
  On the OCL report at the source/transit PINX: The subscriber call number valid within the PISN is entered here.
  On the OCL report at the gateway PINX: The subscriber's DDI number is entered here.

On a stand-alone PBX the entries are output analogue to a source PINX.

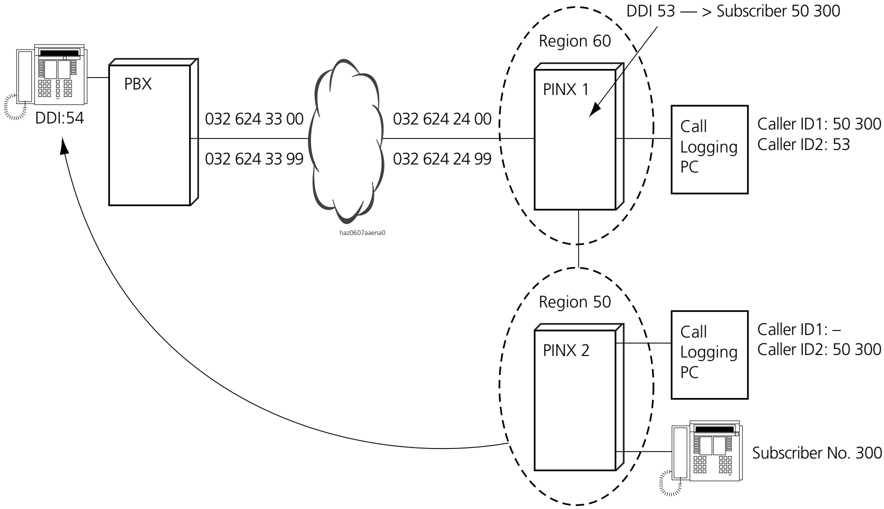


Fig. 2.134: Caller identification outgoing

Destination number 1 and destination number 2

These fields have a different meaning depending on the direction (incoming or outgoing calls).

- Destination number 1, incoming:
  - For incoming calls: no entry.
  - For calls to the DDI number for external remote control: Enter the instruction sequence selected in DTMF mode.

- Destination number 2, incoming:
  - For the gateway PINX and the stand-alone PBX: Enter the destination number received from the network provider (e.g. DDI number).
  - For the transit and destination PINX: Enter the PISN subscriber number of the called subscriber.

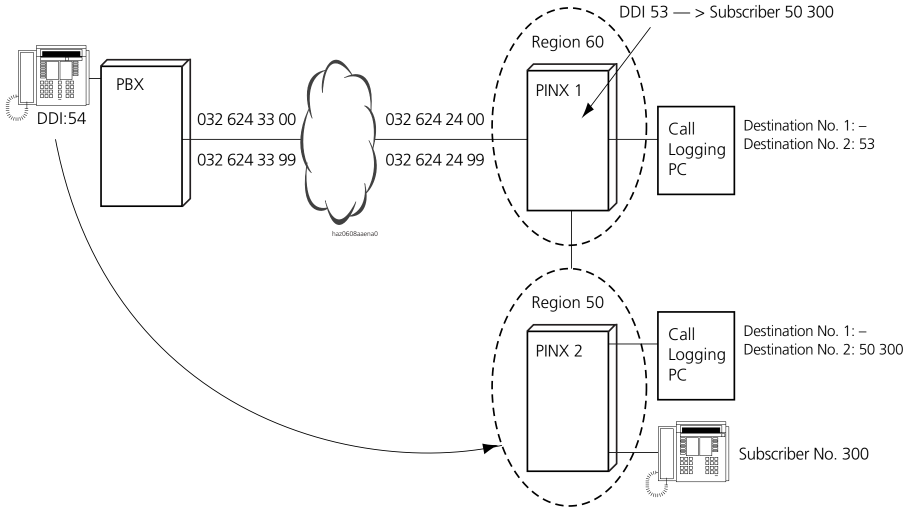


Fig. 2.135: Destination number incoming

- Destination number 1, outgoing:
  Enter the call number dialled by the PINX / PBX. Depending on the LCR configuration this call number may differ from the call number dialled by the subscriber.
- Destination number 2, outgoing:
  Enter the call number dialled by the subscriber.

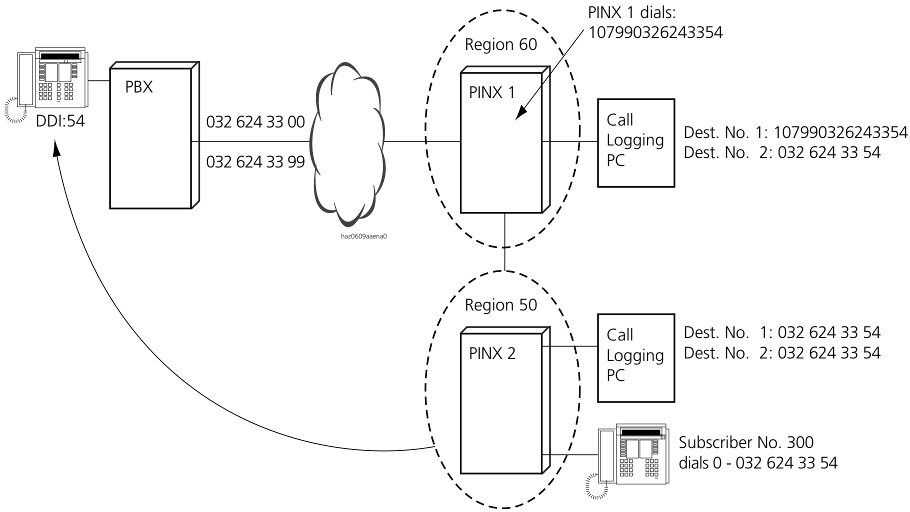


Fig. 2.136: Destination number outgoing

Time To Answer (TTA response time)

In the case of calls transferred internally the call time is logged at the transferred subscriber. The amount of time from the start of the ringing phase to the answering of a direct call is entered here (in seconds).
In the case of unanswered calls the ringing time is logged. Rejected calls are given TTA = 0.

Sequence number

Transferred calls have the same sequence number but separate serial numbers. Each incoming call is allocated a sequence number. However, since not all calls are necessarily logged (logging may be deactivated individually per network interface or call distribution element), the numbering is not necessarily continuous.

Serial number

The serial number is incremented by 1 each time an incoming or outgoing call is logged.

- After initialization the serial number is reset to the value 0.
- The serial number is not reset after a normal start.
- The serial number cannot be set manually.

7.6.3 Examples of the PC5 output on a stand-alone PBX

Outgoing calls to the public network

A business call is set up with the public network using subscriber dialling. The digit sequence 010 is therefore entered as the sort character. Least Cost Routing function is deactivated.

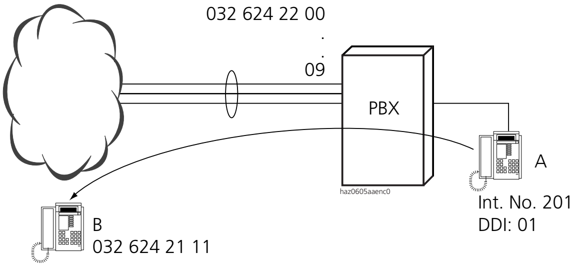


Fig. 2.137: Outgoing call to the public network

Tab. 2.94: OCL output for an outgoing call to the public network

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
201		010	060798	10:20	00H14M05	1.00	00010	00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
	01	6242111	6242111			0001

Incoming calls from the public network

Answered calls

All answered calls have a call duration greater than 0. The "Time" and "Date" fields indicate when the call was set up. The "TTA" field specifies the duration of the ringing phase. The sort character is 321.

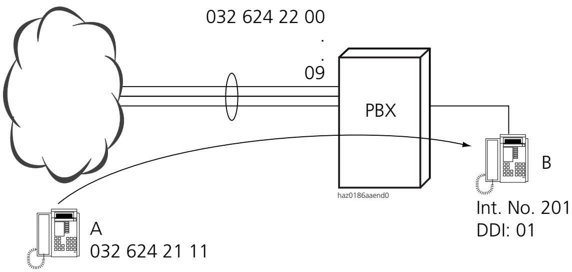


Fig. 2.138: Call to a free subscriber and phone conversation

- Subscriber A (032 624 21 11) calls Subscriber B (032 624 22 01).
- Subscriber B's terminal rings.
- Subscriber B answers the call.
- Subscriber A talks to Subscriber B.
- At the end of the conversation the call is ended by the two subscribers.

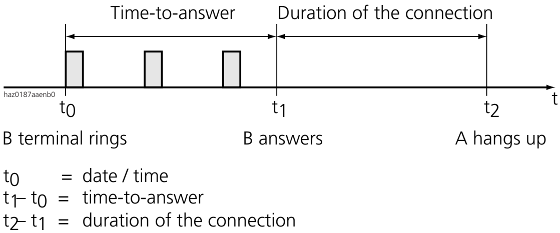


Fig. 2.139: Duration of ringing phase and established connection

Tab. 2.95: ICL output for an answered incoming call

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
201		321	020798	10:24	00H01M12			00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0326242111	0326242111		01	005	55	0114

Unanswered calls

0 is entered in the "Duration" field in the case of unanswered calls. The "Time" and "Date" fields indicate the time at which the call was received. The sort character is 322. The time entered in the "TTA" field indicates how much time elapsed before the caller hung up.

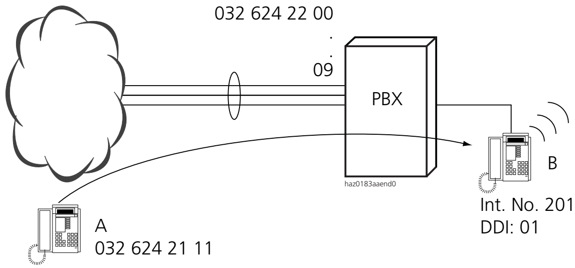


Fig. 2.140: Call to an absent subscriber

- Subscriber A (032 624 21 11) calls Subscriber B (032 624 22 01).
- Subscriber B does not answer.
- Subscriber A hangs up.

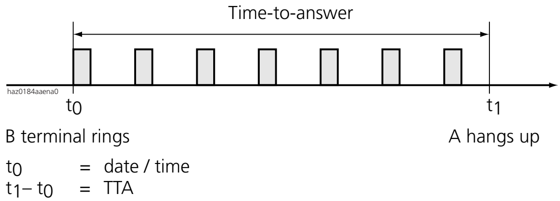


Fig. 2.141: Duration of the TTA ringing phase

Tab. 2.96: ICL output for an unanswered incoming call

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
201		322	020798	10:20	00H00M00			00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0326242111	0326242111		01	020	53	0112

Calls to a busy subscriber

If a busy subscriber is called and call waiting is disabled, 0 is entered in the "Duration" field. The "Time" and "Date" fields indicate when the call was received. The sort character is 322. Time To Answer is 0.

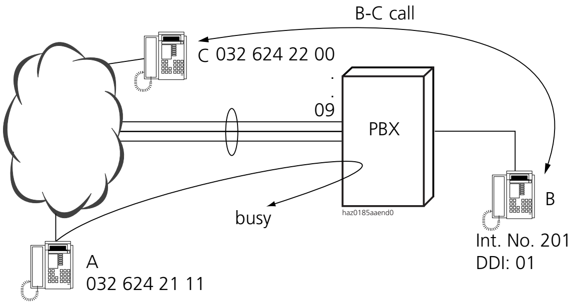


Fig. 2.142: Call to a busy subscriber

- Subscriber B is busy (call with call waiting not enabled).
- Subscriber A (032 624 21 11) calls Subscriber B (032 624 21 01).
- Subscriber A hears the busy signal.

Tab. 2.97: ICL output for a call to a busy subscriber

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
201		322	020798	10:22	00H00M00			00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0326242111	0326242111		01	000	54	0113

Transferred call

If a call was transferred to another subscriber, the subsequent ICL handling will depend on the charge management configuration.

Transferred call, charge management deactivated

The transferred phase of the connection is logged on a separate ICL. The call initially answered is given sort character 321. The sort character for the second ICL line is 320.

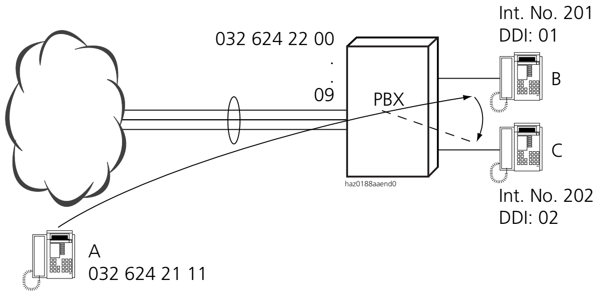


Fig. 2.143: Transferred call

Without prior notice:

- Subscriber A (032 624 21 11) calls Subscriber B (032 624 22 01).
- Subscriber B's terminal rings.
- Subscriber B answers the call.
- Subscriber A talks to Subscriber B.
- Subscriber B activates an enquiry call to Subscriber C
- Subscriber B hangs up.
- Subscriber C's terminal rings.
- Subscriber C answers the call.
- Subscriber A talks to Subscriber C.
- At the end of the conversation the call is ended by the two subscribers.

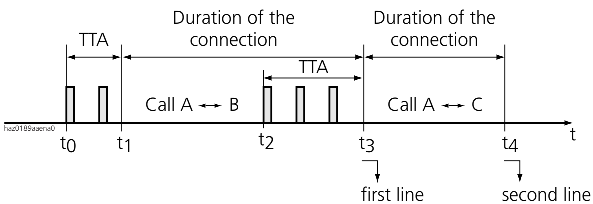


Fig. 2.144: Time phases for a transferred call without prior notice

Tab. 2.98: ICL output for a transferred call without prior notice

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
201		321	020798	10:26	00H01M00			00101
202		320	020798	10:27	00H12M03			00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0326242111	0326242111		01	004	56	0115
0326242111	0326242111		01	006	56	0116

With prior notice:

- Subscriber A (032 624 21 11) calls Subscriber B (032 624 22 01).
- Subscriber B’s terminal rings.
- Subscriber B answers the call.
- Subscriber A talks to Subscriber B.
- Subscriber B activates an enquiry call to Subscriber C
- Subscriber B does not hang up.
- Subscriber C’s terminal rings.
- Subscriber C answers the call.
- Subscriber B talks to Subscriber C.
- Subscriber B hangs up.
- Subscriber A talks to Subscriber C.
- At the end of the conversation the call is ended by the two subscribers.

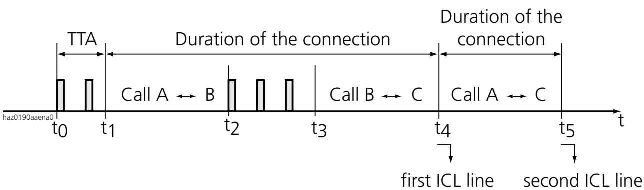


Fig. 2.145: Time phases for a transferred call with prior notice

Tab. 2.99: ICL output for a transferred call with prior notice

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
201		321	020798	10:26	00H01M00			00101
202		320	020798	10:27	00H12M03			00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0326242111	0326242111		01	004	57	0117
0326242111	0326242111		01	000	57	0118

Transferred call, charge management deactivated

The entire call is logged in a single line. The connection duration is entered in the "Duration" field. The "No." field contains the subscriber number of the last subscriber in the call. The sort character is 320.

Tab. 2.100: ICL output for a call to a busy subscriber

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
202		320	020798	10:26	00H13M03			00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0326242111	0326242111		01	007	58	0119

7.6.4 Examples of PC5 output in a PISN

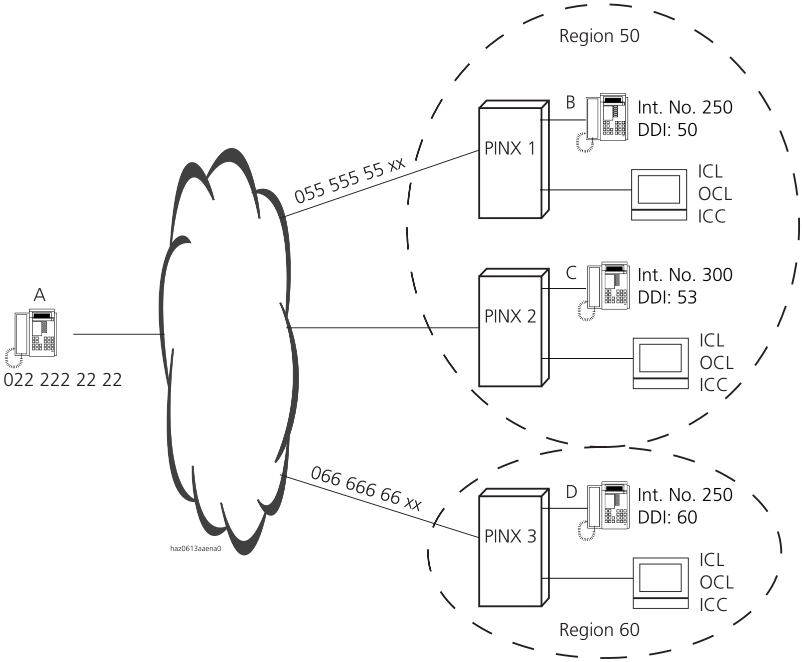


Fig. 2.146: PISN with 2 regions and joint numbering plan for Region 50

Tab. 2.101: Configuration of the PISN above

Numbering plan for	Separate prefix code	Internal (local) subscribers	PISN Subscriber
PINX 1	50	200...299	3xx, 60xxx
PINX 2	50	300...399	2xx, 60xxx
PINX 3	60	200...299	50xxx

The following examples are based on this PISN.

Direct outgoing connection

A connection is set up directly to the public network using subscriber-dialling (cost type: business).

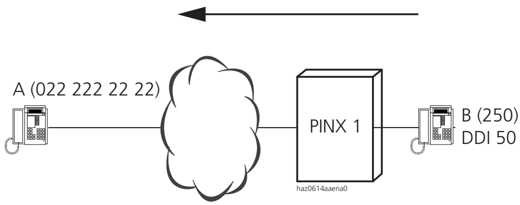


Fig. 2.147: Subscriber B dials Subscriber A (0 022 222 22 22)

Tab. 2.102: OCL output on PINX 1

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
50250		010	180598	14:50	00H02m10	0.20	00002	00102

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
	50	0222222222	0222222222			123

NO	PISN number of Subscriber B.
SC	Outgoing call to the public network. Subscriber-dialled network access, business.
ID1	Nothing is entered here as PINX 1 is both the source and gateway PINX.
ID2	Direct dial number via which Subscriber B can be reached directly from the public network.
DEST1, DEST2	The number dialled by the Subscriber (DEST2) was forwarded unchanged by the PINX (DEST1) as LCR is not activated.

Outgoing connection via a gateway PINX

A connection is set up to the public network via a gateway PINX using subscriber dialling (cost type: business).

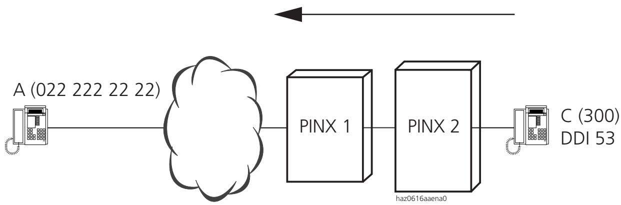


Fig. 2.148: Subscriber C dials Subscriber A (0 022 222 22 22)

Tab. 2.103: OCL output on PINX 2 (source PINX)

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
50300		010	180598	14:50	00H03m05	0.00	00000	00103

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
	50300	0222222222	0222222222			5677

NO	PISN number of Subscriber C.
SC	Outgoing call to the PISN. Subscriber-dialled network access, business.
CHARGES, METPUL	0 is entered here as the charges are incurred at PINX 1 and are not forwarded to PINX 2.
ID1	Nothing is entered here as PINX 2 is the source PINX.
ID2	PISN number of Subscriber C.
DEST1, DEST2	The number dialled by Subscriber C (DEST 2) is forwarded unchanged by PINX 1 (DEST 1) as LCR is not activated.

Tab. 2.104: OCL output on PINX 1 (gateway PINX)

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
		040	180598	14:51	00H03m05	1.50	00015	00104

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
50300	53	107070222222222	0222222222			1235

NO	Nothing is entered here as the caller is not a PINX 1 subscriber.
SC	Outgoing exchange-to-exchange call to the public network.
CHARGES, METPUL	The call charges are entered here.
ID1	PISN number of Subscriber C.
ID2	DDI number via which subscriber C can be reached from the public network.
DEST1, DEST2	The number dialled by the subscriber (DEST2) was converted into another call number (DEST1) by the LCR function. This is the number actually dialled by PINX 1.

Direct incoming call

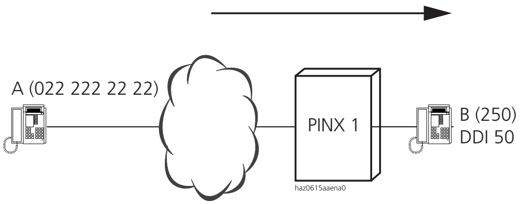


Fig. 2.149: Subscriber A calls Subscriber B (055 555 55 50)

Tab. 2.105: ICL output by PINX (destination PINX)

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
50250		321	180598	14:56	00H01m12	1.50	00015	00101

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
022222220	0222222222		50	0023	014	1236

NO	PISN number of Subscriber B.
SC	External call, answered directly.
ID1	Subscriber A wants to use this CLIP to present himself. It appears on Subscriber B’s system terminal display.
ID2	Caller’s CLIP number verified by the public network. Displayed to the destination subscriber only if no ID1 CLIP is available
DEST 1	Nothing is entered here with ICL output.
DEST 2	50 is Subscriber’s B direct dial number.

Incoming connection via a gateway PINX

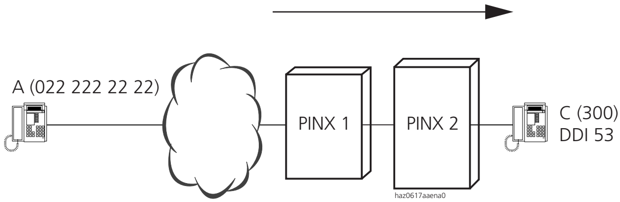


Fig. 2.150: Subscriber A calls Subscriber C (055 555 55 53)

Tab. 2.106: ICL output (line 1) and OCL output (line 2) at PINX 1 (gateway-PINX)

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
		343	180598	14:56	00H01m12			00103
		140	180598	14:56	00H01m12	0.00	00000	00119

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0222220000	0222222222		53	0012	015	1237
0222220000	0222220000	50300	53			1238

NO	Nothing is entered here with Gateway PINX.
SC	343: External incoming and answered transit call. 140: Outgoing transit connection to the PISN.
ID1	Subscriber A wants to use this CLIP to present himself. It appears on Subscriber C’s system terminal display.
ID2	Caller’s CLIP verified by the public network. Displayed to the destination subscriber only if no ID1 CLIP is available.
DEST1	Nothing is entered here with ICL output.
DEST2	53 is Subscriber C’s direct dial number.

Tab. 2.107: ICL output at PINX 2

NO	CC	SC	DATE	TIME	DURATION	CHARGES	METPUL	EXCH
50300		421	180598	14:56	00H01m12			00102

ID1	ID2	DEST1	DEST2	TTA	SEQ.NO.	SERIAL NO.
0222220000	0222222222		50300	0012	007	5678

NO	PISN number of Subscriber C.
SC	Incoming call from the PISN, answered directly.
ID1	Subscriber A wants to use this CLIP to present himself. It appears on Subscriber C's system terminal display.
ID2	Caller's CLIP verified by the public network. Displayed to the destination subscriber only if no ID1 CLIP is available.
DEST1	This field is always empty for ICL output.
DEST2	PISN number of Subscriber C.

7.6.5 Protocol format

This format is used for direct output on the printer. It is used if data acquisition is not carried out on the data carrier of a corresponding system.
The structure with page header and subsequent data lines is designed to make the protocol printout easier to read.

Page header

(does not contain any user data)

Tab. 2.108: Page header for protocol format

Content, text	Structure	Length	Print offset
Form Feed	FF, 0CH	1	0
Carriage Return	CR, 0DH	1	0
Line Feed	LF, 0AH	1	0
Space (2)	SP	2	0
NO (CC)	'NO' ('CC')	2	2
Space (4)	SP	4	4
SC	'SC	'2	8
Space (1)	SP	1	10
DATE	'DATE	'5	11
Space (2)	SP	2	16
TIME	'TIME	'4	18
Space (2)	SP	2	22
DURATION	'DURATION	'5	24
Space (4)	SP	4	29
EXCH	'EXCH	'3	33
Space (5)	SP	5	36
CHARGES	'CHARGES	'7	41

Content, text	Structure	Length	Print offset
Space (2)	SP	2	48
DIALLED	'DIALLED	'9	50
Space (1)	SP	1	59
NUMBER	'NUMBER	'6	60
Space (2)	SP	2	66
SERIAL NO.	'SERIAL NO.	'7	68
Line 1 end	CR	1	75
New line	LF	1	76
Space (2)	SP	2	0
'Underline	''—..—	'74	2
Line 2 end	CR	1	75
New line	LF	1	76

The page header

- Can be suppressed with the setting "..._OCL page length: 99".
- Output every time at the beginning of each page.
- Contains only formatting, no user data.

User data appears on the next line.

Example:
(see "Example of Protocol format", page 411)

Data lines

Tab. 2.109: Data lines for protocol format

Content, meaning	Structure	Format		Length	Print offset
Space	SP			2	0
Subscriber (cost centre) number[1)]	ttttt	|-	SP	5	2
Sort character	ooo	00	-|	3	8
Date of start of connection	ddmmyy	00	-|	6	12
Time of start of connection	hh:mm	00	-|	5	19
Duration of connection	hhHmmMss	00	-|	8	25
Trunk card number / network interface number / primary channel group[2)]	bb.pp/c	00	-|	5	34
Charges	gggggggg.gg	SP	-|	10	40
Call number dialled [3)]	zzzzzzzzzzzzzzzzzzzz	|-	SP	20	51

Content, meaning	Structure	Format		Length	Print offset
Serial number	IIII	00	-|	4	72
Carriage Return	CR			1	76
Line Feed	LF			1	77

1) Dialling determines whether Subscriber No. or CC No. is displayed. With exchange access 0 or 10, the SC No. is displayed; if exchange access with CC No. 13 is used or if charging is switched to the cost centre during the call using *78, the CC No. is displayed. Subscriber numbers are always output with format "|- SP"; cost centre numbers, always with format "00 -|".
As a cost centre number this field can be 5 or 9 digits long. Depending on the configured cost centre length ≤ 5, the field is 5 characters long. As of cost centre length ≥ 6, the length is 9 characters. If the cost centre length ≥ 6, all the offsets following the cost centre are incremented by 4 characters.

2) The trunk card number is output in position "bb"; the network interface number in "pp"; and the primary channel group "0" in "c" (see example on page 411).

3) With "Data protection ON" the last 4 digits of the number are replaced by "." (full stop) characters. In Switzerland and other countries this applies to private calls (data protection for business calls never active); in Germany, business calls (data protection for private calls never active).

Example of Protocol format

(combined with header line):

CL output	❏ OFF ❏ PC1 ❏ PC2 ❏ PC3 ❏ PC4 ❏ PC5 ☒ Protocol ❏ Receipt

NO	SC	DATE	TIME	DURATION	EXCH.	CHARGE	NUMBER DIALLED	SERIAL NO
21	098	040597	07:23	00H00M20	03.02	1.20	0121. . . .	0089
21	098	040597	07:24	00H00M34	03.02	3.40	0123. . . .	0090
22	010	040597	07:25	00H00M15	03.02	0.30	165	0091
17	010	040597	07:26	00H00M11	03.01	0.30	164	0092

haz0162aaenb0

Including surcharge

With data protection

Numbers with up to 4 digits are printed out in full

Fig. 2.151: CL output in protocol format

7.6.6 Individual receipt format

This format is used for output on the receipt printer for the purpose of confirmation and cash collection of the call made immediately beforehand.

As this structure is unlikely to be covered by an electronic system, no detailed description of the format will be given here.

Remarks

Example of individual receipt format:

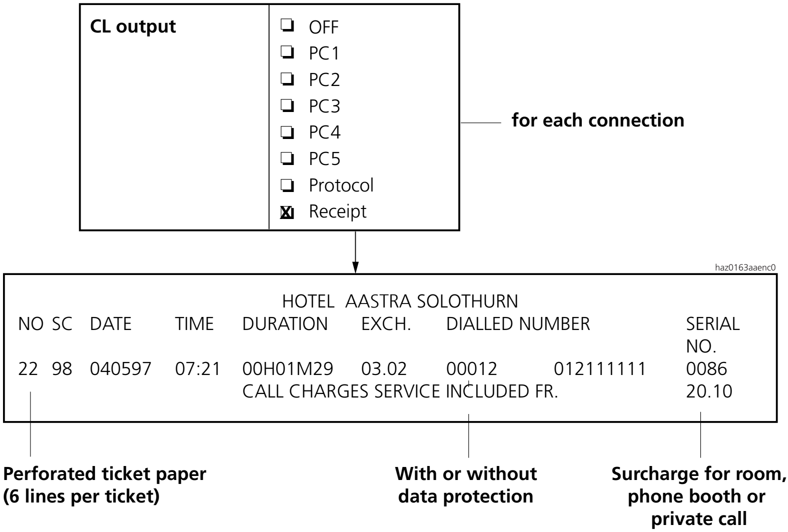


Fig. 2.152: CL output in Individual receipt format

If "Data protection" is configured, the field "call number dialled" will contain the " " (space) character in the last 4 places.

The printout of the individual receipt ends with the character "ETX" (End of Text, 03 hexadecimal). This character is required by certain types of receipt printers to actuate the cutting device.

7.6.7 Output formats PC1 to PC4

Output formats PC1 to PC4 are older formats which, although they are still supported, are no longer being expanded. Output format PC 5 is therefore recommended for new applications.

At the end of each call the call data logged is printed out on one of the system's V.24 interfaces in the "CL printer" output type.

Data record field structure

The fields are separated by one or more "Space" ASCII characters. The data import mask must therefore take account of the position of the beginning of the field ("Offset" column in the structural descriptions below).

The symbols and conventions listed in Tab. 2.110 are used to format the fields:

Tab. 2.110: Format conventions

Symbol	Meaning
– I	Right justified
I –	Left justified
00	00 Padded with "0" up to the defined data field length
SP	Padded with spaces

Certain fields take on different formats depending on the system configuration. These exceptions are appended as notes directly after the structural descriptions.

"Format" field in the structural descriptions below:
Certain fields take on different formats depending on the system configuration.
I– SP: means left justified and padded with spaces.

Sort character

Special characters used in the data string
In principle, all outputs are in the form of text based on the ASCII standard. Special, non-printing ASCII characters are used for structuring the data records:

Tab. 2.111: Special characters

Abbreviation	Meaning	Hexadecimal value	Usage
HT	Horizontal tab	09	Start of a data record
SP	Space	20	Field separator
CR	Carriage Return	0D	End of a data record
LF	Line Feed	0A	End of a data record

Sort characters for the "CL printer" output type
Sort characters (SC) denote the type of connection and are shown in the "CL printer" output type.

Tab. 2.112: Printout with sort characters

NO.	SC	DATE	TIME	DURATION	EXCH.	CHARGE	NUMBER DIALLED	SERIAL NO.
691	10	311290	05:20	01H03M45	10.02	67.70	005688223211	0678
21	90	311290	07:18	00H01M20	03.01	0.80	065248755	0679
23	16	311290	07:22	00H19M50	04.03	11.90	065243024	0680

haz0158aaena0

Sorting character

Tab. 2.113: The first digit of the sort character

Value	Meaning
0	Outgoing business exchange traffic, transferred
1	Outgoing business exchange traffic, subscriber-dialled
2	Incoming traffic
6	Outgoing cost centre exchange traffic, transferred
7	Outgoing cost centre exchange traffic, subscriber-dialled
8	Outgoing private traffic, transferred
9	Outgoing private traffic, subscriber-dialled

Tab. 2.114: The second digit means

Value	Meaning
0	Direct connection. Appears whenever "7" or "8" does not apply unambiguously.
1	Answered directly (incoming traffic)
2	Unanswered (incoming traffic)
4	Exchange-to-exchange connection, established by CFU / CFNR / CD to the network
5	Exchange-to-exchange connection, transferred by internal subscriber
6	Outgoing data service connections
7	Outgoing connections on phone booth extensions
8	Outgoing connections on room extensions

Tab. 2.115: Examples

Value	Meaning
00	Outgoing business exchange traffic, transferred
10	Outgoing business exchange traffic, subscriber-dialled (normal case for business traffic)
14	Outgoing business exchange traffic, subscriber-dialled, established by CFU / CFNR / CD to the exchange
16	Outgoing data service connection, subscriber-dialled
80	Outgoing private exchange traffic, transferred

Value	Meaning
00	Outgoing business exchange traffic, transferred
87	Outgoing private exchange traffic, transferred (phone booth extensions)
88	Outgoing private exchange traffic, transferred (room extensions)
90	Outgoing private exchange traffic, subscriber-dialled (normal case for private traffic)
97	Outgoing private exchange traffic, subscriber-dialled (phone booth extensions)
98	Outgoing private exchange traffic, subscriber-dialled (room extensions)

Maximal number length

If the internal numbers are longer than is possible in the output format, they will be truncated from the left.

If the external numbers are longer than is possible in the output format, they will be truncated from the right.

PC1 format

This format covers requirements for direct transfer to a PC (PC1).

Format structure

Tab. 2.116: PC1 format

Data field, meaning	Structure	Format		Length	Offset
Start of data record	HT			1	0
Subscriber (cost centre) number[1]	ttttt	I-	SP	5	1
Sort character	oo	00	-I	2	17
Date	yymmdd	00	-I	6	10
Start time	hh:mm	00	-I	5	17
Duration of connection	hhHmmMss	00	-I	8	23
Primary channel group / trunk card number / network interface number[2]	cbbpp	00	-I	5	32
Number of metering pulses	iiiii	00	-I	5	38
Call number dialled [3]	zzzzzzzzzzzzzzzzzzzz	I-	SP	20	44
Serial number	llll	00	-I	4	65
Carriage Return	CR			1	69
Line Feed	LF			1	70

1) Dialling determines whether Subscriber No. or CC No. is displayed.
As a cost centre number this field can be 5 or 9 digits long. Depending on the configured cost centre length ≤ 5, the field is 5 characters long. As of cost centre length ≥ 6, the length is 9 characters. If the cost centre length ≥ 6, all the offsets following the cost centre are incremented by 4 characters.

2) The primary channel group is output in position "c", the trunk card number in position "bb" and the network interface number in position "pp" (see example on page 412).

3) If "Data protection" is configured, the last 4 digits of the number are replaced by the space character "SP".

Example of PC1 format

The charge data is printed out every time the handset goes on-hook. This also applies in cases where an external connection is forwarded.

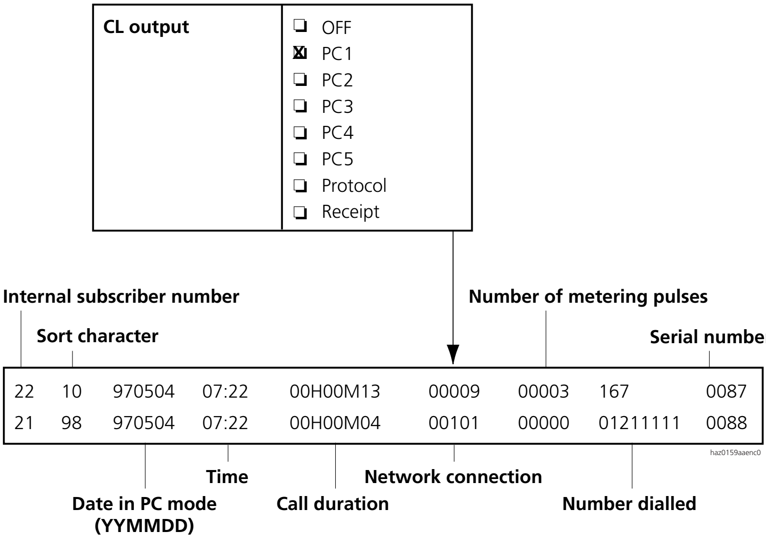


Fig. 2.153: CL output with PC1

PC2 format

This format is an extension of the PC1 format. Here the cost centre number is also output as a separate field, along with the DDI number.

Format structure

Tab. 2.117: PC2 format

Data field, meaning	Structure	Format		Length	Offset
Start of data record	HT			1	0
Subscriber number	ttttt	|-	SP	5	1
Cost centre number	kkkkkkkkk	|-	SP	9	7
Sort character	oo	00	-|	2	17
Date of start of connection	yymmdd	00	-|	6	20
Time of start of connection	hh:mm	00	-|	5	27

Data field, meaning	Structure	Format		Length	Offset
Duration of connection	hhHmmMss	00	-|	8	33
Primary channel group / trunk card number / network interface number[1]	cbbpp	00	-|	5	42
Direct dial number[2]	ddddddddddd	|-	SP	11	48
Number of metering pulses	iiiii	00	-|	5	60
Call number dialled [3]	zzzzzzzzzzzzzzzzzzzz	|-	SP	20	66
Serial number	llll	00	-|	4	87
Carriage Return	CR			1	91
Line Feed	LF			1	92

1) The primary channel group is output in position "c", the trunk card number in position "bb" and the network interface number in position "pp" (see example on page 418).

2) This is the direct dial number that is displayed as the CLIP to an external call partner.

3) If "Data protection" is configured, the last 4 digits of the number are replaced by the space character "SP".

Example of PC2 format

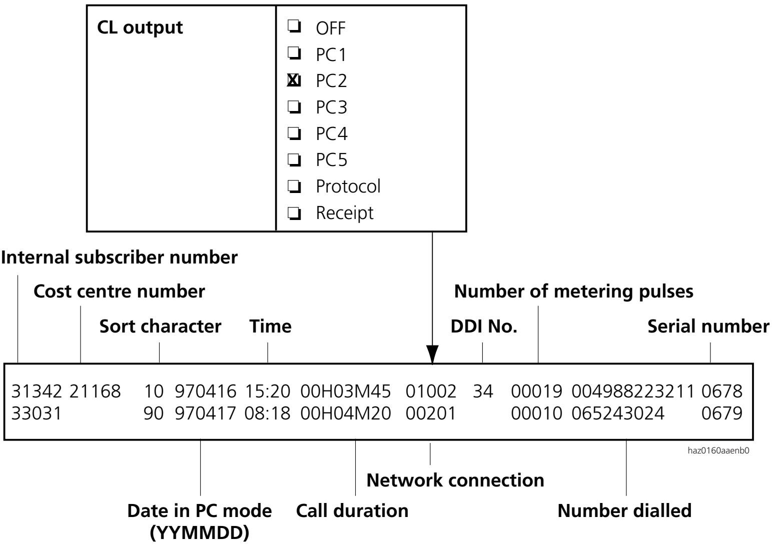


Fig. 2.154: CL output with PC2

PC3 format

The PC3 format has been expanded to include the fields TTA (Time to answer) and Seq. (Sequence). However, these fields are relevant only to incoming traffic.

PC4 format

If the feature "Least Cost Routing" is used in a PBX, this format can be used to carry out the corresponding analysis. This format features an additional field that contains the call number actually dialled by the PBX (Least-Cost-Routing function).

Tab. 2.118: PC4 format

Data field, meaning	Structure	Format		Length	Offset
Start of data record	HT			1	0
Subscriber number	ttttt	|-	SP	5	1
Cost centre number	kkkkkkkkk	|-	SP	9	7
Sort character	oo	00	-|	2	17
Date of start of connection	yymmdd	00	-|	6	20
Time of start of connection	hh:mm	00	-|	5	27
Duration of connection	hhHmmMss	00	-|	8	33
Primary channel group / trunk card number / network interface number[1)]	cbbpp	00	-|	5	42
DDI number	ddddddddddd	|-	SP	11	48
Number of metering pulses	iiiii	00	-|	5	60
Phone number dialled by PBX [2)]	zzzzzzzzzzzzzzzzzzzz	|-	SP	40	66
Phone number dialled by subscriber[2)]	zzzzzzzzzzzzzzzzzzzz	|-	SP	20	107
TTA (Time to Answer)	iii	00	-|	3	128
Sequence number	sss	00	-|	3	132
Serial number	llll	00	-|	4	136
Carriage Return	CR			1	140
Line Feed	LF			1	141

1) The primary channel group is output in position "c", the trunk card number in position "bb" and the network interface number in position "pp" (see example on page 420).

2) If "Data protection" is configured, the last 4 digits of the number are replaced by the space character "SP".

Example of PC4 format

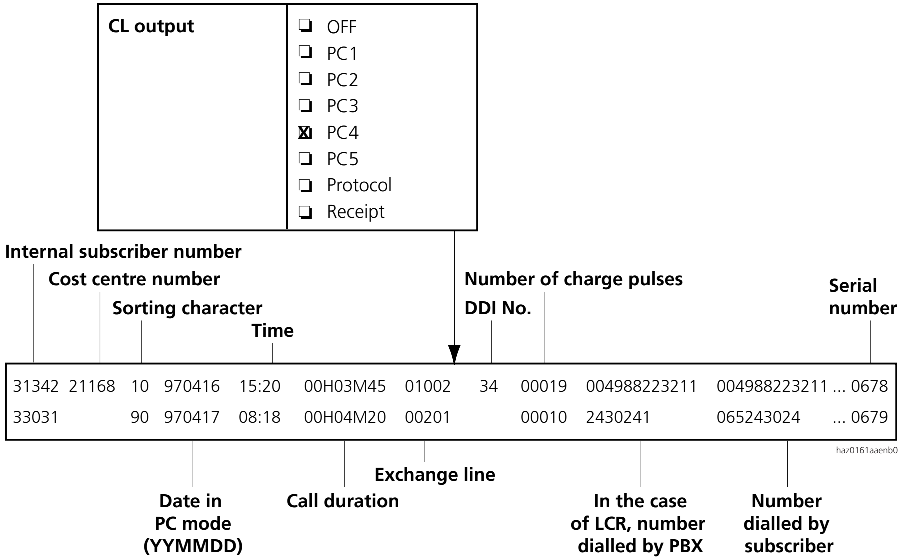


Fig. 2.155: CL output with PC4

Depending on the number dialled by the subscriber and the configuration in the LCR tables the number actually dialled by the PBX may be different or identical.

8 Features

The PBX provides a multitude of features that can be activated or operated by the user. All these features are described individually in this chapter and then listed alphabetically in an overview table in the next chapter. The features described in this chapter are as follows:

- Call Forwarding Unconditional functions: Call Forwarding Unconditional, Follow me, Call Forwarding on No Reply, Twin Mode / Twin Comfort, Take, Do not disturb, Substitution.
- Connections involving several subscribers: Enquiry call, Brokering, Three-party conference, Conference, Call transfer, Recall, Call acceptance
- Added features: Features that simplify day-to-day phone communication, e.g. Call waiting, Leave message, Park call, Team functions.
- Special features: Features that require a special application or hardware, such as Voice Mail or paging systems.

The remote control of features, the hotel function and the alarming facilities using external text messages are described at the end of the chapter.

Tab. 2.119: The following features / functions do not form part of this chapter:

Feature / function	Description / document
Routing functions	Chapter "Routing elements", page 209 and Chapter "Call Routing", page 267
Identification and presentation functions	Chapter "Identification Elements", page 178
Data-service functions	Chapter "Data services", page 353
Call logging	Chapter "Call logging (CL)", page 361
Functions specific to the system terminals	Operating Instructions
VoIP features in detail	Separate documentation for IP Gateway AIP
Voice Mail features	Documentation for the AVS Voice Mail System AVS

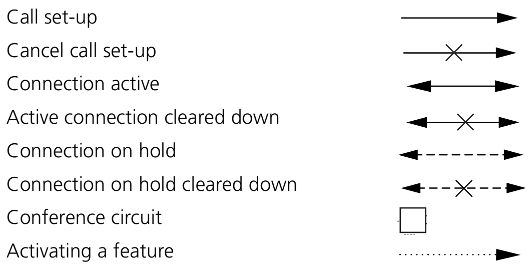
Description categories and terminology

Each feature consists of a detailed description with the following headings:

- Scenario
- Detailed Description
- Prefix and suffix dialling procedures

- System configuration
- Reference to Other Features

An illustration represents the feature scenario in a simple, clearly structured form. The following symbols are used:



Detailed Description

This heading contains:

- A description of the feature-relevant signalling on the system terminals.
- A definition of the scope within which the feature can be carried out.
- Remarks, tips or information on the sequence of operation of the feature or on exceptional cases.

Prefix and suffix dialling procedures

Features are controlled using procedures. A procedure can be triggered under three different sets of conditions:

- In prefix dialling: dialling takes place before any connection is made.
- In suffix dialling: dialling takes place during a connection or call.
- During the ringing phase: dialling takes place during the ringing phase of an incoming call

Depending on the nature of the feature a procedure is triggered either in prefix dialling, in suffix dialling, during the ringing phase or in several call states.

On Office terminals, subscriber-related features are activated or deactivated using the Foxkey, under which a variety of functions can be stored. Equipment-specific settings can be found in the relevant Operating Instructions for the terminals concerned.

System configuration

Designation of the parameters concerned in the system configuration and their settings.

Reference to Other Features

List of associated or linked features.

Information on the system terminals

Unless otherwise specified, information on the system terminals Office 35 also includes Office 35IP; Office 45 also includes Office 45pro; and Office 135 also includes Office 135pro.

Terminology

The following terms are used:

Tab. 2.120: Terms are used

Term	Usage
Internal subscribers	An internal subscriber is connected locally to the system.
External subscriber	An external subscriber is located in the public network (outside the private network).
PISN Subscriber	A PISN subscriber is connected to another node (PINX) of the private network (PISN: Private Integrated Services Network). Subscriber of a virtual PINX.
Subscriber A	1. Subscriber in a feature scenario (the person who sets up a call for example)
Subscriber B	2. Subscriber in a feature scenario (the person who answers subscriber A's call for example).
Subscriber C	3. Subscriber in a feature scenario (enquiry call between subscriber B and subscriber C for example)
Service	Function offered by the network provider and carried out in the public network, in particular an ISDN supplementary service.
Feature	Function provided by the system and carried out locally in the PBX.

8.1 Network services, authorizations and operation

8.1.1 ISDN services supported by the system

The system supports a whole series of ISDN supplementary services, which are provided by network providers in addition to ISDN bearer services.

External services and internal features

In this document a distinction is made between features and services.

Features designate functions that are provided locally in the PBX.

Services designation functions that are offered at the network interfaces by the public ISDN network provider and supported, i.e. used, by the PBX.

ISDN services are further differentiated into bearer services and supplementary services.

A certain number of functions such as the three-party conference with two external subscribers can be carried out both externally in the public network and internally in the PBX.

The example of a three-party conference is used to illustrate the interaction between the PBX and the public network.

Example of a three-party conference

The figures below show variants of three-party conferences with internal and external subscribers.

The left-hand side of the figure below shows a conference set up in the PBX with three internal subscribers (three-party conference feature); the right-hand side shows a conference in the public network with three exchange subscribers (three-party conference supplementary service):

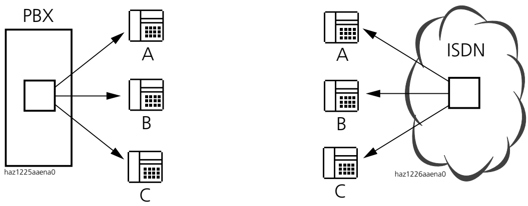


Fig. 2.156: Conference circuit feature and three-party conference supplementary service

The figure below shows a three-party conference connected in the PBX with one internal subscriber and two exchange subscribers:

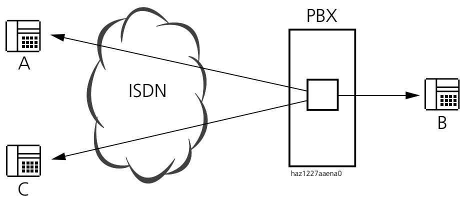


Fig. 2.157: Three-party conference feature with 1 internal and 2 external subscribers

The three-party conference feature is implemented locally in the PBX. Two B channels are seized as a result.

If the system requirements are met, the three-party conference with one internal and two external subscribes can also be transferred to the exchange.

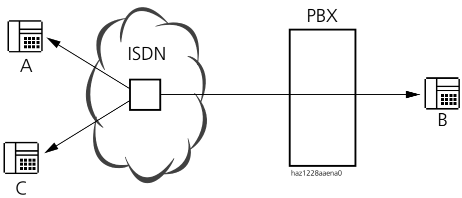


Fig. 2.158: Three-party conference as a service in the public network, with 1 internal and 2 external subscribers

The supplementary service three-party conference is activated locally but moved from the system to the public network. Only 1 B channel is seized as a result.

ISDN supplementary services supported

In the following overview ISDN supplementary services are categorized as follows:

- Identification services
- Connection services
- Rerouting services
- Call charge services
- Other services

As a rule network interfaces are wired as a point-to-point connection (P-P). However, the point-to-multipoint (P-MP) connection type is also possible. Not all ISDN supplementary services are provided by network providers on both connection types or supported by the PBX.

Identification services

Tab. 2.121: Identification services

Name of the service		Remark	P-P	P-MP
CLIP	Calling Line Identification Presentation	Displays the caller's number to the called party	✓	✓
CLIR	Calling Line Identification Restriction	Suppresses the display of the caller's number to the called party	✓	✓
COLP	Connected Line Identification Presentation	Displays the called party's number to the caller	✓	✓
COLR	Connected Line Identification Restriction	Suppresses the display of the called party's number to the caller	✓	✓
DDI	Direct Dialling In	Direct dialling	✓	–
MCID	Malicious Call Identification	Records malicious calls	✓	✓
MSN	Multiple Subscriber Number	Multiple subscriber number	–	✓

Connection services

Tab. 2.122: Connection services

Name of the service		Remark	P-P	P-MP
HOLD	Call Hold	Holds a connection in the exchange. Precondition for enquiry calls, brokering and three-party conferences in the exchange	–	✓
ECT	Explicit Call Transfer	Call transfer in the exchange	–	✓
CCBS	Completion of Calls to Busy Subscriber	Callback if busy in the exchange	✓	✓
3PTY	Three-Party Conference	Three-party conference in the exchange	–	✓

Rerouting services

Tab. 2.123: Rerouting services

Name of the service		Remark	P-P	P-MP
CFU	Call Forwarding Unconditional	CFU in the exchange, supported via */# procedure	✓	✓
CFB	Call Forwarding Busy	CFB in the exchange, supported via */# procedure	✓	✓
CFNR	Call Forwarding on No Reply	CFNR in the exchange, supported via */# procedure	✓	✓
CD	Call Deflection	Supported as a subscriber-related feature and used by the system to transfer CFU / CFNR / CD to the exchange.	–	✓
PARE	Partial Rerouting	Used by the system to transfer CFU / CFNR / CD to the exchange.	✓	–

Call charge services

Tab. 2.124: Call charge services

Name of the service		Remark	P-P	P-MP
AOC-D	Advice of Charge (During)	Call charge information during the call	✓	✓
AOC-E	Advice of Charge (End)	Call charge information at the end of the call	✓	✓

Other services

Tab. 2.125: Other services

Name of the service		Remark	P-P	P-MP
UUS-1	User-to-User Signalling	Signalling from one user to another user Supported only during setup and only for ISDN terminals on the S interface.	✓	✓
SUB	Subaddressing	Subaddressing	✓	✓
	Keypad Signalling	*/# procedures in the exchange	✓	✓

8.1.2 Notifications supported by the system

Notifications are used for transmitting information on a connection's current status and can be shown for example on the display of Office terminals. The notifications supported by the public ISDN network are also partly supported by the system, converted accordingly for private QSIG networks, or forwarded transparently to connected ISDN terminals.

Notifications sent to the public ISDN network by the PBX can be inhibited in the trunk group configuration using the parameter "Send notifications = No".

The following table provides an overview of the notifications supported by the PBX or forwarded transparently:

Tab. 2.126: Notifications supported:

Notification	Incoming on the terminal:		Outgoing	Meaning / Remarks
	Office	ISDN		
Remote hold	yes	transparent	yes	Subscriber on hold
Remote retrieval	yes	transparent	yes	Return to previous subscriber or connect with new subscriber
User suspended	yes	transparent	yes	Subscriber parked
User resumed	yes	transparent	yes	Subscriber retrieved
Conference established	yes	transparent	yes	Conference set up
Conference disconnected	yes	transparent	yes	Conference terminated
Call is diverting	yes[1)]	transparent	yes[1)]	Diverted call
Call is a waiting call	yes	transparent	yes	Call is a waiting call

[1)] Depending on the network provider, redirecting information is also transmitted with incoming calls, in addition to the notification. With outgoing calls the PBX also sends the redirection information instead of the notification (see also "Display for Call Forwarding Unconditional", page 191).

Note:
Notifications in networks are not supported via the S-external interface with the protocol DSS1.

8.1.3 Features in the private network

Here describes subscriber-related features in a PISN.

Standardized operation and signalling

The way in which a feature is actuated on the terminal and its signalling are identical whatever the network used (local, PISN or public network).

Scope of performance

The range of services in a PISN is determined by the following criteria:

- Local features of the system
- Type of networking (QSIG or virtual with DSS1)
- Offer available from the public network provider

Networking with QSIG

The standardized QSIG protocol supports a wide range of basic and supplementary services. The system supports the following services:

- Call Forwarding Unconditional
- Call transfer
- Hold
- Conference (three-party conference, variable conference, preconfigured conference)
- Brokering
- Enquiry call
- Callback if busy
- Call Forwarding on No Reply
- Do not disturb
- Recall
- Call Deflection
- Call Rejection

Under QSIG the activated feature is displayed on the system terminals (excl. Office 10 and Office 20), e.g. "On hold" or "Conference".

Virtual Networking in the ISDN Network

In a virtual networking or in a virtual PINX in the public network the following conditions have to be met:

- The feature is supported end-to-end by the public ISDN network.
- Service compatibility between private ISDN and public ISDN is guaranteed for the feature.

Example: Callback if busy

"Callback if busy" is supported within the private network. Compatibility for this feature between private network (QSIG protocol) and public network (DSS1 protocol) is guaranteed. It is possible to activate callback between A and C and between B and C (Fig. 2.159) if the public network supports the feature end-to-end.

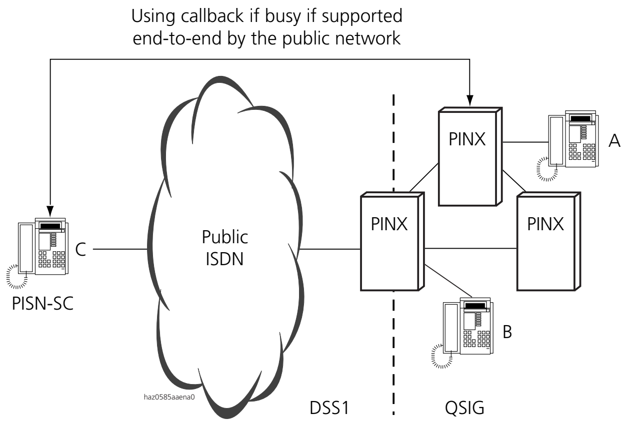


Fig. 2.159: Using a feature via the public network

Note:
With the overflow procedure (see "Testing overflow routing in the PISN", page 343) calls within the PISN are routed via the public network. In this case the conditions for networking with DSS1 apply. The ranges of services available can be restricted for such calls.

8.1.4 Features in the up-circuit PBX

A number of features can be triggered in the up-circuit PBX using route selection. For more details please refer to the Operating Instructions of the terminals and the features overview of the up-circuit PBX (see also "Down-circuit connection of a PBX with corded terminals", page 605).

8.1.5 Features operated via QSIG

In private QSIG networks a number of features can be operated on third-party PINXs via QSIG (applies only to Ascotel IntelliGate systems among themselves). In such cases it is irrelevant whether the QSIG networking is effected via a basic access, a primary rate access or Ethernet connections (IP network with AIP 6400), and which QSIG variant is selected as the protocol. The executing subscriber obtains (visual) confirmation as to whether or not the feature was successfully carried out.

Subscriber-unrelated Features

Subscriber-unrelated features are operated via abbreviated dialling numbers defined on the destination PINX and containing the corresponding procedures. These abbreviated dialling numbers are entered on one's own PBX as PISN subscribers in the numbering plan.

Note:
Make sure that these abbreviated dialling numbers are barred in the digit barring on all PINXs for unauthorized subscribers and that no names are assigned to the abbreviated dialling numbers (bypassing the digit barring).

The following procedures are supported:

Tab. 2.127: Subscriber-unrelated QSIG features

Feature	Activate	Deactivate
Operate switching groups	*85<Swith group><Pos.>	
Actuate door opener	*751, *752	
Switch relays	*755 - 757	#755 - 757
Enable/bar a one-off remote access	*754	#754
Answer coded ringing on general bell	*82	
Answer ring call on general bell	*83	
Dial door intercom	851,852 (initialisation values)	

Subscriber-related features

Operation of the subscriber-related features is subject to the definition of the PISN subscribers in one's own numbering plan. The features can be divided into two groups:

Features That Set up a Call Connection

The following subscriber-related features are supported by the PBX and can be activated via the keypad, the function key or the Foxkey:

Tab. 2.128: QSIG features with call connection

Feature	Activate
Fetch call	*86 PISN SC No.
Answer PSA	*82 PISN Pager No.

Features That Can Be Activated/Deactivated

All remote-controlled subscriber-related features listed in Tab. 2.298 are supported by the PBX and can be activated or reset via the keypad or function key. The only requirement is that remote control authorisation has been enabled for the PISN subscriber concerned, and that *06 is not barred in the internal digit barring for the subscriber executing the feature.

Example: Clearing CFU of a PISN subscriber: *06 <PISN SC No.> #21

System configuration

Due to the proprietary protocol attempts at operating a subscriber-related feature of an older or third-party PINX via QSIG can lead to incorrect interpretations. That is why the protocol extension can be inhibited in the trunk group configuration using the parameter "QSIG extension = No" (initialisation setting = No).

8.1.6 Subscriber-related authorizations

A class-of-service authorization in the subscriber configuration is required in order to run the following subscriber-related features.

- Activate call waiting
- Activate intrusion
- Set up announcement
- Activate MESSAGE (*38)
- Remote control
- Access to hotel functions
- Access to the system configuration (System Assistant authorization only)

- Prepare to take over a data connection
- Prepare to take over a voice connection
- Opening doors
- Operate system relays
- Operate switch group
- Force routing (LCR)
- Priority exchange allocation
- Remote maintenance
- Rerouting in the Exchange
- LCR
- Send SMS
- Change terminal password

In addition specific features and call destinations can be disabled using the internal digit barring (see "Digit Barring Facilities", page 288).

8.1.7 Exchange access authorizations

Exchange access authorization

Exchange-to-exchange connections have to be authorized to enable the features conference, Call Forwarding Unconditional and call transfer between two external subscribers (exchange-to-exchange connections can be further restricted, see "Exchange-to-Exchange Traffic", page 321).

Authorization to transfer exchange-to-exchange functions to the exchange

For exchange-to-exchange three-party connections to be transferred to the exchange, the relevant authorizations have to be enabled in the trunk group configuration.

For exchange-to-exchange Call Forwarding Unconditional to be transferred to the exchange, the relevant authorizations have to be enabled in the trunk group and subscriber configurations.

8.1.8 Operating the features on the terminal

Feature activation

With system terminals, features can be operated in the following ways:

- Menu-supported with Foxkey, for all system terminals with a display and for the PC Operator
- Using the Foxkey alone, for system terminals without a display
- Using function keys (see Tab. 2.129)
- With suffix dialled digits, in a specific connection status (e.g. suffix dialling digit 2 switches back and forth between 2 connections)

With commercially available system terminals by other manufacturers, features can be operated in the following ways:

- ISDN terminals:
  - By menu for ISDN services supported by the system on the S bus as per ETSI (see Tab. 2.311 and Tab. 2.312)
  - With */# procedures
- Analogue terminals: With */# procedures or control key

Changing the standard mode for DTMF

A number of functions can be operated in suffix dialling (e.g. for Voice Mail) by keying in DTMF dial signals. For this the terminal has to be switched to DTMF mode (Transparent Mode).

The new Office system terminals automatically switch over to DTMF mode as standard once a connection has been set up. This default setting can be altered for each terminal using the Foxkey or via AIMS.

If required, the terminals Office 20, Office 30, Office 40, Office 100 and Office 150 have to be switched over to DTMF mode manually by default after each call is set-up. This is done with a long-click of the *-key or with the Foxkey. This default setting can be altered for each terminal via AIMS only.

Configurable keys

The possibility of configuring keys with various functions means that system terminals offer a practical way of operating features (for more information see User's Guide of the terminals).

Configuration of a configurable key by the end user can be disabled by the system administrator using AIMS.

Tab. 2.129: Configurable keys of the system terminals

Number key	Function key	Team key
Office 25, Office 35, Office 45, Office 10, Office 20, Office 30, Office 40, Office 100, Office 135, Office 150, Office 155pro PC Operator Console	Office 25, Office 35, Office 45, Office 10, Office 20, Office 30, Office 40, Office 100, Office 135, Office 150, Office 155pro PC Operator Console	Office 35, Office 45, Office 30, Office 40 PC Operator Console

Number key

A number key can be used to store a frequently used external or internal call number. The number is then dialled with a simple keypress.

Tip:
The call number of a call distribution element can also be stored under a number key, provided it is listed in the internal numbering plan.

Function key

A frequently used function can be stored under a function key. The function is activated and deactivated simply by pressing the key. The system terminals Office 35, Office 45, Office 30 and Office 40 and the PC Operator support dual configurable keys: The activation and deactivation of the function are stored on the first and second storage space respectively. In this case pressing the key activates the function and the corresponding LED or display; pressing the key a second time deactivates both.

Function keys can be set up using the terminal or AIMS. Important functions are predefined and provided on the menu.

On system terminals without a display the function keys can only be set up using AIMS.

Team key

The team functions make it easier for members of a team (for example a sales or marketing team) to communicate with one another and stand in for one another where required. One team key is configured for each team member and allows the following functions and signalling states:

- Calling a team member using a simple keypress
- Signalling an incoming call for the team member and fetching the call using a simple keypress
- Signalling an existing connection to the team member (differentiating between internal and external call – only on Office 35, Office 45, Office 30, Office 40 and PC Operator)
- Depending on the terminal, other telephony functions (e.g. setting up an announcement to the team member)

8.1.9 Languages supported

The system supports a multitude of languages for the texts used on the user interface of system terminals and AIMS. The languages available vary depending on the user interfaces:

Tab. 2.130: Languages supported

Languages[1)]		System terminals	PC Operator	PBX	AIMS
Danish	da	✓	✓	✓	–
German	de	✓	✓	✓	✓
English	en	✓	✓	✓	✓
Estonian	et	✓	–	–	–
Finnish	fi	✓	✓	–	–
French	fr	✓	✓	✓	✓
Greek	el	✓	✓	–	–
Dutch	nl	✓	✓	✓	✓
Italian	it	✓	✓	✓	✓
Norwegian	no	✓	✓	✓	✓
Portuguese	pt	✓	✓	✓	✓
Swedish	sv	✓	✓	–	✓

Languages[1)]		System terminals	PC Operator	PBX	AIMS
Spanish:					
• Standard Spanish (castilliano)	es	✓	✓	✓	✓
• Basque (euskera)	eu	✓	–	–	–
• Galician (gallego)	gl	✓	–	–	–
• Catalan	ca	✓	–	–	–

1) Other languages may be added

System Terminals

Operating languages on the terminals:

- All the languages listed are available.
- Can be set via menu and AIMS.
- Office 100 and Office 150 support only 10 languages. The choice depends on the distribution channel.
- Cordless terminals in the "Out of Range" state and GAP terminals have thirteen languages stored locally (excl. the Spanish regional languages).
- The System Assistant in Office 45 automatically takes over the language set in the terminal.
- Standard text and event messages are available in the specified languages with the exception of Greek. A country-specific language as well as de, fr, and en are provided with the basic licence. The default language is specified for each country after initialization.

PC Operator Console

Operating languages on the PC Operator:

- All the languages listed are available
- Can be set via the menu

PBX

Operating, display and output language of PBX-generated texts:

- Operating languages for PBX settings via Office 45: Choice of language depends on the operating language selected on the terminal

- Display language for event messages. Can be set in the Fault & Maintenance Manager
- Output language for call logging can be set on the Account Manager.
- Office 45: Can be set via the menu and AIMS

Initialization values

- During initialization the currency is defined by country based on international abbreviations. It can be configured in all specified languages at a later time with AIMS.
- The title for an OCL / ICL printout is defined after initialization per country based on international abbreviations. It can be configured in all specified languages at a later time with AIMS. This also applies to the language of the OCL / ICL printouts (excluding Greek).

AIMS

AIMS operating languages:

- All the languages listed are available.
- Can be set via the menu. Restart the application after selecting a language.

8.2 Call Forwarding Unconditional functions

8.2.1 Call Forwarding Unconditional (CFU)

Calls intended for B are diverted to destination C.

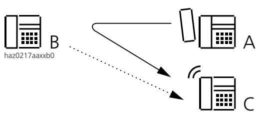


Fig. 2.160: Call Forwarding Unconditional

Call forwarding responds differently depending on the system configuration and the procedure used. The various CFU types are as follows:

- CFU to a variable destination:
  The subscriber specifies the chosen call forwarding destination on his terminal. This CFU can be either unconditional or only if busy.

- Predefined CFU:
  The call forwarding is made unconditionally to a destination entered in the subscriber configuration. This destination is also used with the Leave message feature if the caller is unable to read messages on his terminal.

- CFU if unobtainable:
  Where a call is to be diverted to in the event of unavailability can be defined for DECT handsets in system configuration.

Note:
The unconditional call forwarding to a preconfigured destination can be overwritten with *67 procedure (see Tab. 2.132).

Detailed Description

Tab. 2.131: Call Forwarding Unconditional

Interface	Operating sequence / signalling on the terminal	Scope
B	• Obtains acknowledgement tone when activating and resetting the CFU • If "CFU first ring = yes" is configured and C is an internal subscriber, B obtains an attention tone (short ring) and has 5 seconds in which to answer the call.	Restriction: B can only activate a single Call Forwarding Unconditional. Each new one overwrites the old one.
C		Possible destinations: • Subscribers: internal, external[1], PISN[2] • PS/coded ringing • UG: 17 to 21 (2025 / 2045 system) or 25 to 29 (2065 system) and user groups configured as "large". • Standard text (leave message) Requirement: C is not protected against Call Forwarding Unconditional (*02).

1) see "Call Forwarding Unconditional to exchange", page 441.

2) The conditions for Call Forwarding Unconditional to exchange apply to PISN subscribers in the public network or on a virtually connected PINX.

CFU to a call distribution element:
The internal number of a call distribution element is not possible as the destination for a Call Forwarding Unconditional.

Call forwarding chains:

- Internal: CFU chains can be set up locally (maximum 20);
- In the PISN: CFU chains are permitted. They are however restricted by the transit counter.

Note:
The procedure *67 (CFB) interrupts a chain of call forwarding.

Call forwarding loops:

- Internal: not permitted.
- In the PISN: restricted by the transit counter.

C is the only subscriber who can still reach B.

Prefix Dialling Procedures

Tab. 2.132: Call Forwarding Unconditional: Procedures

	*/# procedure
Activate CFU / CFB to any subscriber No.	*21 destination No./ *67 destination No.
Activate CFU / CFB to SC last configured	*21# / *67#
Clear CFU / CFB	#21 / #67
Activate preconfigured CFU	*22
Clear preconfigured CFU	#22
Activate CFU to standard text	*24 text No. [param.] #
Clear CFU to standard text	#24
Activate CFU to PS/general bell (coded ringing)	*28
Clear CFU to PS/general bell (coded ringing)	#28
Protect (own set) against CFU	*02
Allow CFU (to own set)	#02

System configuration

Tab. 2.133: Call Forwarding Unconditional: System configuration

Parameter	Parameter value	Remarks
Predefined CFU	<SC No.>	
CFU first ring	Yes / No	
Rerouting in the Exchange	Yes / No	Subscriber Configuration
Rerouting in the Exchange	Yes / No	Trunk group configuration
Wait for connection	Yes / No	see "Call Forwarding Unconditional to exchange", page 441
Forward to the last mail-box	Yes / No	see "Response with Call Forwarding Unconditional:", page 481

Reference to Other Features

Features:
- "Leave message", page 496
- "Follow me", page 444
- "Call Forwarding on No Reply (CFNR)", page 445
- "User group: Logging in and logging out", page 533
- "Searching via a paging system", page 536
- "Deflecting a call during the ringing phase (CD)", page 448"

Call Forwarding Unconditional to exchange

Settings for exchange-to-exchange traffic (see also "Exchange access authorizations", page 433)

- Exchange-to-exchange connection enabled:
  - External and internal calls are diverted to an external destination; a first-ring CFU is not carried out. Requirement: "Subscriber with direct dial" is defined.
  - If the conditions for transferring the Call Forwarding Unconditional to the exchange are also met, the connection is transferred to the network (see "Transferring Call Forwarding Unconditional to the Exchange", page 328).

Note:
Exchange-to-exchange connections can be further restricted, see "Exchange-to-Exchange Traffic", page 321.

- Exchange-to-exchange connection not enabled:
  - External calls are not diverted to an external destination.
  - Internal calls are diverted to an external destination.

Calls that reach the subscriber via user group are not diverted externally. For this reason the PBX switches the subscriber out of the user group if he defines an external CFU.

A user group in use must not be empty, which means that the last remaining subscriber cannot activate CFU to an external destination.

"Wait for connection" setting

Specifies whether a Call Forwarding Unconditional of an external call to the exchange is always switched through or only if the called party answers a call (and a connection is therefore set up):

- "Wait for connection = no"
  The Call Forwarding Unconditional is always switched through.
- "Wait for connection = yes"
  The Call Forwarding Unconditional is switched through only if a connection is set up.
  If the destination subscriber is busy or unobtainable, this setting ensures that the caller does not incur charges for the connection up to the PBX.

Example

Call Forwarding Unconditional to a mobile network subscriber who has switched off his handset:

- If "Wait for connection = no" has been set, the Call Forwarding Unconditional will be switched through: The callers will obtain a spoken text provided by the mobile network provider, indicating that the required subscriber cannot be reached at present.
- If "Wait for connection = yes" is set, the Call Forwarding Unconditional is not switched through and the caller will obtain the ring-back tone.

Scope

This feature is available only with stand-alone PBXs and gateway PINXs.

Examples of Call Forwarding Unconditional

The following examples illustrate three different cases of call distribution:

- Digital network interface without DDI or DDI number to subscriber.
- Digital network interface with DDI number to subscriber + UG busy.
- Digital network interface with DDI number to subscriber + KT and subscriber + KT busy.

Digital network interface without DDI or DDI number to subscriber

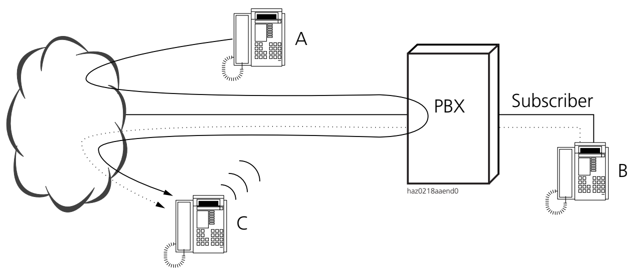


Fig. 2.161: Digital network interface without DDI or DDI number to subscriber

- B makes a CFU to C.
- A calls B, PBX sets up direct connection with C, C rings.
- If subscriber C is busy, A obtains the busy tone.

Digital network interface with DDI number to subscriber + UG busy

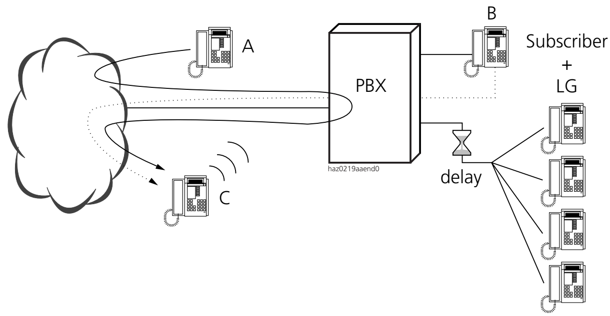


Fig. 2.162: DDI number to subscriber + UG busy

- UG is delayed.
- B makes a CFU to C.
- A calls B, PBX sets up direct connection with C, C rings.
- The user group will become activ, irrespective of configuration of the parameter "Wait for connection".
- If subscriber B is busy, A obtains the busy tone.

Digital network interface with DDI number to subscriber and KT and subscriber + KT busy

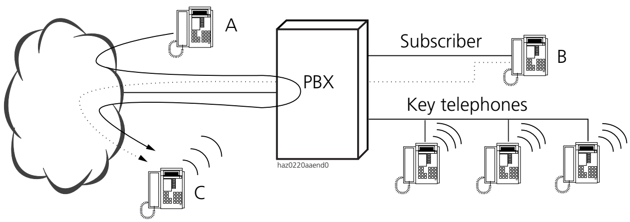


Fig. 2.163: DDI number to subscriber + KT and subscriber + KT busy

- B makes a CFU to C.
- A calls B, PBX sets up direct connection with C, C rings.
- The KT terminals with the line key also ring.
- If KT line is busy and C is busy, A obtains the busy tone.
- If C is busy, the KT line will ring. A obtains ring-back tone.

8.2.2 Follow me

Subscriber B wants to divert calls originally made to his own terminal to a terminal C, where he is currently located. He therefore configures a Call Forwarding Unconditional directly on destination terminal C.

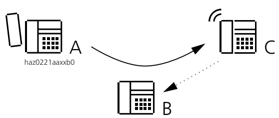


Fig. 2.164: Follow me

Detailed Description

Tab. 2.134: Follow me

Interface	Operating sequence / signalling on the terminal	Scope
C	Once the feature has been activated, the subscriber obtains an acknowledgement tone.	Possible interfaces: Internal

The call forwarding from B to C remains active until subscriber B cancels Follow me on his own terminal.

The functions configured on the subscriber's own terminal (e.g. exchange access) are not transferred to the destination terminal.

A call forwarding already activated will be overwritten by Follow me.

Follow me will interrupt any Call Forwarding Unconditional chains.

Prefix Dialling Procedures

Tab. 2.135: Follow me: Procedures

	*/# procedure
Activate Follow me on the destination set	*23 SC No. B
Clear Follow me on the subscriber's own terminal	#23

System configuration

Tab. 2.136: Follow me: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:
- "Call Forwarding Unconditional (CFU)", page 438

8.2.3 Call Forwarding on No Reply (CFNR)

Unlike Call Forwarding Unconditional, the call to subscriber B's telephone set is initially signalled in the normal way when CFNR is activated. If the called party B does not answer the call after (0), 3, 5 or 7 ringing cycles, the call will also be signalled (in parallel) on the terminal of subscriber C, who has been forwarded.

If the call was forwarded to C and was not answered by B, the next call will immediately be signalled to both subscribers B + C. The delay in the call to C is reactivated only once the call has been answered directly by called party B.

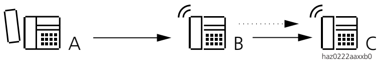


Fig. 2.165: Call Forwarding on No Reply

Call Forwarding on No Reply responds differently depending on the system configuration and the procedure used.

- Normal CFNR:
  The subscriber specifies the chosen call forwarding destination on his terminal.
- Preconfigured CFNR:
  The call forwarding is made to a destination entered in the subscriber configuration.
- CFNR can also be effected for both types if subscriber B is busy. For this the option "CFNR if busy" must be activated in A's subscriber configuration.

Detailed Description

Tab. 2.137: Call Forwarding on No Reply

Interface	Operating sequence / signalling on the terminal	Scope
B	Once the feature has been activated, B obtains an acknowledgement tone.	
C		Possible destinations: • Subscribers: internal, external[1], PISN • PS/coded ringing • UG: 17 to 21 (2025 / 2045 system) or 25 to 29 (2065 system) and user groups configured as "large". Requirement: C is not protected against calls (Do not disturb, *26) or Call Forwarding Unconditional (*02).

1) If caller A is an external subscriber or a virtual PISN subscriber, the settings authorising exchange-to-exchange traffic (see "Call Forwarding Unconditional to exchange", page 441) will have to be observed. (If the connection is not authorized, the call is not forwarded.)

Note:
The internal number of a call distribution element is not possible as the destination for Call Forwarding on No Reply.

Chain of Call Forwarding on No Reply:

- Internal: Call Forwarding on No Reply are not chained together locally. (The call is routed to C but cannot be routed any further.)
- In the PISN: CFNR chains within the PISN are possible if B and C are connected to different PINXs.

Note:
CFNR chains in the PISN result in long ringing times.

For the delay always to be active, "CFNR immediate ring = No" must be configured throughout the system.[1]

Note:
If a CDE destination is defined under "Default CF if no answer" in the subscriber configuration, it is possible to configure whether CFNR or the default call forwarding is to be executed (see also "Default Call Forwarding per Subscriber", page 277).

CFNR to the exchange

With Call Forwarding on No Reply to the public or private network the subscriber remains activated in his user group.
Incoming calls on the user groups that reach this subscriber are therefore routed to the CFNR destination (this applies to ordinary user groups, not large user groups, see "User Group", page 236).

Note:
If in a user group several subscribers have configured CFNR to the exchange, it may not be possible to set up some of the calls. The number of calls that can be set up depends on the resources available at the time (free B channels in the corresponding trunk group).

Prefix dialling procedures

Tab. 2.138: Call Forwarding on No Reply: Procedures

	*/# procedure
Activate CFNR to subscriber	*61 destination No.
Clear CFNR to subscriber	#61
Activate CFNR to subscriber last configured	*61#
Clear CFNR to subscriber last configured	#61

[1] See country-specific initialization configuration.

	*/# procedure
Activate preconfigured CFNR	*62
Clear preconfigured CFNR	#62
Activate CFNR to PS / general bell (coded ringing)	*68
Clear CFNR to PS / general bell (coded ringing)	#68
Protect (own set) against CFNR	*02
Allow CFNR (to own set)	#02

System configuration

Tab. 2.139: Call Forwarding on No Reply: System configuration

Parameter	Parameter value	Remarks
CFNR if busy	Yes / No	
Predefined CFNR	<SC No.>	
CFNR if unobtainable	Yes / No	
CFNR immediate ring	Yes / No	
CFNR, forwarding time	<3, 5 or 7 rings>	
Rerouting in the Exchange	Yes / No	Subscriber Configuration
Rerouting in the Exchange	Yes / No	Trunk group configuration

Reference to Other Features

Features:

- "Call Forwarding Unconditional (CFU)", page 438
- "Searching via a paging system", page 536
- "Deflecting a call during the ringing phase (CD)", page 448

8.2.4 Deflecting a call during the ringing phase (CD)

Calls intended for B are deflected to destination C during the ringing phase. (CD: Call Deflection). In such cases the call is not forwarded automatically but manually by subscriber B. Unlike call forwarding the call is signalled only at destination C after it has been forwarded.

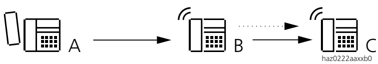


Fig. 2.166: Forwards a call during the ringing phase

Detailed Description

The response and properties of Call Deflection are similar to those of CFU Unconditional.

Tab. 2.140: Call Deflection

Interface	Operating sequence / signalling on the terminal	Scope
B	Once the feature has been activated, B obtains an acknowledgement indication on his display.	• Office 25, 30, 35, 40, 45, 100, 130, 150, 155 via the Foxkey • Office 20 via the function key (configuration possible only with AIMS) • ISDN terminals that support the feature
C		Possible destinations: • Subscribers: internal, external[1], PISN • PS/coded ringing • UG Requirement: C is not protected against Call Forwarding Unconditional (*02).

[1] If caller A is an external subscriber or a virtual PISN subscriber, the settings authorising exchange-to-exchange traffic (see "Call Forwarding Unconditional to exchange", page 441) will have to be observed. (If the connection is not authorized, the call is not forwarded.)

Other properties:

- The internal number of a call distribution element is not possible as the destination for Call Deflection.
- If the called subscriber is busy and the calling subscriber activates call waiting, the call can also be forwarded. The response and options available as the same as for a free subscriber.
- Calls on the line of a key telephone or an Operator Console cannot be forwarded (exception: the Personal key on an Operator Console).
- If the call is not answered at the destination, a recall is not made.
- If an attempt is made to forward the call to an invalid or busy internal call number, the function is not executed and the ringing is signalled as before. By contrast Call Deflection to an external subscriber is always executed.

Procedures During the Ringing Phase

Tab. 2.141: Call Deflection: Procedures

	System terminals (excl. Office 10/20)
Deflecting a call during the ringing phase (Call Deflection)	1. 2. The call number is entered via the keypad, using dialling by name, the call list, etc. 3.

System configuration

Tab. 2.142: Call Deflection: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- "Call Forwarding Unconditional (CFU)", page 438
- "Call Forwarding on No Reply (CFNR)", page 445
- "Call waiting", page 477
- "Reject call", page 450

8.2.5 Reject call

Calls for B are rejected during the ringing phase. This immediately clears down the call set-up and therefore the ringing at B. Subscriber A obtains the busy tone.

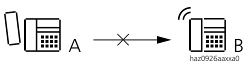


Fig. 2.167: Rejecting a call during the ringing phase

Detailed Description

Tab. 2.143: Reject call

Interface	Operating sequence / signalling on the terminal	Scope
B	The activation of the feature is not confirmed.	• Office 25, 30, 35, 40, 45, 100, 130, 150, 155 via the Foxkey • ISDN terminals that support the feature. (The response after rejection varies from one manufacturer to the next)

Exceptions where rejection results in forwarding:

- If the free subscriber rejects a call but has also activated CFB, the call is forwarded to the CFU destination.
- If an alternative destination is entered in the CDE configuration under "CDE if busy", the call is fowarded to the corresponding destination after its rejection.

Other properties:

- If the called subscriber is busy and the calling subscriber activates call waiting, the call can also be rejected.
- A configured CFNR, a default call forwarding or an entry in the CDE configuration under "CDE if no answer" are not executed after a call has been rejected.
- If a subscriber in a user group with other subscribers rejects a call the other subscribers will continue to ring. If all the UG members reject the call, the call setup is cleared down and the calling subscriber obtains the busy tone.

Procedures During the Ringing Phase

Tab. 2.144: Rejecting a call: Procedures

	System terminals (excl. Office 10/20)
Rejecting a call during the ringing phase	

System configuration

Tab. 2.145: Rejecting a call: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- "Call Forwarding Unconditional (CFU)", page 438
- "Call Forwarding on No Reply (CFNR)", page 445
- "Call waiting", page 477
- "Deflecting a call during the ringing phase (CD)", page 448

8.2.6 Twin Mode / Twin Comfort

Twin Mode and Twin Comfort are used to couple a user's corded terminal and DECT handset.

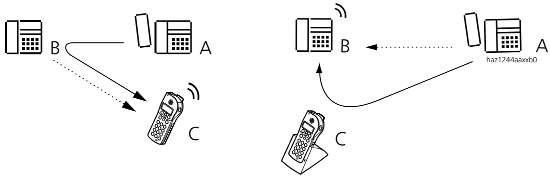
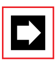


Fig. 2.168: Twin Mode / Twin Comfort

Twin Mode automatically activates a Call Forwarding Unconditional from subscriber B to subscriber C as soon as the cordless terminal (subscriber C) is removed from the charging bay. Conversely, a call for C is automatically diverted to B if C is in the charging bay.

While Twin Comfort provides the same functionality as Twin Mode, it also temporarily replaces the following phone lists of the DECT handset with the corresponding lists of the corded terminal:

- Private phone book
- Unanswered call list
- Answered call list
- Last number redial list:
- Message list



Note:
AIMS is used to determine whether Twin Mode or Twin Comfort is available on the DECT handset. If the Twin Comfort function is activated, no other function can be allocated to the charging contact; instead it has to be deactivated again via AIMS.

Detailed Description

Tab. 2.146: Twin Mode / Twin Comfort

Interface	Operating sequence / signalling on the terminal	Scope
C	• Activating via the charging bay • "Twin Mode" or "Twin Comfort" indicated on the terminal display	Activation is possible only if no Call Forwarding Unconditional is active.

System configuration

Tab. 2.147: Twin Mode / Twin Comfort: Key configuration

Function type	Note
In AIMS or on the handset the charging bay is configured as "Key" for "Twin Mode" or "Twin Comfort".	Twin Mode and Twin Comfort are mutually exclusive.

8.2.7 Take

The Take function allows subscribers to transfer calls or data from the corded terminal to their DECT handset or vice versa, without having to interrupt the connection or forward the ringing.

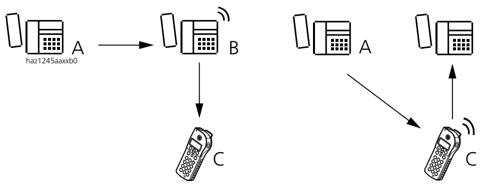


Fig. 2.169: Take

In the left-hand example subscriber A calls subscriber B, who transfers the call to his DECT handset C with a keystroke. In the right-hand example subscriber A calls on the DECT handset C and the call is continued on the corded terminal A. In both cases the caller will not be aware that the call has been transferred.

Detailed Description

Tab. 2.148: Take

Interface	Operating sequence / signalling on the terminal
C	• Activating via the handset's configurable key • Displays names and numbers of the registered subscriber as well as "Take" on the DECT handset display

System configuration

Tab. 2.149: Take: Key configuration

Function type	Note
In AIMS or on the handset the following command is used to prepare a configurable key to take subscriber 1's call on subscriber 2: I*87(No.SC1)*(No.SC2)#XI*88#	Take requires that the parameter "Voice handover" is set on "yes".

Note:
The Take function can also be enabled for data handover from one PC to another using the "Data handover" parameter in the subscriber configuration. (See also "Taking over an active connection", page 517.).

8.2.8 Do not disturb

To ensure that subscriber B is not disturbed, all incoming calls are automatically diverted to an alternative destination C, which has to be specified using the system configuration.

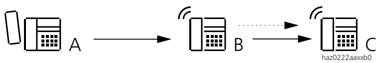


Fig. 2.170: Do not disturb

Detailed Description

Tab. 2.150: Do not disturb

Interface	Operating sequence / signalling on the terminal	Scope
B	Once the feature has been activated, B obtains an acknowledgement tone.	
C		Possible destinations: • Subscribers: internal, PISN[1)] • Operator Console Requirement: C is not protected against calls (Do not disturb, *26) or Call Forwarding Unconditional (*02).

1) The settings authorising internetwork traffic need to be observed for PISN subscribers in the public network or on a virtually connected PINX (see "Exchange-to-Exchange Connections", page 321). (If the connection is not authorized, the call is not forwarded.)

C is the only subscriber who can still reach subscriber B.

The alternative destination C is valid for the entire system.

A destination for Do not disturb cannot be diverted to the exchange.

Prefix Dialling Procedures

Tab. 2.151: Do not disturb: Procedures

	*/# procedure
Activate Do not disturb	*26
Clear Do not disturb	#26

System configuration

Tab. 2.152: Do not disturb: System configuration

Parameter	Parameter value	Remarks
Do not disturb	<CFU destination>	

Reference to Other Features

Features:

- "Call Forwarding Unconditional (CFU)", page 438

8.2.9 Proxy

In the attendant's absence, calls to Operator Console B can be forwarded to a pre-configured substitution terminal C.

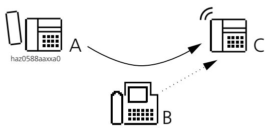


Fig. 2.171: Proxy activated

Detailed Description

Tab. 2.153: Substitution

Interface	Operating sequence / signalling on the terminal	Scope
B	• All the PBX operator consoles indicate the fact that the proxy is activated. • When the proxy is activated, calls are still signalled at the operator console but no longer acoustically.	Possible interfaces: • Operator Console
C		Possible destinations: • Subscribers: internal, PISN • General Bell • Both (subscriber + general bell) Requirement: C is not protected against calls (Do not disturb, *26) or Call Forwarding Unconditional (*02).

The substitution can only be switched on and off at an operator console and is then valid for all the operator consoles in the PBX.

Personal calls are not diverted.

Calls that were signalled on the operator console before proxy was switched on are not diverted.

If "Subscriber" destination for the substitution is busy, caller A obtains the busy tone. Call waiting is not automatic.

If "General bell" is configured as the substitution's destination, the call is placed in the general bell's queue and caller A obtains the ring-back tone.

Prefix Dialling Procedure

Tab. 2.154: Substitution: Procedure

	Office 45 as an Operator Console
Switch proxy on and off	

System configuration

Tab. 2.155: Substitution: System configuration

Parameter	Parameter value	Remarks
Substitution	<SC No.>.	

Reference to Other Features

Features:
- "Call Forwarding Unconditional (CFU)", page 438

8.2.10 DECT Follow Me

The system is such that a DECT call cannot be handed over from one system to another (Handover). However, with the new DECT Follow Me feature the accessibility of DECT subscribers in a PISN has been improved. This allows a DECT subscriber to be reached without delay in 4 PINXs (DECT Follow me must not be confused with the feature "Follow me", page 444).

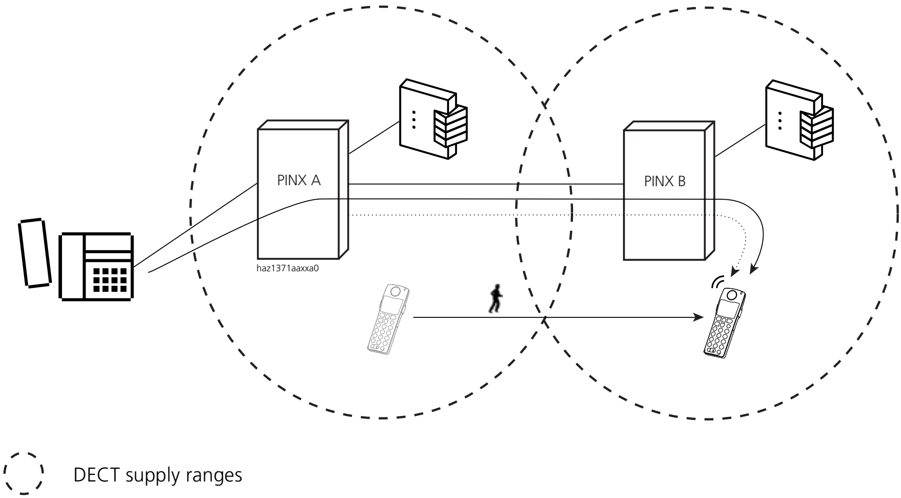


Fig. 2.172: Automatic activation of DECT Follow me

DECT Follow Me in a Network with 2, 3 or 4 Systems

This configuration can be used to find a DECT handset in up to 4 systems without delay. The handset has to be logged on in all 4 systems and the system search mode be configured to "Automatic" on the handset.

Detailed Description:

The handset is logged on under System A on its own PBX and under B, C and D on the other PINX. On each PBX a number is configured under the corresponding DECT subscriber ("Follow Me Number") which is dialled automatically as soon as the handset registers with the system. On a PINX this action activates a diversion from the handset's own PBX to the system onto which the handset has just registered. If the handset registers back with its own PBX, the previously activated diversion is deactivated.

Other properties:

- Twin Mode is possible on one's own PBX
- Not possible in virtual networks
- Possible only with Office 135 and Office 155pro

Notes:
If the "Follow Me Number" cannot be dialled when the handset registers with a system, because the QSIG link is either interrupted or overloaded, the handset will be unable to register. It will continue trying until registration is successful.

With Office 135 and Office 155pro "Long-Click 1" switches over only temporarily to manual search of the next system. What is relevant is the setting in the handset's configuration menu. This prevents accidentally switching the handset over from "automatic" to "manual" system search and therefore unintentionally deactivating DECT Follow Me.

The following configuration option can be used as an alternative to DECT Follow me in a network with only 2 systems:

CFU if Unobtainable in a Network with 2 Systems

If the majority of phone calls take place in the same system, the AIMS parameter "CFU if Unobtainable" can be used to find the handset on the second system:

- The handset has the same internal number in both systems.
- With the first call the diversion takes approx. 13 seconds. The diversion is immediate as of the second call.
- Twin Mode is possible on one's own PBX
- Also possible in virtual networks

Application Notes:
Application Notes are available for both configuration options (see https://pbxweb.aastra.com).

8.3 Connections involving several subscribers

8.3.1 Music on hold

In the following chapters a subscriber is put on hold in each case in connection with the features Hold, Brokering, Three-Party Conference and Call Transfer. Depending on the configuration selected for the parameter "Music on Hold" the subscriber on hold will obtain the following:

"None": The subscriber hears nothing.

"External": Music from the audio equipment connected to the PBX's music input.

"Internal": Internal PBX melody.

"Hold tone": Regularly recurring dual tone.

There is a choice of 8 different volume levels.

These settings also apply to a caller on hold after the Courtesy announcement, if the playback mode is set on "Music".

The settings apply throughout the system.

8.3.2 Hold (enquiry call)

An A – B connection is put on hold if one of the callers, e.g. subscriber B wants to set up an enquiry call connection with C.

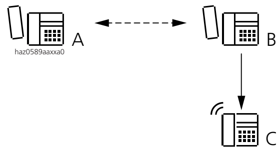


Fig. 2.173: Putting a call on hold

Detailed Description

Tab. 2.156: Hold (enquiry call)

Interface	Operating sequence / signalling on the terminal	Scope
A	"Music on hold" is played to subscriber A, who is on hold[1)]	Possible interfaces: internal, external, PISN
C		Possible interfaces: internal, external, PISN

1) With hold in the public exchange, the signalling depends on the network provider.

If A is on hold and B hangs up before setting up a ringing or call connection to C, B's terminal will ring continuously for 10 seconds. As soon as B picks up the handset, he is again connection with A.

If A is on hold and B waits for more than 10 seconds before setting up a ringing or call connection to C, B will obtain the busy tone. The return to the initial connection is not automatic.

Suffix dialling procedures

Tab. 2.157: Hold (enquiry call): Procedures

	System Terminals	a/b terminal
Set up internal enquiry call	with or without call preparation	R SC No. (R = control key)
Setting up an inquiry call to a subscriber of the up-circuit PBX (condition: the user's own PBX is in an analogue down-circuit connection and the existing call connection already seizes a trunk line to the up-circuit PBX)	via function key with control key function (macro ".*42")	R*42 SC No.

System configuration

Tab. 2.158: Hold (enquiry call): System configuration

Parameter	Parameter value	Remarks
Hold in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Music on hold	None / External / Internal / Hold tone	Throughout the system

Reference to Other Features

Features:
- "Brokering (switching back and forth between two calls)", page 462
- "Enquiry call with return to the initial call", page 461
- "Three-party conference from an enquiry call", page 464
- "Call transfer (switching)", page 467"
- "Recall", page 472
- "Call acceptance", page 474

8.3.3 Enquiry call with return to the initial call

A subscriber (B) can initiate an inquiry call connection during a call (A – B) and as a result hold a short conversation with another call partner (C), without interrupting the first connection. The original connection is restored once the enquiry call is completed.

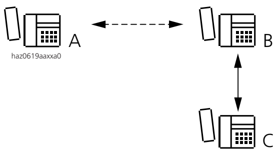


Fig. 2.174: Enquiry call

Detailed Description

Tab. 2.159: Enquiry call with return to initial call

Interface	Operating sequence / signalling on the terminal	Scope
A	"Music on hold" is played to subscriber A, who is on hold[1)]	Possible interfaces: internal, external, PISN
C		Possible interfaces: internal, external, PISN

1) With hold in the public exchange, the signalling depends on the network provider.

Suffix dialling procedures

Set up enquiry call: see "Hold (enquiry call)", page 460.

Tab. 2.160: Enquiry call with return to the initial call: Procedure

	System Terminals	a/b terminal
Return to the initial call	with the disconnect key	• with R1 (R = control key) or wait for more than 2 seconds after pressing the control key • by putting the handset on-hook and then taking it off-hook again after recall

System configuration

Tab. 2.161: Enquiry call with return to the initial call: System configuration

Parameter	Parameter value	Remarks
Hold in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Music on hold	None / External / Internal / Hold tone	Throughout the system

Reference to Other Features

Features:

- "Hold (enquiry call)", page 460
- "Brokering (switching back and forth between two calls)", page 462
- "Three-party conference from an enquiry call", page 464
- "Call transfer (switching)", page 467
- "Call waiting", page 477

8.3.4 Brokering (switching back and forth between two calls)

A subscriber can switch back and forth as often as required between his call party and the subscriber on hold.

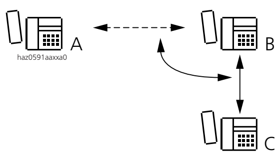


Fig. 2.175: Brokering

Detailed Description

Tab. 2.162: Brokering (switching back and forth between two calls)

Interface	Operating sequence / signalling on the terminal	Scope
A	"Music on hold" is played to subscriber A, who is on hold[1)]	Possible interfaces: internal, external, PISN
C		Possible interfaces: internal, external, PISN

1) With hold in the public exchange, the signalling depends on the network provider.

Brokering is also possible from a conference with a subscriber.

Suffix dialling procedures

Tab. 2.163: Brokering (switching back and forth between two calls): Procedures

	System Terminals	a/b terminal
Brokering	• • With digit suffix dialling: 2	with R2 (R = control key)

System configuration

Tab. 2.164: Brokering: System configuration

Parameter	Parameter value	Remarks
Hold in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Music on hold	None / External / Internal / Hold tone	Throughout the system

Reference to Other Features

Features:

- "Hold (enquiry call)", page 460
- "Enquiry call with return to the initial call", page 461
- "Three-party conference from an enquiry call", page 464
- "Call transfer (switching)", page 467

8.3.5 Three-party conference from an enquiry call

In an enquiry call (with A on hold), B can set up a three-party conference with C.

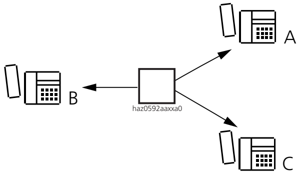


Fig. 2.176: Three-party conference

Detailed Description

Tab. 2.165: Three-party conference (conference from enquiry call)

Interface	Operating sequence / signalling on the terminal	Scope
A, C	Depending on the system configuration the conference participants will obtain[1)]: • no tone at all • the conference tone only once • the conference tone regularly	Possible interfaces: internal, external[2)], PISN

1) With three-party conference in the public exchange, the signalling depends on the network provider.

2) If both A and C are external subscribers or virtual PISN subscribers, the settings authorising exchange-to-exchange traffic will have to be observed (see "Exchange-to-Exchange Connections", page 321).

Note:
Conferences take up hardware resources.

Suffix dialling procedures

Tab. 2.166: Three-party conference (conference from enquiry call): Procedures

	System Terminals	a/b terminal
Set up three-party conference	• • Use digit suffix dialling: 3	with R 3 (R = control key)
Three-party conference in the exchange: Return to enquiry call	• With digit suffix dialling: 5	R 5 (R = control key)
Three-party conference in the exchange: Return to enquiry call with brokering	• • With digit suffix dialling: 2	R 2 (R = control key)
End three-party conference in the exchange	• hang up • Disconnect key	• hang up

System configuration

Tab. 2.167: Three-party conference: System configuration

Parameter	Parameter value	Remarks
Hold in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Three-party conference in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Conference tone / intrusion tone	On / off / only once	Throughout the system

Reference to Other Features

Features:
- "Hold (enquiry call)", page 460
- "Conference", page 465

8.3.6 Conference

Subscriber A can set up a conference call with several subscribers. There are two types of conference:

- Variable conference: The conference is set up one subscriber at a time. The subscribers are called one after the other.
- Preconfigured conference: Here the conference participants are preconfigured in the system configuration and are all called up at the same time.

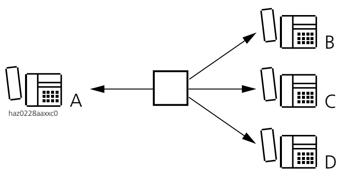


Fig. 2.177: Conference

Detailed Description

Tab. 2.168: Conference

Interface	Operating sequence / signalling on the terminal	Scope
A	The conference leader obtains a ring-back tone when setting up the conference.	
B, C, D	The preconfigured or dialled subscribers obtain ringing signalling during the conference setup and during the conference – depending on the system configuration[1]: • no tone at all • the conference tone only once • the conference tone regularly	Possible interfaces: internal, external[2] [3], PISN[4] Restriction: Each conference circuit allows: • 3 or • 6 (System 2065) subscribers[5].

1) With three-party conference in the public exchange, the signalling depends on the network provider.

2) In a preconfigured conference a maximum of one external subscriber can be switched into the circuit. If more than one external subscriber is to be switched into a variable conference, the settings authorising exchange-to-exchange traffic need to be observed (see "Exchange-to-Exchange Connections", page 321).

3) With three-party conference in the public exchange, only external interfaces are possible.

4) The settings authorising exchange-to-exchange traffic need to be observed for PISN subscribers in the public network or on a virtually connected PINX.

5) Only three subscribers are permitted if three-party conference in the public exchange is activated.

Note:
If the conference tone is deactivated in the system configuration, a subscriber who is intruded upon will not hear an attention tone. Observe the national data protection regulations. With a three-party conference in the public exchange, the signalling depends on the network provider.

On the system 2025 / 2045 a maximum of 3 three-party conferences and 1 six-party conference are possible simultaneously on the PBX.

On the system 2065 a maximum of 10 three-party conferences and 4 six-party conferences are possible simultaneously on the PBX.

Procedures

Tab. 2.169: Suffix dialling procedures

	System Terminals	a/b terminal
Set up conference from enquiry call (three-party conference, three-party conference in the public exchange)	• • Use digit suffix dialling: 3	with R3 (R = control key)
Expand variable or preconfigured conference from enquiry call:	• • Use digit suffix dialling: 3	with R3
Exclude (internal) conference participants. The external connection is maintained.	Procedure: #71	with R#71

Tab. 2.170: Conference: Prefix Dialling Procedures

	*/# procedure
Set up preconfigured conference	*70 conf. No. (1…4)
Set up variable conference	*71 SC No. *SC No. #

System configuration

Tab. 2.171: Conference: System configuration

Parameter	Parameter value	Remarks
Conferences	Member <SC No.> for group <1 to 4>	
Conference tone / intrusion tone	<On / off / only once>	

Reference to Other Features

Features:
- "Three-party conference from an enquiry call", page 464

8.3.7 Call transfer (switching)

Subscribers A and B are in a call. Subscriber B hands over the call with or without prior notice to subscriber C.

See also:
For more information on the transfer functions and the Operator Consoles, see "Operator Console", page 248.

Call transfer with prior notice

A subscriber B can transfer a call with subscriber A to subscriber C after an enquiry call. In this transfer type subscriber B waits for subscriber C to answer (he gives notice of the call) before handing over the call.

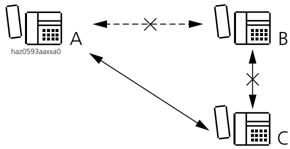


Fig. 2.178: Call transfer with prior notice

Detailed Description

Tab. 2.172: Call transfer with prior notice

Interface	Operating sequence / signalling on the terminal	Scope
A	If A is on hold, he hears "Music on hold"	Possible interfaces: internal, external[1)], PISN[2)]
B	If C hangs up during the enquiry call, B obtains the busy tone.	
C	Internal call / external call[3)]	Possible interfaces: internal, external[1)], PISN[2)]

1) If both A and C are external subscribers, the settings authorising exchange-to-exchange traffic will need to be observed (see "Exchange-to-Exchange Connections", page 321).

2) The settings authorising exchange-to-exchange traffic need to be observed for PISN subscribers in the public network or on a virtually connected PINX.

3) Depending on the PBX setting, C will obtain either an internal or an external ringing tone

If C and B hang up before the call transfer has been made, B will obtain 10 seconds of continuous ringing.

Suffix dialling procedures

Tab. 2.173: Call transfer with prior notice: Procedure

	All terminals
Call transfer	hang up

System configuration

Tab. 2.174: Call transfer with prior notice: System configuration

Parameter	Parameter value	Remarks
Hold in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Call transfer in the exchange	Yes / No	• In the group configuration. • Local feature does not require a setting
Music on hold	None / External / Internal / Hold tone	Throughout the system

Reference to Other Features

Features:
- "Hold (enquiry call)", page 460
- "Call acceptance", page 474

Call transfer without prior notice

A subscriber B can transfer a call with subscriber A to subscriber C after calling subscriber C. In this type of transfer, subscriber B does not wait for subscriber C to answer (he does not give notice of the call) before handing over the call.

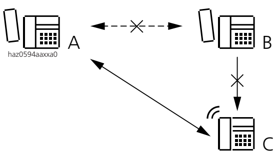


Fig. 2.179: Call transfer without prior notice

Detailed Description

Tab. 2.175: Call transfer without prior notice

Interface	Operating sequence / signalling on the terminal	Scope
A	If A is on hold, he obtains the ring-back tone or "Music on hold".	Possible interfaces: internal, external[1], PISN
B	• When B calls subscriber C, he obtains the ring-back tone (B must hear this tone before he can hand over the call) • On the operator console the line is signalled as switched until subscriber C answers the call or a recall takes place.	
C	Internal call / external call	Possible interfaces: internal, external[1], PISN

1) If both A and C are external subscribers or virtual PISN subscribers, the settings authorising exchange-to-exchange traffic will have to be observed (see "Exchange-to-Exchange Connections", page 321).

If the call is not answered by C within the configured recall time and C is an internal subscriber, the call will ring again at B (see "Recall", page 472). If the recall is not answered within 15 s, the call is rerouted to Capolinea.[1]

Suffix dialling procedures

Tab. 2.176: Call transfer without prior notice: Procedure

	All terminals
Call transfer	hang up

System configuration

Tab. 2.177: Call transfer without prior notice: System configuration

Parameter	Parameter value	Remarks
Handover without notification	Ring / hold	The value determines whether the caller obtains the ring-back tone or "Music on hold".

Reference to Other Features

Features:

- "Hold (enquiry call)", page 460
- "Recall", page 472

1) Only in Italy

Call transfer if busy

A subscriber B can hand over a call with subscriber A to the busy subscriber C after making an enquiry call to C by activating a recall and then hanging up. As soon as the busy subscriber C is free again, C's set will automatically ring. When C answers, he is connected with A.

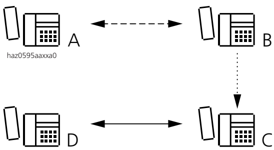


Fig. 2.180: Call transfer if busy

Detailed Description

Tab. 2.178: Call transfer if busy

Interface	Operating sequence / signalling on the terminal	Scope
A	If A is on hold, he hears "Music on hold"	Possible interfaces: internal, external[1], PISN[2]
B	• After the enquiry call to C, B obtains a busy tone. • After recall has been activated B obtains an acknowledgement tone. • On the operator console the line is signalled as switched until subscriber C answers the call or a recall takes place.	
C		Possible interfaces: internal, external[1)3)], PISN[2)3)]

1) If both A and C are external subscribers, the settings authorising exchange-to-exchange traffic will need to be observed (see "Exchange-to-Exchange Connections", page 321).

2) The settings authorising exchange-to-exchange traffic need to be observed for PISN subscribers in the public network or on a virtually connected PINX.

3) For subscribers in the public network or reached via the public network, the feature Callback if busy (CCBS) must be supported end-to-end by the public network.

If subscriber B signals call waiting to C and then goes on hook, the call with A is transferred. This applies only if C does not reject B's call. For the full scope of this feature see "Call waiting", page 477.

If the call is not answered by C within the configured recall time (C still busy or does not answer), B again obtains ringing (see "Recall", page 472).

If subscriber B intrudes on C's call and then goes on-hook, the call with A is also transferred. This applies only if C neither rejects nor answers B's call. For the full scope of this feature, see "Intrusion", page 481.

Suffix dialling procedures

Activate callback: see "Callback if subscriber busy / free", page 504.

Tab. 2.179: Call transfer if busy: Procedure

	All terminals
Call transfer if busy	Activate callback and go on-hook

System configuration

Tab. 2.180: Call transfer if busy: System configuration

Parameter	Parameter value	Remarks
Music on hold	None / External / Internal / Hold tone	Throughout the system

Reference to Other Features

Features:

- "Hold (enquiry call)", page 460
- "Callback if subscriber busy / free", page 504
- "Recall", page 472
- "Call waiting", page 477
- "Intrusion", page 481

8.3.8 Recall

Recall reminds a subscriber that a call has been transferred but not answered.

Recall is triggered if the internal subscriber does not respond within the recall time in the case of transfer without prior notice.

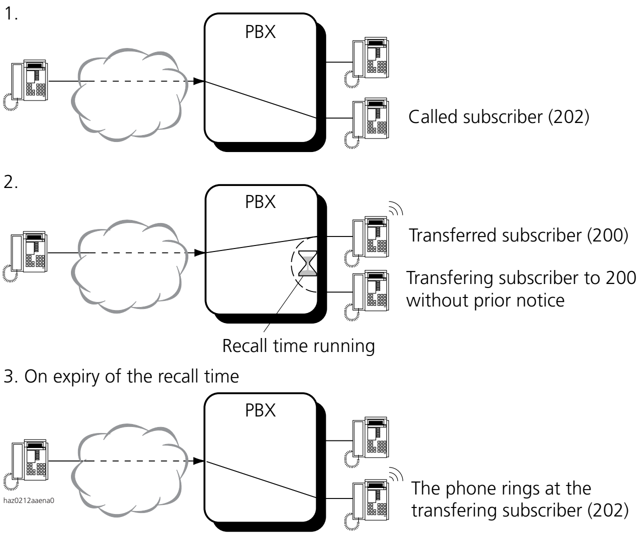


Fig. 2.181: Recall time

The recall time is defined throughout the system. A recall time can also be configuration individually for each subscriber. The recall time defined for the switched subscriber (SC 200) takes priority. A recall (to SC 202) is triggered once that time has elapsed.

In some cases the recall time used depends on the type or the configuration of the switched subscriber (SC 200):

If the transferred subscriber is

- not an individual internal subscriber but in a user group with several other subscribers, the recall time defined throughout the system is used.
- a PISN subscriber or an external subscriber the recall time defined throughout the system is used.
- a virtual subscriber and if no recall time has been defined for that subscriber, a separate recall time defined throughout the system for virtual subscribers is used.

If the transferred subscriber has

- activated "CFU" or "CFB", the recall time used is the one defined at the CFU destination.

- If "CFNR" or "Default CFU if no answer" is activated, the switched subscriber's own recall time is used.
- If the call is forwarded during the ringing phase (Call Deflection), the switched subscriber's own recall time is used.

If a paging system (PS) is integrated, the PS recall time specific to PS subscribers will apply.

A recall is also triggered if a parked call is not retrieved within the monitored parking time.

System configuration

Tab. 2.181: Recall: System configuration

Parameter	Parameter value	Remarks
Recall time normal	<10 to 240 seconds>	Throughout the system
Recall time for virtual subscriber	<10 to 999 seconds>	Throughout the system
Recall time	<10 to 999 seconds>	Subscriber setting

8.3.9 Call acceptance

An internal subscriber C can accept a connection with subscriber A after being contacted in an enquiry call by subscriber B, who was connected with A.

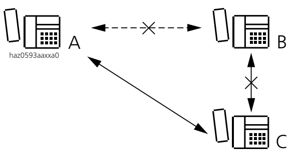


Fig. 2.182: Call acceptance

Detailed Description

Tab. 2.182: Call acceptance

Interface	Operating sequence / signalling on the terminal	Scope
B C	• As soon as C has answered the call, B obtains the busy tone	Possible interfaces: Internal Possible terminals: a/b terminal

Suffix dialling procedures

Tab. 2.183: Call acceptance: Procedure

	a/b terminal
Call acceptance	• with R1 (R = control key) or wait for more than 2 seconds after pressing the control key

System configuration

Tab. 2.184: Call acceptance: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:
- "Hold (enquiry call)", page 460

8.4 Added features

8.4.1 Name selection

Instead of entering subscriber B's phone number, subscriber A can dial subscriber B's name. The PBX supports dialling by name and dialling with Quick-dial. Please refer to the Operating Instructions of the system terminals for more details.

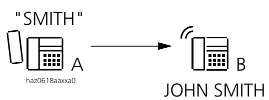


Fig. 2.183: Dialling by name

Interface	Scope
A	Requirement: The name must be stored in the caller's PBX: in the abbreviated dialling list, the phone book, the UG configuration or the subscriber configuration.
B	Possible interfaces: • Subscribers: internal, external, PISN • User group (UG)

Tip:
The name of a PISN subscriber can be configured in a PINX subscriber configuration, provided the subscriber’s number is entered in full (see "Numbering Plan", page 161).

System configuration

Tab. 2.185: Dialling by name: System configuration

Parameter
Subscriber name, abbreviated dialling name, PISN subscriber name, user group name

End-of-selection signal

The input of an external number can be completed with the character #. The PBX (or network system) interprets this as the end of selection and immediately switches through.

Detailed Description

Dialling with end-of-selection signal is important in two cases:

- when dialling an external number in an open numbering plan (Fig. 2.184).
- when the LCR (Least Cost Routing) function is activated: In this case the PBX has to wait until the subscriber has entered all the digits before it can forward the complete number to the network provider configured. The end-of-selection signal does not require any additional waiting time (Fig. 2.185).

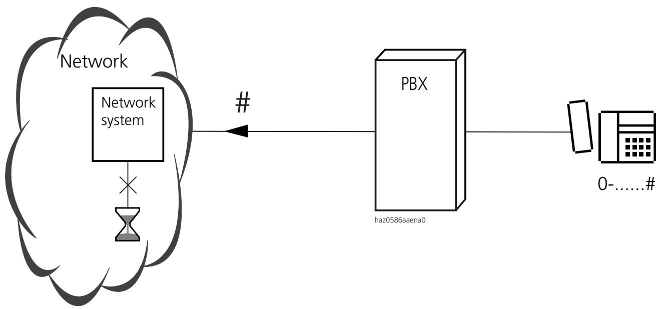


Fig. 2.184: Dialling with end-of-selection signal

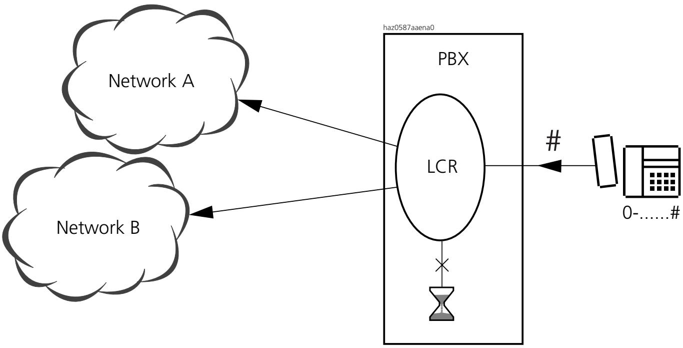


Fig. 2.185: Dialling with end-of-selection signal with the LCR function activated

Prefix Dialling Procedure

Completing dialling with end-of-selection signal: external SC No. #.

System configuration

Tab. 2.186: End-of-selection signal: System configuration

Parameter	Parameter value	Remarks
No settings		

8.4.2 Call waiting

Call waiting is used to notify an internal, busy subscriber B that another subscriber C is waiting to talk.

Subscriber B can choose to take C’s call (and put the original call on hold, end the original call or set p a three-party conference) or reject it.

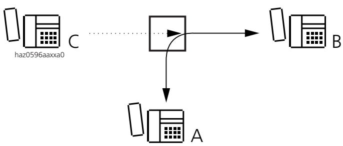


Fig. 2.186: Call waiting

Detailed Description

Tab. 2.187: Call waiting

Interface	Operating sequence / signalling on the terminal	Scope
B	B hears the dampened call waiting tone, which is played into the current call. If B has a terminal with display, the call number or name of caller C is indicated, provided his CLIP / CNIP information is available.	Requirement: • B has allowed call waiting on his set. • B is not in the process of setting up a call, in an enquiry call or in a conference.
C	• C obtains the ring-back tone by way of confirmation. • C obtains the busy tone if call waiting is not allowed or not available and if B rejects the call waiting.	Possible interfaces: • Internal[1)] Requirement: • C is authorized to use call waiting.

1) If C is an external subscriber, call waiting is effected automatically (i.e. C cannot activate call waiting), providing the subscriber receiving the call waiting has enabled the feature.

If B is in an outside call , call waiting will only work if this feature is enabled for outside calls, too (applies to the entire system).

If Courtesy Service (announcement prior to answering) is activated and subscriber B does not respond to the external call waiting, the calling subscriber C will obtain announcement prior to answering.

Tip:
If call waiting is disabled, the Attendant for example has the possibility of sending a text message to subscribers who have a system terminal with display, and to do so even during a call (e.g. "Urgent international call").

Procedures

Tab. 2.188: Call waiting: Suffix dialling procedures

	System Terminals	a/b terminal
Activate call waiting	• • Procedure: *43	R6 or R*43 (R = control key)
Answer without hold → End call and answer other subscriber	• • Use digit suffix dialling: 1	R1
Answer with hold → Hold call and answer other subscriber	• • Use digit suffix dialling: 2	R2
Answer with conference → Include other subscriber in the current call	• • Use digit suffix dialling: 3	R3
Reject → Continue with original call	• • Use digit suffix dialling: 0	R0

Tab. 2.189: Prefix Dialling Procedures

	*/# procedure	System Terminals
Protect own set against call waiting	*04	
Allow call waiting on own set	#04	

Now also possible via menu prompting.

System configuration

Tab. 2.190: Call waiting: System configuration

Parameter	Parameter value	Remarks
Call waiting	Yes	Subscriber Configuration
Call waiting / intrusion on exchange connection	Yes / No	Throughout the system

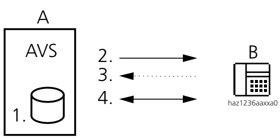
Reference to Other Features

Features:

- "Intrusion", page 481
- "Hold (enquiry call)", page 460
- "Conference", page 465

8.4.3 Ascotel Voice Mail

The Ascotel Voice Mail System (AVS) notifies all system terminals of any voice mails received. Notification depends on the type of terminal. Along with the notification on the terminal itself there is also an option to retrieve voice mails directly. For more information see the System Manual for the Voice Mail System AVS 5150.



1. Voice mail in the memory
2. Notification of voice-mail received
3. Trigger voice mail retrieval
4. Play back voice mail

Fig. 2.187: Notification by the Voice Mail System

Detailed Description

Tab. 2.191: Notification by the Voice Mail System

Interface	Operating sequence / signalling on the terminal	Scope
B	• System terminals with display: – Text message – Attention ringing tone – LED display (Office 25, Office 35, Office 45 only) • Office 20: Display: "MESSAGE". • Office 10: LED display is lit	Possible interfaces: • Internal Requirement: • System terminal • (AVS) Voice Mail System

Cancelling the notification:

- Once the notification is cancelled, the voice mails themselves remain stored in the mailbox. The notification is repeated after a while.
- The notification is cleared once the Voice Mail has been retrieved.
- If the Voice Mails were retrieved by an authorized third party (subscriber C; e.g. a handset), the notification at subscriber B is cleared.

Several users:

- If a group of users shares a mailbox, all the users will be notified. As soon as one of the users has listened to the voice mails, the notifications to all the group members will be cancelled.

Several notifications simultaneously:

If new notifications are received before the existing ones have been processed, they will be placed into a queue by order of priority as determined by the type of notification:

- Callback requests are indicated with a higher priority (ahead of notifications by the Voice Mail System).
- Text messages are indicated with a lower priority (after notifications).

Response with Call Forwarding Unconditional:

If subscriber A activates a Call Forwarding Unconditional to subscriber B, who himself has diverted to the Voice Mail user group, the response dpends on the following configuration setting of subscriber A:

- If the parameter "Last mailbox for CFU/CD" is configured to "No" (initialisation setting), a caller will be connected with the Voice Mail box of subscriber A. If subscriber A has not set up a personal Voice Mail box, the caller will obtain the spoken introduction text of the operating mode currently set. This response also applies to CFU chains.
- If the parameter "Last mailbox for CFU/CD" is configured to "Yes", a caller will be connected with the Voice Mail box of subscriber B. In CFU chains the subscriber is connected with the Voice Mail box of the last subscriber in the chain.

Prefix Dialling Procedures

Tab. 2.192: Notification by the Voice Mail System: Procedures

	System Terminals
Retrieve message Cancel the notification	 (only in the case of a notification text)

System configuration

Tab. 2.193: Notification by the Voice Mail System: System configuration

Parameter	Parameter value	Remarks
Voice Mail System	Yes	

Reference to Other Features

Features:

- "Sending and reading text messages", page 493

8.4.4 Intrusion

If the called internal subscriber B is busy, the internal caller C has the possibility of intruding into the current call. Subscriber C hears the current call and has the possibility of talking to subscriber B in to whose call C has intruded.

Subscriber B can choose to take C's call (and put the original call on hold, end the original call, set up a three-party conference) or reject it.

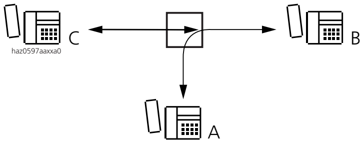


Fig. 2.188: Intrusion

Detailed Description

Tab. 2.194: Intrusion

Interface	Operating sequence / signalling on the terminal	Scope
A	If B has an analogue connection, A will hear C's intrusion.	
B	Subscriber B who is intruded upon obtains the intrusion tone to signal that in addition to the current call he also has an internal call to the intruding subscriber C	Requirement: • B has allowed intrusion on his set. • B is not in the process of setting up a call, in an enquiry call or in a conference.
C	C will obtain the busy tone if intrusion is not enabled or not available and if B rejects the intrusion	Possible interfaces: • Internal Requirement: • C is authorized to intrude.

Note:
If the conference tone is deactivated in the system configuration, subscriber B will not hear an attention tone. The national terms and conditions for data protection need to be observed in this respect.

If B is making an exchange all, intrusion will only work if this feature is also enabled for exchange calls, throughout the system.

Tip:
If intrusion is disabled, it is possible to send a text message to an intruded subscriber if he has a system terminal with display, and to do so even during a call.

Procedures

Tab. 2.195: Intrusion: Suffix dialling procedures

	System Terminals	a/b terminal
Activate intrusion	• Use digit suffix dialling: 7 • Procedure: *44	R7 or *44 (R = control key)
Answer without hold → End call and answer other subscriber	• • Use digit suffix dialling: 1	R1
Answer with hold → Hold call and answer other subscriber	• • Use digit suffix dialling: 2	R2
Answer with conference → Include other subscriber in the current call	• • Use digit suffix dialling: 3	R3
Reject → Continue with original call	• • Use digit suffix dialling: 0	R0

Tab. 2.196: Intrusion: Prefix Dialling Procedures

	*/# procedure
Protect own set against intrusion	*04
Allow intrusion on own set	#04

System configuration

Tab. 2.197: Intrusion: System configuration

Parameter	Parameter value	Remarks
Intrusion	Yes	Subscriber Configuration
Call waiting / intrusion on exchange connection	Yes / No	Throughout the system
Conference / intrusion / call waiting tone	On / off / only once	Throughout the system

Reference to Other Features

Features:

- "Call waiting", page 477
- "Hold (enquiry call)", page 460
- "Conference", page 465

8.4.5 Announcement to one or more subscribers

The announcement feature allows subscriber A to address subscriber B (or a group of subscribers) directly via the loudspeaker on B's terminal, without waiting for his answer. Subscriber B has the possibility to answer the announcement (in which case the announcement is converted into a normal, internal connection) or to interrupt it (clear down the connection).

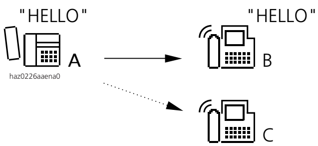


Fig. 2.189: Announcement

Detailed Description

Tab. 2.198: Announcement

Interface	Operating sequence / signalling on the terminal	Scope
A		Requirement: • A is authorized to make announcements
B	When announcement is activated, subscriber B or all the subscribers in the group will hear a warning tone over the loudspeaker	Possible interfaces: Internal only: • Subscriber • Group of subscribers Requirement: The terminal supports the announcement (Office 25, Office 35, Office 45, Office 20, Office 30, Office 40) and B has allowed announcement to his own set

Creating subscriber groups for the announcement:

- Up to 8 (System 2025 / 2045) or 16 (System 2065) announcement groups can be defined.
- Each group can consist of up to 16 subscribers.
- These subscriber groups are also used for the feature Send Text Messages (see "Sending and reading text messages", page 493).
  Cordless subscribers in the subscriber group can only be a destination for text messages and not for announcements.

Tip:
This feature can be combined with the transfer of an outside call to a paged person. If the announcement is answered, the subscriber searched for is automatically connected with the outside subscriber put on hold.

Prefix Dialling Procedures

Tab. 2.199: Announcement: Procedures

	*/# procedure	System Terminals
Set up announcement to subscriber or group	*7998 SC No. or *79 group No.	• • Office 35, Office 45, Office 30 and Office 40: double-click team key of individual subscribers
Answer announcement (called party)		answer
Answer announcement from a terminal outside the group	*89 (the other group subscribers are switched out)	
Reject announcement (called party)		or Loudspeaker key
Protect own set against announcement / Allow announcement to own set	-	

Note:
Only one announcement to one group can be active at any one time. Answering with *89 is therefore unambiguous.

System configuration

Tab. 2.200: Announcement: System configuration

Parameter	Parameter value	Remarks
Announcements	Yes	Subscriber Configuration
Message and announcement groups	<SC No.>	

Reference to Other Features

Features:

- "duplex mode", page 487

8.4.6 Charge recall

By activating a charge recall subscriber B can transfer an exchange line to an internal subscriber A. At the end of the exchange call subscriber B is called back with an indication of the call charges.

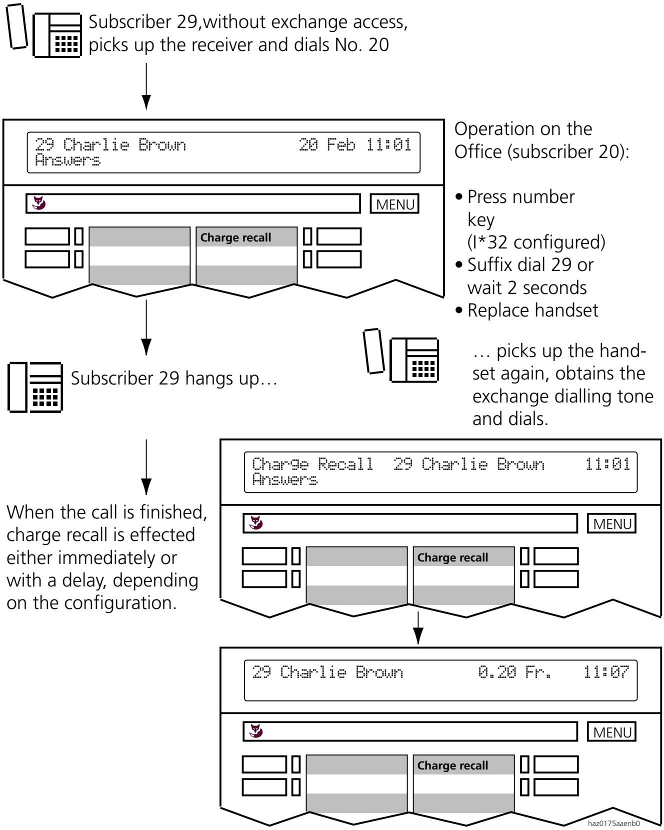


Fig. 2.190: Charge recall

Typical cases for charge recall are:

- Phone booth connection
- Exchange-barred subscribers
- Printer jam during CL output

Detailed Description

Subscriber B: Charge recall can only be activated from digital system terminals with a display.

Subscriber A: At the end of the call the subscriber's exchange access is automatically barred again.

Under Charge Recall in the AIMS Account Manager a time can be configured for both standard and phone booth connections by which a charge recall is delayed when the handset goes on-hook. This means that more than one exchange call can be made before the charge recall is effected. If the configured time is greater than zero, the internal subscriber automatically obtains the exchange free signal when he picks up the handset again and is able to dial a new number directly. If the subscriber does not pick up the handset within the time delay, a charge recall is effected.

Tip:
Store charge recall (*32 phone booth No.) under a function key.

Prefix Dialling Procedures

Tab. 2.201: Charge recall: Procedure

	*/# procedure
Activate charge recall to standard connection	*32 SC No.

System configuration

Tab. 2.202: Charge recall: System configuration

Parameter	Parameter value	Remarks
Charge recall, normal	0 to 120 seconds	
Charge recall, phone booth	0 to 120 seconds	

Reference to Other Features

Features:
- "Phone booth", page 551"

8.4.7 duplex mode

Duplex mode is a special of announcement whereby the called system terminal B immediately transform A's announcement into an internal call.

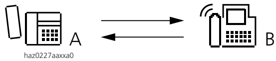


Fig. 2.191: Duplex mode

Detailed Description

Tab. 2.203: Duplex mode

Interface	Operating sequence / signalling on the terminal	Scope
A	Activates announcement	Requirement: • A is authorized to make announcements
B	The announcement is signalled by a warning tone (3 short signal tones). The call connection is then switched through (loudspeaker and microphone active).	Possible interfaces: Internal only: • Individual subscriber Requirement: • The terminal supports the automatic announcement facility (Office 35, Office 45, Office 30, Office 40) • Automatic announcement is activated on the terminal

In duplex mode the connection setup is the same as for ordinary announcements to subscribers.

Prefix Dialling Procedure

Tab. 2.204: Duplex mode: Procedure

	*/# procedure	System Terminals
Set up announcement or duplex mode (calling party)	*7998 SC No.	• • Office 35, Office 45, Office 30 and Office 40: double-click team key
Setting on the destination set		Automatic hands-free talking facility on "Announcement" or "ON"

Note:
The automatic hands-free talking setting on a terminal can be either disabled, enabled (all internal incoming calls incl. announcements are automatically seized) or enabled for announcement only.

System configuration

Tab. 2.205: Duplex mode: System configuration

Parameter	Parameter value	Remarks
Announcements Message and announcement groups	Yes <SC No.>	Subscriber Configuration

Reference to Other Features

Features:
• "Announcement to one or more subscribers", page 483

8.4.8 Making calls with your own settings on a third-party telephone

This feature allows an authorized user (internal subscriber) to use a third-party terminal with his valid password to make a single call with the following personal settings:

- Internal and external digit barring settings
- Charge counters
- CLIP display

Detailed Description

After the unlock procedure #36 the user dials his own internal subscriber number and his personal password. This activates the digit barring settings of his subscriber connection and the call charges are charged to his charge counter: The called party sees the caller's subscriber number and not the number of the terminal being used by the caller.

For reasons of data protection no entry is made in the redial register.

This same function is also used to unlock locked terminals to make a single call. For more details on this feature and how to set the password, see "Unlocking the terminal for each call", page 511.

Once the function has been activated, the user has the possibility to redial within 12 seconds without going on-hook; alternatively he can go on-hook and then dial within 60 seconds using prefix dialling.

In both cases operation is subject to the following restrictions:

- The terminal settings cannot be altered.
- The terminal's private phone book cannot be used.
- Dialling by name is not possible.

Once the call is completed, the terminal returns to its normal mode, i.e. the terminal's digit barring settings are reactivated.

Procedures

See "Unlocking the terminal for each call", page 511.

System configuration

See "Unlocking the terminal for each call", page 511.

Reference to Other Features

Features:

- "Unlocking the terminal for each call", page 511
- "Private calls with password", page 514

8.4.9 Fetch call

An incoming call from subscriber A to subscriber B can be fetched from any terminal C and then answered.

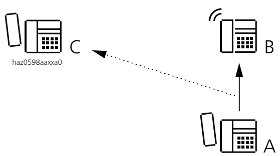


Fig. 2.192: Fetch call

Detailed Description

Tab. 2.206: Fetch call

	Operating sequence / signalling on the terminal	Scope
A-B		Incoming call to be fetched: • On a subscriber • On a user group (UG) • Excluded: Call to line key, appointment reminder call, recall
B		Possible interfaces: internal only

Tip:
Subscribers who are not at their desk can take their calls from another terminal.
Calls from persons who have not configured CFU can be fetched and answered.

Prefix Dialling Procedure

Tab. 2.207: Fetching a call: Procedure

	*/# procedure	System Terminals
Fetch call	*86 SC No. or *86 UG No. for any SC called in the UG at that particular moment.	• • Office 35, Office 45, Office 30 and Office 40: click the Team key

System configuration

Tab. 2.208: Fetching a call: System configuration

Parameter	Parameter value	Remarks
No settings		

8.4.10 Hotline

Subscriber A can be allocated one of five different hotline destinations. Whenever the handset on subscriber A's terminal is picked up, the configured hotline destination number B will automatically be dialled.

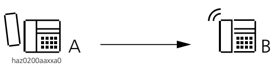


Fig. 2.193: Automatic dialling with hotline

Detailed Description

Tab. 2.209: Hotlines

Interface	Scope
B	Possible interfaces: internal, external, PISN

Other digits can be suffix dialled (for example, for a fax terminal the network access prefix is entered as the hotline destination).

The Disconnect key disconnects the call to the hotline destination and the user has the possibility of dialling another call number.

Typical applications:

- Lift telephone
- Emergency telephone
- Door phone (entrance gates)
- Phone booth connection
- Fax

Additional applications:

- Temporary hotline for hotel room and phone booth sets
- Baby alarm on hotel room phones
- Hotline to network in conference rooms
- Hotline to reception in unoccupied hotel rooms
- Hotline from rooms with sick or handicapped guests (homes, hospitals, etc.)

Prefix Dialling Procedure

Activate hotline: Pick up handset or press Loudspeaker key.

System configuration

Tab. 2.210: Hotline: System configuration

Parameter	Parameter value	Remarks
Hotline Hotline	<Variant> <Call number> for variant <1 to 5>	Subscriber Configuration

8.4.11 Sending and reading text messages

This feature provides a means of sending a text message within the system. Potential destinations include:

- An internal subscriber
- A subscriber group
- All internal subscribers

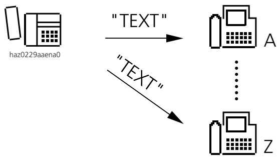


Fig. 2.194: Sending and reading text messages

Detailed Description

Tab. 2.211: Sending and reading text messages

Interface	Operating sequence / signalling on the terminal	Scope
B	When a text message is received the destination subscribers obtain an attention ringing tone.	Possible destinations: Internal only: • Individual subscriber • Subscriber group (user groups are not permitted) • All internal subscribers Requirement: Destination subscribers are equipped with a digital terminal with alphanumerical display (except for Office 20).

Subscriber groups for text messages:

- Up to 8 (System 2025 / 2045) or 16 (System 2065) groups can be defined.
- Each group can consist of up to 16 subscribers.
- These subscriber groups are also used for the feature Announcement (see "Announcement to one or more subscribers", page 483).

The text of a text message is either user-definable or can be selected from 16 texts (standard texts) predefined by the system (see "Standard texts", page 498). In addition 5 personal message texts can also be stored on the Office 45.

A message text can be up to 160 characters long.

Standard texts can be activated with or without additional text (parameters).

In principle callback requests and notifications by the Voice Mail System are displayed with a higher priority on the terminal, i.e. before any text messages.

A maximum of 16 text messages are stored for any given destination subscriber.

Tip:
A busy subscriber who has also disabled intrusion and call waiting can still be reached using text messages.

Prefix Dialling Procedures

Tab. 2.212: Sending and reading text messages: Procedures

	*/# procedure	System Terminals
Send standard text with/without parameters to the subscriber	*3598 SC No. text No. [Param] #	
Send standard text with / without parameters to group	*35 Gr. No. text No. [Param] #	
Send standard text with / without parameters to all	*3599 text No. [Param] #	
View text messages		

System configuration

Tab. 2.213: Text messages: System configuration

Parameter / action	Parameter value	Remarks
Text messages	<Message text> for message <1 to 16>	Texts can be edited
Message and announcement group	Member <SC No.> for group <1 to 16>	
Reset to initialization standard texts	<Language>	Individual standard texts cannot be reset.
Delete all pending messages		Deletes the messages on all system terminals
Delete messages more than 3 days old		Deletes the messages on all system terminals

Reference to Other Features

Features:

- "Leave message", page 496
- "Standard texts", page 498
- "Ascotel Voice Mail", page 479
- "Operation with external messaging, monitoring and alarm systems", page 556

8.4.12 Message function

A MESSAGE can be sent from any terminal to all system terminals. Depending on the terminal the receipt of a MESSAGE is signalled by a callback request.

Fig. 2.195: Activate MESSAGE

Detailed Description

Tab. 2.214: Activate MESSAGE

Interface	Operating sequence / signalling on the terminal	Scope
A	Once the procedure has been executed, A obtains the acknowledgement tone.	Requirement: The activating subscriber A must be authorized to use this function.
B	• Office terminals with display: – Text messages – Attention ringing tone – LED display (Office 25, Office 35, Office 45 only) • Office 20: "MESSAGE" appears on the display • Office 10: LED display is lit	Possible interfaces: Internal Requirement: Office terminal

Number of callback requests:
The number of callback requests that can be stored for each user-network interface depends on the terminal.

Display priority:
External alarm messages have maximum priority. Callback requests are displayed with a higher priority than Voice Mail notifications and text messages.

Tip:
With the Message function a subscriber has the possibility of activating several callbacks simultaneously, depending on his terminal.

Prefix Dialling Procedures

Tab. 2.215: Activating MESSAGE: Procedures

	*/# procedure
Activate MESSAGE	*38 SC No.
Answer MESSAGE (trigger callback)	*#38
Clear MESSAGE on one's own terminal	#38#
Clear MESSAGE on the destination set	#38 SC No.

System configuration

Tab. 2.216: MESSAGE: System configuration

Parameter	Parameter value	Remarks
MESSAGE	Yes	Subscriber Configuration

Reference to Other Features

Features:

- "Callback if subscriber busy / free", page 504
- "Wait until free", page 507

8.4.13 Leave message

If subscriber B is absent or unobtainable for longer period of time, he can leave a message in the PBX for internal subscribers. If subscriber A now calls subscriber B from a system terminal with display, the PBX will send to A's display the text left by B.

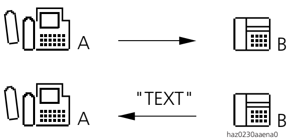


Fig. 2.196: Leave message

Detailed Description

Tab. 2.217: Leave message

Interface	Operating sequence / signalling on the terminal	Scope
A		Possible interfaces: internal only Requirement: The subscribers is equipped with a digital terminal with alphanumerical display (except for Office 20).
B	The subscriber obtains the acknowledgement tone every time he activates / deactivates the feature.	

If the conditions for subscriber A are not met (A is not an internal subscriber or does not have an alphanumerical display):
The call is routed to the number of the configured Call Forwarding Unconditional. If this number is not configured, the call is routed in the normal way to the subscriber who has left the message. The call is stored in the list of callers.

Message:

- The message is either user definable or can be selected from a choice of 16 standard texts (see "Standard texts", page 498).
- The standard texts can be configured to the customer's special requirements.
- The standard texts can be activated with or without additional parameters. Their length is limited to 160 characters.

Note:
Activating a call forwarding deletes the message.

Prefix Dialling Procedures

Tab. 2.218: Leaving a message: Procedures

	*/# procedure
Activate leave message	*24 Text. No. [Param] #
Clear leave message	#24

System configuration

Tab. 2.219: Leaving a message: System configuration

Parameter	Parameter value	Remarks
Predefined CFU	<SC No.>	

Reference to Other Features

Features:

- "Call Forwarding Unconditional (CFU)", page 438
- "Sending and reading text messages", page 493

8.4.14 Standard texts

Tab. 2.220: Message texts predefined in the system

Number	Text
1	MEETING AT >
2	PLEASE CALL BACK >
3	FOLLOWING MEETING HAS BEEN CANCELLED >
4	REQUIRED INFORMATION ON >
5	URGENT DELIVERY >
6	PLEASE DROP BY IMMEDIATELY >
7	PLEASE COLLECT MAIL >
8	MAIL WAITING >
9	I'M IN THE WAREHOUSE >
10	I'M IN THE OFFICE >
11	I'LL BE BACK ON >
12	I'M AWAY UNTIL >
13	I'M AWAY. MY REPLACEMENT IS >
14	I'M AWAY BRIEFLY >
15	PLEASE DO NOT DISTURB >
16	I CAN BE REACHED UNDER NO. >

Standard texts can be complemented or reworded before they are sent. The changes are not stored.

With AIMS the language for the standard texts can be selected independently of the language setting on the system terminals.

With AIMS the standard texts can be adapted to suit requirements but also reset to the original text as defined by the initialization values.

If the Call Centre is connected, text message No. 8 must not be reconfigured.

System configuration

Tab. 2.221: Standard texts: System configuration

Parameter / action	Parameter value	Remarks
Text message	<Message text> for message <1 to 16>	Texts can be edited
Reset to initialization standard texts	<Language>	Individual standard texts cannot be reset.

Reference to Other Features

Features:

- "Sending and reading text messages", page 493
- "Leave message", page 496

8.4.15 Park

Local call parking

A subscriber B has put his call with on hold to answer C's call waiting signal. To transfer C to a subscriber D, B must first park his call with A so that he can put C on hold and set up the enquiry call connection to D. Once he has transferred the call, B can retrieve the parked call and continue his call.

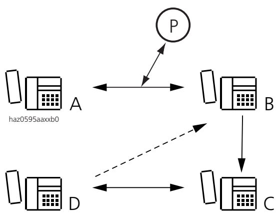


Fig. 2.197: Local call parking

Detailed Description

Tab. 2.222: Local call parking

Interface	Operating sequence / signalling on the terminal	Scope
A	Once the procedure has been executed the subscriber obtains an acknowledgement tone.	Requirement: The subscriber has a digital terminal. Restriction: A maximum of one call can be parked locally on each terminal.
B	The parked subscriber will obtain the signalling for "Music on hold".	

If the parked call is not retrieved within the preset parking time[1)], subscriber A will receive a recall.

Some terminals allow configuring a separate parking key (see "Configurable keys", page 435).

The PC Operator also allows locally parked calls to be retrieved from other subscriber terminals.

Procedures

Tab. 2.223: Local parking: Suffix dialling procedures

	System Terminals
Park call locally	

Tab. 2.224: Local parking: Prefix Dialling Procedures

	System Terminals
Retrieve call	

System configuration

Tab. 2.225: Local parking: System configuration

Parameter	Parameter value	Remarks
No settings		

1) The parking time varies from country to country

Reference to Other Features

Features:

- "Configurable keys", page 435
- "Park", page 499
- "Hold (enquiry call)", page 460

Central call parking

Subscriber A wants to continue a current call with B on a different terminal. He can park the call on the PBX’s central call parking space and then retrieve the call from terminal C.

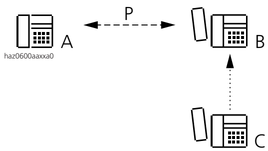


Fig. 2.198: Parking and retrieving a call centrally

Detailed Description

Tab. 2.226: Central call parking

Interface	Operating sequence / signalling on the terminal	Scope
A	Once the procedure has been executed the subscriber obtains the acknowledgement tone.	Restriction: In the PBX, only 1 call can be parked centrally throughout the system at any given time.
B	The parked subscriber will obtain the signalling for "Music on hold".	Possible interfaces: Random
C		Possible interfaces: Internal

If the parked call is not retrieved within the preset parking time[1], subscriber A will receive a recall.

[1] The parking time varies from country to country

Suffix dialling procedures

Tab. 2.227: Central call parking: Procedures

	*/# procedure
Park call centrally	*76
Retrieve call	#76

System configuration

Tab. 2.228: Central parking: System configuration

Parameter	Parameter value	Remarks
Music on hold	None / External / Internal / Hold tone	Throughout the system

Reference to Other Features

Features:

- "Local call parking", page 499
- "Hold (enquiry call)", page 460

Call parking function of the key telephone

A call signalled on a line key can be parked on the line key:

- The call is parked automatically if another call arrives on another line key and is answered.
- The call can also be explicitly parked by the subscriber.

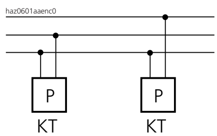


Fig. 2.199: Parking on a line key (key telephone)

Detailed Description

On a through line the call is signalled as parked on the other key telephones and can therefore also be retrieved and continued on those terminals.

Whether or not the parking time is monitored by the PBX varies from country to country.

Several calls can be parked simultaneously on different line keys.

Suffix dialling procedures

Tab. 2.229: Call parking function of the key telephone: Procedures

	Key Telephones
Park call on line key (explicit)	• Using the park key • Initiate enquiry call and hang up
Park call on line key 1 when receiving call on line key 2 (automatic)	Press line key 2 on which the other call is signalled
Retrieve call	Press line key again

Call parking function on the operator console

Attendant B is talking to subscriber A when another call from subscriber C arrives in the call queue. The active call is not to be transferred just yet and the attendant answers the incoming call. The original call is automatically parked on the corresponding line key (Office 45) or in the call queue (PC Operator).

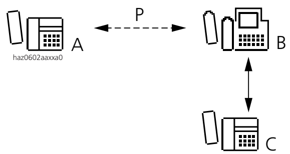


Fig. 2.200: Call parking function on the operator console

Detailed Description

Whether or not the parking time is monitored by the PBX varies from country to country.

The number of calls parked simultaneously using this call parking function is limited only by the display capabilities of the terminal in question.

On the Office 45 a call can also be parked explicitly on the line key.

Suffix dialling procedures

Tab. 2.230: Call parking function of the operator console: Procedures

	Operator console (Office 45, PC Operator)
Park call with the OC parking function	Answer other call in the call queue
Park call explicitly on the line key (Office 45)	Press hold key and then clear key
Retrieve call	Activate signalling element (Office 45: Line key) once again

8.4.16 Callback if subscriber busy / free

This feature is used to obtain an automatic callback if a subscriber is busy or if a call to a subscriber who is signalled as free goes unanswered.

Callback if subscriber busy

Subscriber A has the possibility of activating a callback to busy subscriber B (call-back request). As soon as busy subscriber B becomes free, subscriber A will be called back within 10 s. As soon as A picks up the phone, the system automatically calls subscriber B, who is now free.

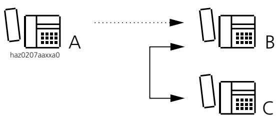


Fig. 2.201: Callback if subscriber busy

Detailed Description

Tab. 2.231: Callback if subscriber busy

Interface	Operating sequence / signalling on the terminal	Scope
A	Once the callback procedure has been executed, A obtains the acknowledgement tone.	Restriction: Subscriber A can only initiate one callback at a time.
B		Possible interfaces: internal, external[1)], PISN[2)] Restriction: Only one callback at a time can be loaded onto an external or PISN subscriber.

1) Callback to busy external subscriber is possible only if the public network supports the service "Completion of Calls to Busy Subscriber" (CCBS) end-to-end.

2) If the PISN subscriber is reached via the public network, the conditions of the public network for callback when busy will apply.

The callback is triggered only to subscriber A, who set the callback, regardless of whether a CFU or CFNR to a subscriber C has been activated at A.

Amount of time a callback if busy remains valid:

- B is internal: 45 min
- B is external: 30 min
- B is in the PISN: can vary in an heterogeneous PISN (system: 45 minutes)

Callback to an external busy subscriber:
If the B subscriber is a PBX subscriber, he must have his own direct dial number and his PBX must also support the feature. There are three possible DDI variants:
DDI number → SC B
DDI number → SC B + UG
DDI number → SC B + RA

Suffix dialling procedures

Tab. 2.232: Callback if subscriber busy: Procedures

	System Terminals	a/b terminal
Activate callback		R9 or R*37
Clear callback		#37

Note:
Completion of calls to busy is provided on the terminal even if it is not available. "not available" is then signalled after activation.

System configuration

Tab. 2.233: Callback if subscriber busy: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:
- "Callback to free subscriber", page 506
- "Wait until free", page 507
- "Message function", page 495

Callback to free subscriber

Subscriber A can activate a callback to subscriber B, if B does not answer A's call. After subscriber B has made another call (gone off-hook and then back on-hook again), subscriber A is called within 10 s. As soon as A picks up the phone, the system automatically calls subscriber B.

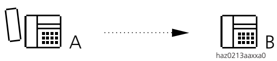


Fig. 2.202: Callback to free subscriber

Detailed Description

Tab. 2.234: Callback to free subscriber

Interface	Operating sequence / signalling on the terminal	Scope
A B	Once the callback procedure has been executed, A obtains the acknowledgement tone.	Restriction: Subscriber A can only initiate one callback at a time. Possible interfaces: Internal

The callback is triggered only to subscriber A, who set the callback, regardless of whether a CFU or CFNR to a subscriber C has been activated at A.

Amount of time a callback to free subscriber remains valid: 45 minutes.

If B has a system terminal with display, a text message with a callback prompt will appear, i. e. the callback is not automatically initiated by the PBX. In principle call-back requests are displayed with maximum priority on the terminal, i. e. before notifications by the Voice Mail System and before any text messages.

Suffix dialling procedures

Tab. 2.235: Callback to free subscriber: Procedures

	System Terminals	a/b terminal
Activate callback		R9 or R*37
Clear callback		#37

System configuration

Tab. 2.236: Callback if busy free: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:
- "Callback if subscriber busy", page 504
- "Wait until free", page 507
- "Message function", page 495

Wait until free

The "Wait until free" feature is a "Callback if busy" feature without the sub-scriber who initiates the call having to hang up. He stays on the phone and waits until the busy subscriber becomes free. The callback is triggered as soon as the called subscriber has been free for 5 seconds. The connection is then set up auto-matically.

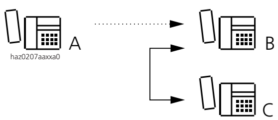


Fig. 2.203: Wait until free

Detailed Description

Tab. 2.237: Wait until free

Interface	Operating sequence / signalling on the terminal	Scope
A	• Once the callback procedure has been executed, A obtains the acknowledgement tone. • As soon as subscriber B is free, A obtains the ring-back tone.	
B		Possible interfaces: internal, external[1]

[1] Callback to busy external subscriber is possible only if the public network supports the service "Completion of Calls to Busy Subscriber" (CCBS) end-to-end.

Subscriber A must carry out the callback procedure with the handset off-hook and not via the loudspeaker key.

"Wait until free" works only with cordless terminals.

Suffix dialling procedures

Tab. 2.238: Wait until free: Procedures

	System Terminals	a/b terminal
Activate callback		R9 or R*37
Clear callback		#37

System configuration

Tab. 2.239: Wait until free: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- "Callback if subscriber busy / free", page 504
- "Message function", page 495

8.4.17 Team functions

The team functions make it easier for members of a team (for example a sales or marketing team) to communicate with one another and stand in for one another where required.

Team keys can be set up either on the system terminals themselves or via AIMS.

One team key is configured for each team member and allows the following functions and signalling states:

- Calling a team member using a simple keypress
- Signalling an incoming call for the team member and fetching the call using a simple keypress
- Signalling an existing connection to the team member
- And, depending on the terminal, other telephony functions (e.g. setting up an announcement to the team member)

Scope

Terminals that support the team key: Office 35, Office 45, Office 30, Office 40, PC Operator

System configuration

Tab. 2.240: Team function: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- "Configurable keys", page 435

8.4.18 Locking and unlocking a terminal

Terminals are locked to prevent misuse or to force the allocation of call charges on a user-pays basis.

Terminals on the system can be locked and unlocked in two different ways:

- Locking / unlocking the terminal (phone lock):
  The user can lock his terminal during his absence or restrict the dialling possibilities. He then uses a password to unlock the terminal.
- Unlocking the terminal for each call:
  Locking a terminal or restricting its dialling possibilities is activated in the system configuration.
  With his password a user can lift the restriction and make one outgoing call. The terminal is locked again automatically after the call. Permanent unlocking by the user as is the case with the phone lock is not possible.

Internal and external digit barring is used for locking / restricting dialling. This means the user is free to define what is restricted and by how much.

A terminal can be set up for one of these variants.

The password for both variants is the same.

All terminal types can be locked; on system terminals with display the function is menu-supported.

Locking / unlocking a terminal (telephone lock)

The phone lock inhibits or restricts the following operating possibilities:

- Dialling possibilities for internal and external calls, by activating internal and external digit barring.
- Operation of terminal settings.

The dialling restriction can be lifted by entering a password:

- The password is valid only for the terminal concerned
- Initialization password: "0000"
- Make sure you change the password the first time the feature is activated
- Password syntax:
  - System terminals: 2 to 10 digits (digits 0 to 9)
  - Other terminals: 4 digits exactly (digits 0 to 9)

Procedures

Tab. 2.241: Phone lock: Procedures

	*/# procedure	System Terminals
Activate phone lock with new password Activate phone lock with the password last used	*33 <Password> *33#	Office 40, Office 45 • With the code key System terminals with display: • (*33 does not work)
Deactivate phone lock	#33<Password>	(#33 does not work)
Change password	*47 <old password> * <new password> * <new password> #[1)]	System terminals with display: (*47 also works)

1) For reasons of data protection no entry is made in the redial register.

The "Change password" feature can be remote-controlled, which means it can also be used for virtual subscribers (see "Remote control features", page 542).

System Configuration

Tab. 2.242: Phone lock: System configuration in AIMS

Parameter	Parameter value	Remarks
Change terminal password	Yes / No	Setting per subscriber in the subscriber authorization
Internal digit barring settings for the unlocked state	Enable *33 and #33	Allow phone lock locking variant
Internal phone lock External phone lock	Internal digit barring 1 to 8 External digit barring 1 to 8	Definition of internal dialling possibilities in the locked state Definition of external dialling possibilities in the locked state

Tab. 2.243: Resetting the password with AIMS or Office 45

Parameter	Parameter value	Remarks
Telephone barring Password	Off 0000	Deactivate phone lock without password Resets the password

Unlocking the terminal for each call

Unlocking the terminal for each call allows the authorized user to enable any locked terminal system so that he can make a single outgoing call.
After unlock procedure #36 the user dials his own internal subscriber number and his personal password. This activates the digit barring settings of his subscriber

connection and the call charges are charged to his charge counter: The called party sees the caller's subscriber number and not the number of the terminal being used by the caller.

In this way an authorized user can use even unlocked terminals with his own settings.

For reasons of data protection no entry is made in the redial register.

See also:
"Making calls with your own settings on a third-party telephone", page 489

Unlocking a third-party terminal

An authorized user unlocks someone else's terminal. After unlocking it, he can either dial directly within the next 12 seconds or hang up and prefix dial a number within 60 seconds.

The following remain locked and inaccessible:

- Operation of terminal settings
- Use of the terminal's private phone book
- Dialling by name

Typical application: Unlocking non-personal terminals in publicly accessible premises (meeting rooms, entrance lobbies, coffee-break areas).

Tip:
Configure a key with the unlock function.

Unlocking your own terminal

An authorized user unlocks his own terminal. After unlocking it he can either dial directly within the next 12 seconds or hang up and dial a number within 60 seconds using prefix dialling or dialling by name. Both the terminal settings and the private phone book are available during those 60 seconds.

Authorized Users

For a user to be able to operate the "Terminal per call" feature, he must be known as an internal subscriber to the system and have his own personal password. He defines the password on his own terminal:

- Password syntax:
  - System terminals: 2 to 10 digits (digits 0 to 9)
  - Other terminals: 4 digits exactly (digits 0 to 9)
- Password validity
  - The password is valid for unlocking all terminals that were locked with this phone lock variant.
  - The initialization password "0000" cannot be used to unlock a terminal that was locked with this phone lock variant.

The password is stored in the system under the user's subscriber configuration.

Procedures

Tab. 2.244: Unlocking the terminal for each call: Procedures

	*/# procedure	System Terminals
Unlocking a third-party for each call	#36 <SC No.> <Password>	System terminals: • The function can be configured onto a key
Unlocking one's own terminal for each call	#36 <SC No.> <Password>	Office 40, Office 45 • With the code key System terminals with display: •

System Configuration

Tab. 2.245: Unlocking the terminal for each call: System configuration

Parameter	Parameter value	Remarks
Subscriber configuration of the terminal to be locked:		
• Phone lock:[1)]	On	Activates the lock
• Internal phone lock	Internal digit barring 1 to 8	Definition of internal dialling possibilities in the locked state
• External phone lock	External digit barring 1 to 8	Definition of external dialling possibilities in the locked state
• Internal digit barring: Input for the locked state	• #36 enable • #33 bar	• Allows unlocking for each call • Prevents permanent unlocking. Important: Without this input the lock can be lifted at any time by the user.

Parameter	Parameter value	Remarks
• External digit barring	<Barring/enabling sequences>	Restriction of external outgoing dialling options
Subscriber configuration of the unlocking user:		
• Password[1)]	<Password>	• Changes the password (must not be "0000"). • Password syntax: – Digits 0 to 9 – System terminals: 2 to 10 digit – Other terminals: 4 digits exactly

1) Setting also possible with Office 45

Reference to Other Features

Features:

- "Making calls with your own settings on a third-party telephone", page 489
- "Private calls with password", page 514

8.4.19 Private calls with password

This feature is used to charge private phone calls automatically to private charge counters, using the appropriate System Configuration. Subscribers must always enter their valid password beforehand. They can do so both on their own terminal and on a third-party terminal on the same PBX or within a PISN.

Detailed Description

The user dials #46, key in his subscriber number and enters his personal password. This deactivates his external digit barring; the terminal is also unlocked and the subscriber obtains the exchange dial tone. He can then make an external call, which is automatically charged to his private charge counter.

Note:
To prevent unauthorized persons from making private calls at other subscribers’ expense, all private phone calls must be made with a password, even when subscribers are using their own terminals. The procedure is the same for both locked and unlocked terminals.

Prefix Dialling Procedures

Tab. 2.246: Private calls with password: Procedure

	*/# procedure
Private call with password from one's own terminal or from a third-party terminal	#46 <SC No.> <Password> <external call number>

Other properties:

- During a call the procedure can also be made from an enquiry call.
- The called party sees the caller's subscriber number and not the number of the terminal being used by the caller.
- For reasons of data protection no entry is made in the redial register.
- Unlike with procedure #36 (making calls with your own settings but on a third-party phone) you cannot hang up after activating the function and then prefix dial within 60 seconds.
- The same passwords are used as for the phone lock.
- Subscribers without their own terminals can be defined as virtual subscribers, and can then also use this feature.

System Configuration requirements:

- For this feature to be used, the initialization passwords must be changed first (see "Locking / unlocking a terminal (telephone lock)", page 510 for the password syntax).
- Private exchange access must not be defined or the private exchange access prefix must be barred for all subscribers using internal digit barring.

Note:
#46 temporarily bypasses any exchange access barring and the external digit barring of the subscriber identified by means of his subscriber number and password.

Reference to Other Features

Features:

- "Making calls with your own settings on a third-party telephone", page 489
- "Unlocking the terminal for each call", page 511

8.4.20 Appointment call

Each subscriber can configure one individual appointment call and one permanent appointment call in the PBX, which are then stored in the PBX.

Fig. 2.204: Appointment call

Detailed Description

Tab. 2.247: Appointment call

Interface	Operating sequence / signalling on the terminal
A	• Once the procedure has been executed, A obtains the acknowledgement tone. • If the wake-up time is reached, the terminal will ring for 1 minute.

Individual call orders are executed only once over the next 24 hours.

The appointment call is not rerouted even if CFU, CFNR or do not disturb are activated.

Permanent call orders are executed daily (Saturdays and Sundays included). The call order is activated from the corresponding subscriber set. If a subscriber busy, the appointment call is carry out once he has completed his call.

The "Clear configurations" feature (*00) does not cancel appointment reminder calls.

Prefix Dialling Procedures

Tab. 2.248: Appointment call: Procedures

	*/# procedure
Activate individual call order	*55 hh mm (hh = hour 00...23; mm = minute 00...59)
Activate permanent call order	*56 hh mm (hh = hour 00...23; mm = minute 00...59)
Clear individual call order	#55
Clear permanent call order	#56

System configuration

Tab. 2.249: Appointment call: System configuration

Parameter	Parameter value	Remarks
No settings		

8.4.21 Taking over an active connection

Preliminaries

A subscriber D can enable subscriber C to take over an existing voice or data connection A-B.

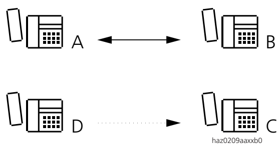


Fig. 2.205: Preparing to take over an active connection

Detailed Description

Tab. 2.250: Preparing to take over an active connection

Interface	Operating sequence / signalling on the terminal	Scope
B	Subscriber B obtains the busy tone once C has taken over the connection to A.	Possible interfaces: Internal
C		Possible interfaces: Internal
D	After preparing to take over the call or taking back the preparations for taking ver the call, D obtains the acknowledgement tone.	Requirement: Authorization has been enabled in the subscriber configuration. This authorization can be set separately for voice and data connections.

Application example

- At three football grounds reporters are reporting the matche. Depending on the state of play, the broadcast director may want to make the connection available to one of the reporters.
  The director can use the preconfigured keys on a terminal to prepare to take over the connections. All the moderator at the broadcast studio has to do is pick up the handset on his terminal (to which a hotline has been allocated with *88#) and he is immediately connected with the football ground. While he is talking, the director can prepare the connection for the next reporter, and so on.
- Takeover of an active connection in which someone is speaking onto the answering machine.

Suffix dialling procedures

Tab. 2.251: Preparing to take over an active connection: Procedures

	*/# procedure
Prepare to take over a call or data connection from SC No. nn to SC No. mm	*87nn*mm# (call) or with *84nn*mm# (data connection)
Clear the preparations for taking over a call or data connection from SC No. nn to SC No. mm	#87mm (call) or with #84mm (data connection)

System configuration

Tab. 2.252: Preparing to take over an active connection: System configuration

Parameter	Parameter value	Remarks
Voice handover	Yes	Subscriber Configuration
Data handover	Yes	Subscriber Configuration

Reference to Other Features

Features:

- "Answer", page 519

Answer

A subscriber C can take over an existing voice or data connection A-B if D has prepared the takeover.

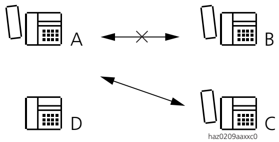


Fig. 2.206: Taking over an active connection

Detailed Description

Tab. 2.253: Taking over an active connection

Interface	Operating sequence / signalling on the terminal	Scope
B	Subscriber B obtains the busy tone once C has taken over the connection to A.	Possible interfaces: Internal

Prefix Dialling Procedure

Tab. 2.254: Taking over an active connection: Procedure

	*/# procedure
Take over call/data connection	*88#

System configuration

Tab. 2.255: Taking over an active connection: System configuration

Parameter	Parameter value	Remarks
Voice handover	Yes	Subscriber Configuration
Data handover	Yes	Subscriber Configuration

Reference to Other Features

Features:

- "Taking over an active connection", page 517

8.5 Special features

Here describes features that are available only in combination with a special application or supplementary equipment, e.g. Courtesy service or door bell.

8.5.1 Coded ringing on general bell

The installation of a general bell feature provides a paging system, albeit with a limited scope. Up to five internal subscribers can be paged using a specific coded ringing on the general bell. A subscriber who recognizes his ringing pattern can answer the call from any terminal B.

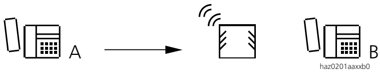


Fig. 2.207: Coded ringing on general bell

Detailed Description

Tab. 2.256: Search via coded ringing on general bell

Interface	Operating sequence / signalling on the terminal	Scope
A	• A obtains the ring-back tone • A obtains the busy tone (the display reads "unavailable") if the general bell is busy (queue full).	Possible interfaces: The function is activated locally on the system.
B		Possible interfaces: Internal

Coded ringing consists of a long tone followed by n number of shorter tones (n = 1…0.5) and is set via the system configuration.

Coded ringing can be used as the destination for a Call Forwarding Unconditional.

The PS must be deactivated.

Procedures

Tab. 2.257: Coded ringing on the general bell: Prefix Dialling Procedures

	*/# procedure
Activate coded ringing	*81 SC No.
Activate CFU to coded ringing	*28
Clear CFU to coded ringing	#28
Answer coded ringing	*82

Tab. 2.258: Coded ringing on the general bell: Suffix dialling procedure

	*/# procedure	System Terminals	a/b terminal
Activate coded ringing	*81		R8 or R*81 (R = control key)

System configuration

Tab. 2.259: Coded ringing on the general bell: System configuration

Parameter	Parameter value	Remarks
Coded ringing	<SC No.> for variant <1 to 5>	
Coded ringing	<Variant>	Subscriber Configuration

Answering general bell

A call can be signalled on the general bell (ringing signal) and be answered by any subscriber B who hears it.

Fig. 2.208: Answer ringing signal on general bell

Detailed Description

General bell is activated via user group (UG) or via proxy.

If other calls are routed to the general bell, they are placed in a queue (max. 10 entries).

Tip:
General bell in the UG of the operator console with delay:
If the attendant is absent for a short time (or is overloaded), the general bell is activated after the delay time. Employees who hear the ringing tone can then answer the call.

Prefix Dialling Procedures

Tab. 2.260: Answer general bell: Procedure

	*/# procedure
Answer ringing signal on general bell	*83

System configuration

Tab. 2.261: Answer general bell: System configuration

Parameter	Parameter value	Remarks
General Bell	Yes	User group
General Bell	Yes	Substitution

General Bell on an Analogue user-network interface a/b

Besides the connection via a potential-free relay contact on the mainboard the general bell can also be connected to an analogue user-network interface a/b. a/b connected. On each system precisely one a/b port can be configured for this purpose. Any subscriber number already assigned is then automatically deleted. Once the connection is made, no calls can be made or received via the port.

System configuration

Tab. 2.262: Analogue port for general bell: System configuration

Parameter	Parameter value	Remarks
General Bell	Yes	Interface configuration

Reference to Other Features

Features:

- "Call Forwarding Unconditional (CFU)", page 438
- "Call Forwarding on No Reply (CFNR)", page 445
- "User group: Logging in and logging out", page 533
- "Searching via a paging system", page 536

8.5.2 Courtesy Service (announcement prior to answering)

The Courtesy service is an announcement service for incoming external calls. If an outside call from A is not answered within a preset delay time by internal subscriber B (who is either free or for whom call waiting is enabled), the caller will hear a recorded message (provided the call has not been rerouted to the alternative destination (Capolinea)[1] beforehand). Once the recorded message has been played, the caller once again obtains the ring-back tone or music.

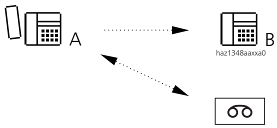


Fig. 2.209: Courtesy Service

As long as caller A is connected with Courtesy, subscriber B's terminal continues to ring. If B answers, the connection is put through immediately.

If B does not answer within 5 minutes, the connection is cleared down.

Recorded messages can be recorded and imported individually for each recorded message group and for each switching variant (e. g. Day, Night, Weekend). Recorded messages can be recorded either with a phone or using audio equipment connected to the audio input. It is also possible to record announcements with a PC, store them as a wave file and load them into the PBX.

The System Configuration determines whether the Courtesy sequence is to consist of a recorded message followed by the ring-back tone or of a recorded message followed by music (music source as per configuration of the "Music on hold" parameter). The configuration also specifies whether the sequence is to be played once or continuously.

1) Only for Italy

In the "Repeat continuously" Courtesy mode the pause during which the ring-back tone or music is played can be configured. If the pause is configured to 0 seconds, the recorded message is repeated without interruption.

Detailed Description

Tab. 2.263: Courtesy Service

Interface	Operating sequence / signalling on the terminal	Scope
A	If the internal subscriber answers during the recorded message, the message is interrupted.	Possible interfaces: External
B	The internal subscriber's set continues to ring while the recorded message is being played.	Requirement: Courtesy is not activated if B has activated a Call Forwarding Unconditional to an external destination (exchange-to-exchange connection).

Note:
To send the recorded message, a through-connection is made on the exchange side, which means that, from this moment onwards, the caller incurs call charges.

The system features:

- 1 recorded message group (basic system2025 / 2045) or
- two recorded message groups (basic system2065).

A recorded message group can send out different recorded messages depending on switch position 1, 2 and 3. Depending on the sales channel, 2 or 3 different recorded messages are available for each group:

- With 2 recorded messages switch positions 2 and 3 provide the same text. Twice 27 seconds are available:
- With 3 recorded messages, three times 16 seconds are available.

There are 4 parallel Courtesy channels:

- If another call occurs during a recorded message, the second call is switched to Courtesy via a second channel once the delay has expired.
- If all 4 channels of a group are busy, the 5th caller is put on a hold position. He will obtain the ring-back tone until a channel once again becomes free or until he can be synchronised with the start of a busy channel.

- In the "Repeat continuously" mode, callers on several Courtesy channels are synchronized on the same channel with the same recorded message . This frees us channels for new callers. This requires that the pause during which the ring-back tone or music is played is not 0 and that the pauses overlap.

Calls to all network interfaces of a trunk group can be routed to the recorded message. The configuration is made individually for each trunk group and DDI number, in the call distribution elements.

The Courtesy service is also available if:

- the external call's destination is a PISN subscriber in a QSIG network who has activated the announcement service locally in his node.
- an internal subscriber has diverted to a PISN subscriber in a QSIG network who has activated the Courtesy service locally in his node.

The call routing, delay setting and modes of the Courtesy recorded messages can only be carried out by the Installer in the system configuration.

The recorded messages are recorded or operated using normal */# procedures; this can be done from any terminal authorized accordingly (as per the digit barring table). A PISN subscriber can only operate the control funtions of his own local PBX using the */#-procedures.

Recording messages

With a telephone or audio equipment

Recorded messages can be recorded either directly with a phone or using audio equipment connected to the audio input.

Tab. 2.264: .Recording procedures

	*/# procedure
Recording with the phone	*91xy
Check the recording	*#91xy
Delete the recording	#91xy
Recording with audio equipment	*92xy

"x" stands for message group 1 or 2.
"y" stands for switch position 1, 2 or 3[1)].

Recording with the phone:
After the procedure a long Start tone is audible and the recorded message can be recorded over the handset. To end the recording, put the handset on-hook. The recorded message is then stored automatically.

Recording with audio equipment:
After the procedure a long Start tone is audible and the recorded message can be played back over the audio equipment connected to the music input of the PBX. The recording can be monitored via the handset. To end the recording, put the handset on-hook. The recorded message is then stored automatically.

With the PC

Announcements can also be recorded with a PC through a connected microphone (e. g. with the Windows Audio Recorder). The recordings have to be stored as wave files in a particular format under a predefined name.

- Format: CCITT A-Law, 8 kHz, 8 bit, mono
- File names: court_xy.wav (Lower-case letters required)

"x" stands for message group 1 or 2.
"y" stands for switch position 1, 2 or 3[1)].

The wave files with the announcement texts must now be loaded onto the PBX using AIMS:

1. From the AIMS Shell use "Tools – Partial upload – Courtesy files" to select one or several wave files
2. Use the Upload window to add or delete other Courtesy files
3. Load the Courtesy files onto the PBX using "Upload"

Once the upload is completed, the texts still have to be activated. This transfers the texts from the Flash memory to the DSP and overwrites the old texts. There are three ways of activating the texts:

- In the AIMS Shell
  Once the Courtesy files have been uploaded, answer "Yes" to the "Activate Courtesy Texts" prompt.

1) The number of different recorded messages for each announcement group depends on the sales channel.

- In the Configuration Manager
  In the Courtesy settings click the "Activate courtesy files" button.
- With the procedure *930
  The start of the transfer is confirmed by a check tone.

After activation we recommend that you check the texts by monitoring them using the corresponding */# procedure (see Tab. 2.264). However this can only be done once the texts have been fully loaded, which takes a little more time than the sum total of the length of all the texts to be activated.

Wave files that are longer than the time available for the recorded messages (see page 524) are cut off.

Wave files with incorrect file names cannot be loaded. AIMS generates a corresponding error message.

Wave files with incorrect format cannot be activated. In such cases the texts already activated are simply retained. No error message is generated.

Prefix Dialling Procedures

Authorised subscribers can activate or deactivate the Courtesy feature for each group.

Tab. 2.265: Courtesy Service: Procedures

Courtesy group	*/# procedure
Activate	• *931 (Group 1) • *932 (Group 2)
Deactivate	• #931 (Group 1) • #932 (Group 2)

System configuration

Tab. 2.266: Courtesy Service: System configuration

Parameter	Parameter value	Remarks
Courtesy group	Group <1 and 2>	only basic system 2065 has 2 groups
Courtesy delay	Delay <0 to 300 seconds>	

Parameter	Parameter value	Remarks
Mode	Ring-back tone / music	Either ring-back tone or music is played after the announcement. (music source in accordance with the configuration of the parameter "Music on hold")
Repeat continuously	Yes / No	
Pause	Pause <0 to 30 seconds>	Effective only if the parameter "Repeat Continuously" is configured to "Yes".

All the parameters can be configured for each group.

Note:
Operation of the Courtesy functions is usually disabled in the internal digit barring (initialization value).

8.5.3 Clear configurations

With this procedure each subscriber has the possibility to clear all the features he has activated with the exception of night service, user group and appointment orders.

Detailed Description

Tab. 2.267: Clear settings

Interface	Operating sequence / signalling on the terminal
A	Once the procedure has been executed, the subscriber obtains an acknowledgement tone.

This applies to the following features:

- Do not disturb
- Follow me
- Call Forwarding Unconditional
- Call Forwarding on No Reply
- Call back
- Protect against Call Forwarding on No Reply
- Protect against intrusion
- Protect against announcement
- Protect against call waiting

Prefix Dialling Procedure

Tab. 2.268: Clearing settings: Procedure

	*/# procedure
Clear settings	*00 or #00

System configuration

Tab. 2.269: Clearing settings: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- See list above

8.5.4 LCR Function

If the LCR function is activated, dialled call numbers are analysed and converted. This means that the PBX may actually dial a different call number than the one entered by the user (see "LCR Function", page 306).

Users can be authorized through the subscriber configuration to dial using network providers of their own choice, contrary to the set LCR criteria (see "Bypassing LCR manually (Forced Routing)", page 317).

If a network provider cannot be reached and the PBX detects this, it will automatically try and reach an alternative network provider (provided that function is activated). If a network provider cannot be reached and the PBX does not detect this, the user has the possibility of dialling the alternative provider manually using *90 (see "Alternative Routing (Fallback Routing)", page 313).

8.5.5 Emergency number

The system is equipped with what is known as an emergency number, which can be used by all internal subscribers. Emergency calls are routed to a destination B preconfigured in the system configuration.

Detailed Description

Tab. 2.270: Emergency number

Interface	Scope
B	Possible interfaces: internal, external, PISN

The emergency number destination depends on the status of switching group 1.

Note:
If an external destination with exchange access prefix code is specified, it is important to ensure that a route is assigned to each subscriber.

System configuration

Tab. 2.271: Emergency number: System configuration

Parameter	Parameter value	Remarks
Emergency number	<Call number>	Numbering Plan
Emergency number	Destination <call No.> for switch position <1 to 3>	

Note:
The emergency number can also be the destination of a hotline and can be configured differently for each of the three possible switch positions.

Example: Hotline on lift telephone

Switch position 1 (day): 11, switch position 2 (night): 175 and switch position 3 (weekend): 0118.

8.5.6 Suppression of the call number display for each call

Besides the possibility of using the System Configuration to permanently suppress the display of a particular subscriber's number on the called party's terminal (CLIR), the function can also be activated for one call only.

Detailed Description

The procedure *31 prior to dialling an external call number ensures that the CLIP is not displayed to the called party. CLIR per call is supported only for external calls via digital network interfaces with the DSS1 protocol. CLIR per call is automatically deactivated again once the call is completed.

The feature is not supported in the following cases:

- Internal calls and calls to PISN or virtual PISN subscribers
- External calls via analogue network interfaces
- In combination with an abbreviated dialling that contains other */# procedures
- In combination with dialling using a line key

The outgoing call is rejected and the subscriber obtains the busy tone.

Notes:
The function is not executed when used in connection with ISDN supplementary services in the exchange such as ECT, PARE or CD, i.e. the call number is displayed to the called party.
Depending on the network provider and service provider it may be necessary to subscribe to CLIR.

Prefix Dialling Procedure

Tab. 2.272: Activate CLIR for one call: Procedure

	*/# procedure
Activate CLIR for one call	*31 external destination No.

System configuration

Tab. 2.273: Activate CLIR for one call: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- "Suppressing CLIP / COLP (CLIR / COLR)", page 188
- "Displaying Numbers (CLIP) and Names (CNIP)", page 179

8.5.7 Recording malicious calls (MCID)

By activating the Malicious Call Identification service, MCID for short, a subscriber B can have the threatening or nuisance calls from an external subscriber A recorded by the network provider so that the caller can be identified. The recording can be activated either during the call or after the call during the busy tone signalling (once the caller has rung off).

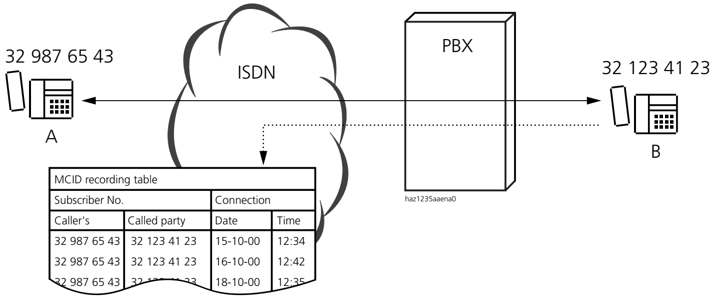


Fig. 2.210: MCID during the call

Detailed Description

This function provided by the network provider as a supplementary service is used for identifying malicious or nuisance callers. The identification is effected by the network provider. The feature is activated by the subscriber.

Suppressing the outgoing number (CLIR) does not protect the caller from identification of his subscriber number by the network provider.

The following data is recorded by the network provider:

- Calling party’s phone number
- Called party's phone number
- Date and time of the call

Tab. 2.274: Recording of malicious calls

Interface	Operating sequence / signalling on the terminal	Scope
B	Activate during the call / after the call during the busy tone signalling[1]. Network provider confirms activation (the type of signalling is specific to the network provider)	internal subscriber Connection restrictions: • Only for external incoming connections • In hands-free mode, activation is practically possible only during the call as system terminals hang up automatically within a few seconds of the end of the call.
A		External subscriber

1) The duration of the busy tone signalling after the call depends on the network provider.

Tab. 2.275: Recording of malicious calls: Requirements

Requirements	PBX
Technical	The PBX must be directly connected with the ISDN network (no support in the private network) Terminals: • System terminals (configurable only with AIMS on the Office 10) • ISDN terminals
Administrative	Must be applied for as a supplementary service from the network provider
Legal	A court injunction must be required depending on the legislation in the region concerned

Suffix dialling procedures

Tab. 2.276: Recording of malicious calls: Suffix dialling procedures

	System Terminals	ISDN terminal
Activate MCID	MCID is available as */# procedure F16#1# in the function selection list, and can be configured onto a function key	Menu or function key

System configuration

Tab. 2.277: Recording of malicious calls: System configuration

Parameter	Parameter value	Remarks
No settings		

Reference to Other Features

Features:

- "Identification Elements", page 178

8.5.8 User group: Logging in and logging out

Members of user groups can log themselves out and back in again. The logout and login procedure can apply simultaneously for all the user groups or specifically for one user group only.

Detailed Description

Tab. 2.278: User group

Interface	Operating sequence / signalling on the terminal	Scope
A	• When logging in or out, A obtains an acknowledgement tone in each case. • If the function is configured on a key with LED display, the status logged out/in will be displayed.	Requirement: • A is member of one or several user groups Restriction: • The last remaining member of a group cannot log himself out. • Does not apply to Operator and general bell

If a member activates a Call Forwarding Unconditional to an external destination or to a destination in another PINX, he will automatically be logged out. With user groups configured as "large" this applies to all types of redirected calls, even internal ones.

Prefix Dialling Procedure

Tab. 2.279: User group: Procedures

	*/# procedure
Log in to all user groups	*4800
Log out of all user groups	#4800
Log into one user group	*48 <UG No.>
Log out of one user group	#48 <UG No.>

System configuration

Tab. 2.280: User group: System configuration

Parameter	Parameter value	Remarks
User group	Members <SC No.> for group <UG No.>	
User group	<Call numbers> for group <UG No.>	Numbering Plan

Reference to Other Features

Features:

- "Coded ringing on general bell", page 520
- "Call Forwarding Unconditional (CFU)", page 438

8.5.9 Switching switch groups

Switch groups defined in the system configuration can be selected by subscriber A using switch contacts or a procedure from the terminal. The switchover can also be carried out automatically using time-controlled functions in the System Configuration (see "Time-controlled functions", page 547)

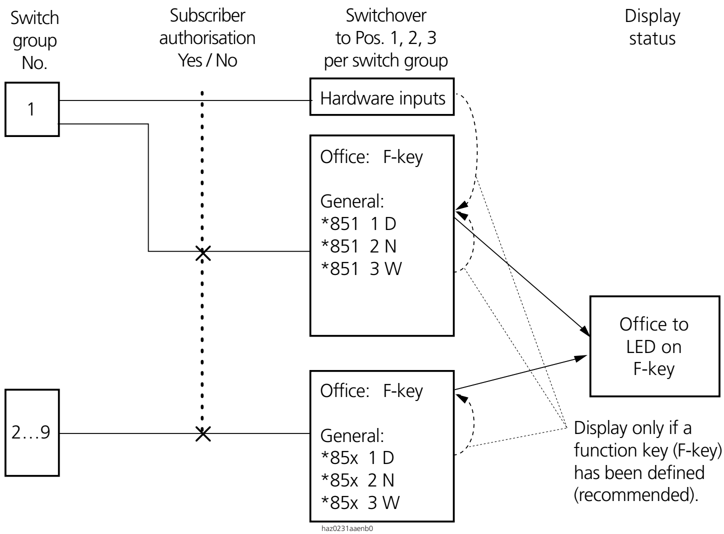


Fig. 2.211: Switching switch groups

Detailed Description

Tab. 2.281: Switching switch groups

Interface	Operating sequence / signalling on the terminal	Scope
A	• The subscriber obtains an acknowledgement tone when switching On / Off. • Terminals connected to the S-bus cannot display the status of switching groups 2 to 9. • System terminals: The switching status is displayed by the status of the LED or the corresponding symbol on the display for the function key configured accordingly.	Possible interfaces: The switch groups are operated locally on the system.

Tip:
The significance of the switching states can be seen from the designation labels created individually for each subscriber.

External switches:
Switching group 1 can also be activated via 2 control inputs on the PBX exchange, e.g. via preconfigured time-switch clock.

External switches have a higher priority, i.e. they must be open (status 0) so that switching via function key or procedure can be carried out.

Prefix Dialling Procedures

Tab. 2.282: Switching switch groups: Procedure

	*/# procedure
Switch switching group x in position y	*85xy (x = 1...9, y = 1..3)

System configuration

Tab. 2.283: Switch groups: System configuration

Parameter	Parameter value	Remarks
Operate switch group	yes	Subscriber Configuration
Switch group	Switch position of switch group 1 to 9	Display

Reference to Other Features

Features:
- "Emergency number", page 529
- "Door bell", page 539
- "Courtesy Service (announcement prior to answering)", page 523

8.5.10 Searching via a paging system

If the PBX is equipped with a paging system (PS), it can be used to search for internal subscribers. The paged subscriber (subscriber B) is reached via his pager and can answer from any terminal.

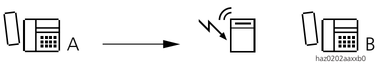


Fig. 2.212: Searching via a paging system

Detailed Description

Tab. 2.284: Searching via a paging system

Interface	Operating sequence / signalling on the terminal	Scope
A	• A obtains the ring-back tone of the paging system. • A obtains the busy tone if the paging system is busy or not available.	Possible interfaces: • The function is activated locally on the system.
B		Possible interfaces: • Internal

The paging system can also serve as the destination for a Call Forwarding Unconditional.

Procedures

Tab. 2.285: Searching via a paging system: Prefix Dialling Procedures

	*/# procedure
Activate search via PS	*81 destination No.
Activate CFU to PS	*28
Clear CFU to PS	#28
Answer PSA	*82 destination No.

Tab. 2.286: Searching via a paging system: Suffix dialling procedure

	*/# procedure	System Terminals	a/b terminal
Activate search via PS	*81		R8 or R*81 (R = control key)

System configuration

Tab. 2.287: Searching via a paging system: System configuration

Parameter	Parameter value	Remarks
Paging system	Yes	
Search duration	<10 to 360 seconds>	
Pager subscriber No.	<SC No.>	Subscriber Configuration
Pager with text display	Yes / No	Subscriber Configuration

8.5.11 Controlling relays

Various equipment or installations can be controlled using relays either on the system or on special card OI-2DOOR. The telephone can be used to operate sun blinds, for example, or to switch the lighting on or off throughout the building.

Detailed Description

Tab. 2.288: Controlling relays

Interface	Operating sequence / signalling on the terminal	Scope
A	When switching on or off, the subscriber obtains in each case an acknowledgement tone.	Possible interfaces: The function is activated locally on the system. Requirement: Authorization has been enabled in the subscriber configuration.

Prefix Dialling Procedures

Tab. 2.289: Controlling relays: Procedures

	*/# procedure
Switch relays on	*755 to *757
Switch relays off	#755 to #757

Tip:
Store the procedure under a function key.

System configuration

Tab. 2.290: Controlling relays: System configuration

Parameter	Parameter value	Remarks
Operate system relays	Yes	Subscriber Configuration

Reference to Other Features

Features:

- "Opening doors", page 540

8.5.12 Door function

The door intercom systems provide the following functions:

- Door bell input
- Opening doors
- Dial door intercom

Door bell

If a door bell or pushbutton with a similar function is connected to the system, its signal can be assigned to any internal destination B, depending on the system configuration.

Detailed Description

Tab. 2.291: Door bell

Interface	Operating sequence / signalling on the terminal	Scope
B	• When the door bell is activated the allocated destination will ring with a special ringing tone. The ringing time is limited to 20 seconds. • If B is busy, he will obtain call waiting except if he himself is already in an enquiry call. "Call waiting on exchange connection" and "Protect own set against call waiting" are not taken into account.	Possible interfaces: Subscribers: internal, PISN, UG Restriction: • If subscriber B has diverted to an external destination, the connection to the door intercom will be connected through. • The connection created with the door intercom is limited to 5 minutes (forced disconnect) if the call partner (PISN or external) is connected to the public network.

Door bell input:

- An internal subscriber can be allocated to the door bell input in each case for Day, Night and Weekend.
- The dialled destination depends on the status of switching group 1.

Prefix Dialling Procedure

Call subscriber: via the door bell.

System configuration

Tab. 2.292: Door bell: System configuration

Parameter	Parameter value	Remarks
Door intercom system	Destinations <SC No.> for switch position <1,2,3>	

Reference to Other Features

Features:

- "Opening doors", page 540
- "Dialling the door intercom", page 541

Opening doors

This function actuates the relays for door openers for 3 seconds.

Detailed Description

Tab. 2.293: Opening doors

Interface	Operating sequence / signalling on the terminal	Scope
A	Once the feature has been activated the subscriber obtains the acknowledgement tone.	Possible interfaces: The function is activated locally on the system. Requirement: Authorization has been enabled in the subscriber configuration.

Prefix Dialling Procedures

Tab. 2.294: Opening doors: Procedure

	*/# procedure
Open door (the relays are activated for 3 seconds):	*751 or *752

Tip:
Store the procedure under a function key.

System configuration

Tab. 2.295: Open door: System configuration

Parameter	Parameter value	Remarks
Opening doors	Yes	Subscriber Configuration
Door intercom system	<Call number>	Numbering Plan

Reference to Other Features

Features:

- "Door bell", page 539
- "Dialling the door intercom", page 541

Dialling the door intercom

A door intercom connected to special card OI-2DOOR can be dialled by subscriber A in the same way as an internal subscriber.

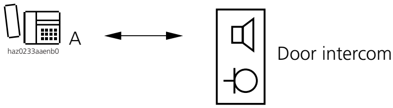


Fig. 2.213: Connection to the door intercom

Detailed Description

Tab. 2.296: Dial door intercom

Interface	Scope
A	The door intercom can be dialled up: • Locally on the system • From another PINX[1)] dialling. Requirement: Authorization has been enabled in the subscriber configuration.

1) The door intercom can be entered in the PINX numbering plan as a PISN subscriber (see "Numbering Plan", page 161).

Prefix Dialling Procedures

Dial the door intercom: Dial the door intercom number. (After initialization: 851, 852)

System configuration

Tab. 2.297: Door intercom: System configuration

Parameter	Parameter value	Remarks
Door intercom system	<Call number>	Numbering Plan

Reference to Other Features

Features:

- "Door bell", page 539
- "Opening doors", page 540

8.6 Remote control features

A large number of features can be remote controlled either from within or outside the system:

- Remote controlling features from within the system:
  Subscriber A activates/deactivates a feature on subscriber B (Tab. 2.298)

- Remote controlling features from outside the system:
  An external subscriber A activates/deactivates a system-related feature (Tab. 2.299) or a subscriber-related feature on internal subscriber B (Tab. 2.298).

Note:
The total number of digits dialled for each remote-controlled feature (for external remote control as of *06) must not exceed 32 (with the remote control of *47 and long passwords for example, this can be critical).

Tab. 2.298: Subscriber-related features remote-controlled from within and from outside the system

Feature	Activate	Reset
Clear configurations	*00	
Protect against / allow CFU/CFNR on own set	*02	#02
Protect against / allow call waiting/intrusion on own set	*04	#04
Activate / clear CFU	*21 destination No.	#21
CFU Unconditional on last configured Activate / clear SC	*21#	#21
Activate / clear CFB	*67 destination No.	#67
CFB on last configured Activate / clear SC	*67#	#67
Activate / clear CFU to preconfigured SC	*22	#22
Activate / clear CFU to standard text or activate / clear leave message	*24 text No. param.	#24
Activate / clear CFU to PS / general bell with coded ringing	*28	#28
Send text messages (standard texts) to SC	*3598 SC No. text No.	
Send text messages to group	*35 Gr. No. text No.	
Send text messages to all	*3599 text No.	
Activate / clear Message function	*38 SC No.	#38 SC No.
Change password (x: old password, y: new password)	*47 x * y * y #	
Activate / clear do not disturb	*26	#26
Log into / out of all UG	*4800	#4800
Log into / out of one UG	*48 UG No.	#48 UG No.
Activate / clear individual order for appointment call	*55 hh mm	#55
Activate / clear permanent order for appointment call	*56 hh mm	#56
Activate / clear CFNR	*61 destination No.	#61

Feature	Activate	Reset
CFNR on last configured Activate / clear SC	*61#	#61
Activate / clear CFNR to preconfigured SC	*62	#62
Activate / clear CFNR to PS / general bell with coded ringing	*68	#68

Tab. 2.299: Features remote controlled from within and from outside the system

Feature	Activate	Reset
Operate switching groups	*85 <Swith group><Pos.>	
Actuate door opener	*751, *752	
Switch relays	*755 - 757	#755 - 757

Note:
When remote controlling the features CFU and CFNR (*21, *61 and *67) the destination number has to be defined in the internal numbering plan. External destinations can therefore only be reached using abbreviated dialling.

8.6.1 Remote controlling features from within the system

A subscriber A can use procedure *06 to carry out features from his terminal on behalf of another authorized subscriber B.

Example:
An internal subscriber activates Call Forwarding on No Reply using the following procedure:

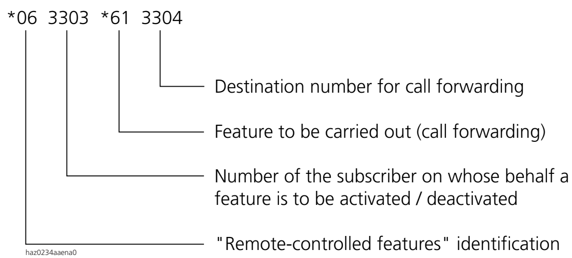


Fig. 2.214: Example of remote control

Detailed Description

Tab. 2.300: Remote controlled features

Interface	Operating sequence / signalling on the terminal	Scope
A	When activating and deactivating the feature the subscriber carrying out the feature obtains the acknowledgement tone.	Possible interfaces: • A and B are on the same system Requirements: • For subscriber A, *06 is not barred in the internal digit barring.
B		Requirement: • Authorization for remote control access.

System configuration

Tab. 2.301: Internal remote control: System configuration

Parameter	Parameter value	Remarks
Remote control permitted (subscriber B)	Yes / No	Subscriber-specific
Digit barring subscriber A	*06 enabled	Subscriber-specific

Reference to Other Features

Features:

- "Remote controlling features from outside the system", page 544

8.6.2 Remote controlling features from outside the system

An external subscriber A use a DDI number specially set up for remote control and a password valid throughout the system to remote control a group of features via the public ISDN network (External Remote Control or ERC). The features concerned can be either subscriber-related features of a subscriber B (Tab. 2.298) or system-related features (Tab. 2.299).

Detailed Description

Tab. 2.302: Externally remote-controlled features and system functions

Interface	Operating sequence / signalling on the terminal	Scope
A	• A dials the call number for remote control • After 5 seconds of ringing the connection is established and A obtains the internal dial tone. The PBX automatically switches to DTMF mode. • A enters his password followed by "#". • A again obtains the internal dial tone • A dials *06 ... (as for internal remote control).	Possible interfaces: • Public ISDN network Requirements: • DTMF-compatible telephone. • Valid password
B		Requirement: • Authorization for remote control access

Procedures

Tab. 2.303: External remote control: Procedure

	*/# procedure
Change */# procedure using external remote control	<DDI No.>< Password> #*06 <SC No.> <*/# procedure>

Example:
An external subscriber activates Call Forwarding on No Reply using the following procedure:

:

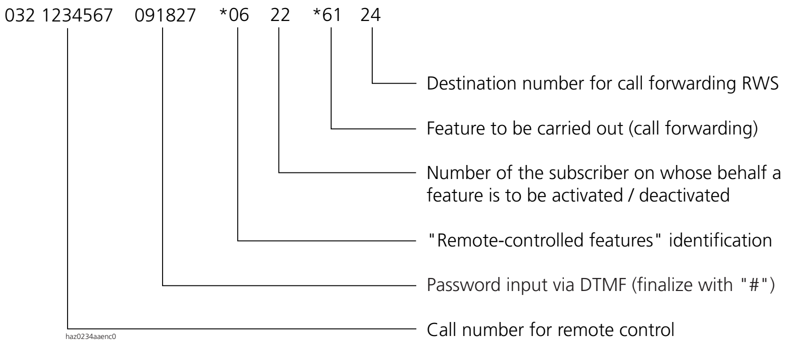


Fig. 2.215: Sequences for external remote control of features

Security elements:

- Password protection (change the password at regular intervals using AIMS or the System Assistant of the Office 45).
- External remote control barring throughout the system.
- Subscriber B authorized to use remote control access.
- Reduction in the choice of remote-controlled features using the special digit barring facilities of the external remote control.
- Time limit when entering the remote-control sequence:
  The */# procedures have to be entered within 12 seconds; thereafter the connection is cleared down automatically.
- Access logging:
  Remote control attempts from the outside are logged in the incoming call logging (ICL). If the attempt is successful, the DTMF sequence is entered in the field "Dest. No.1". Unsuccessful attempts are entered with the sort character xx9 (only in PC5 format; see "Structure of the PC5 output format", page 385).

System configuration

Tab. 2.304: External remote control: System configuration

Parameter	Parameter value	Remarks
Password External remote control (ERC)	6 digits exactly with digits (0 to 9) Enabled	Adjustable throughout the system at the attendant authorization (also with the System Assistant of the Office 45)
Internal digit barring	Selects the internal digit barring for switch position 1 to 3 of switch group 1	Initialization values: • Digit barring settings: 8/8/8 (2025/2045) 16/16/16 (/2065) • Digit barring 8/16: *85, *75, #75 barred
Remote control	Enabled	Subscriber-specific (enable remote control of the subscriber's terminal)
DDI number	<ERC DDI number>	Create DDI number for ERC and link with CDE
CDE destination	ERC	Can only be entered once a valid password has been entered

Reference to Other Features

Features:
- "Remote controlling features from within the system", page 543

8.6.3 Time-controlled functions

Up to 50 time-controlled functions (*/# procedures) can be defined in the System Configuration to be executed once at a particular time on a particular date. It is also possible to define recurring functions to be executed at a particular time on a particular weekday or every weekday. The */# procedures can be used for user-specific features or for settings applicable throughout the system.

Unlike the control of features or the modification of configurations via the terminal, time-controlled functions are not subject to the authorisations or digit barring that apply to individual subscribers.

Tab. 2.305: Examples of time-controlled functions:

No.	*/# procedure	Start (Day)	Stop (Day)	Execution date	Execution time	Switch group	Meaning
1	*0620#21	Monday	Friday		08:00		Deactivate CFU from SC 20
2	*0620*2124	Monday	Friday		16:30		CFU from SC 20 to SC 24
3	#756			20.12.2002	22:00		Relay 2 OFF (e.g. heating)
4	*756			06.01.2003	05:30		Relay 2 ON (e.g. heating)

Tip:
The function is not carried out if neither an execution date nor a start / stop range is defined. This allows you to deactivate entries in the table without having to delete them.

Features and settings can also be activated, deactivated and modified with time control and in parallel via terminals. Each particular status is event-controlled, i.e. the last command chronologically determines the current status. The previous statuses of the functions are not verified. If a function is removed from the table, its status is also retained.

Note:
Invalid entries in the function column which cannot be executed do not trigger an error message.

Switch Group Assignment

Each function can be assigned one of the switch groups 1-9. This allows you, for example during holiday periods, to activate or deactivate whole groups of functions. All functions with allocation of the corresponding switch group are active on switch postion 1 and inactive on switch positions 2 + 3.

Tab. 2.306: Activating/deactivating functions via switch groups:

No.	*/# procedure	Start (Day)	Stop (Day)	Execution date	Execution time	Switch group	Meaning
5	*931	Monday	Friday		07:00	7	Activate Courtesy
6	#931	Monday	Friday		18:00	7	Deactivate Courtesy
7	*8572			20.12.2002	18:30		Switching from switch group 7 to switch position 2: All functions allocated switch group 7 are deactivated.
8	*8571			06.01.2003	06:30		Switching from switch group 7 to switch position 1: All functions allocated switch group 7 are activated.

Note:
When switch groups are switched over, the functions are maintained in their status at the time.

Available Functions

All remote-controlled functions can be activated with time control. User-specific features are activated with *06 <SC No.>. For an overview of the functions available see Tab. 2.298 and Tab. 2.299. In addition to the remote-controlled features the following functions can also be activated using time control:

Tab. 2.307: Additional time-controlled functions

Feature	Activate	Reset
Activate / clear Courtesy	*93<group>.	#93<group>
Enable/bar a one-off remote access	*754,	#754
Enable/bar a permanent remote access	*753	#753

8.7 Hotel function

The hotel functions consist of features that are specially designed for hotel applications.
They include:

- Check-in / check-out menu
- Barring of room-to-room traffic with the possibility of bypassing using a secret code
- The input of wake-up times
- Automatic exchange barring at check-out

Analogue terminals with integrated function keys or the digital terminals with message display are suitable for use in hotel rooms (see also "Message function", page 495).

8.7.1 User-network interface configuration

Each user-network interface can be configured as a

- Standard interface (operating or service numbers)
- Room interface
- Phone booth interface

are configured. The features of room and phone booth interfaces differ from those of standard interfaces.

This configuration is used for differentiation purposes in the OCL (reports, counter readings, threshold values).

Tab. 2.308: Configuring options for the user-network interfaces of a hotel system

	Standard	Room	Phone booth
Business and private calls	✓	private only	private only
Barring room-to-room traffic	1)	✓	✓
Operation via room menu	–	✓	✓
Activate message	✓	✓	✓
Answer message	✓	✓	✓
Activate wake-up orders	✓	✓	✓

1) Possible only via internal digit barring

8.7.2 Room-to-room traffic

If room-to-room traffic is enabled, guests have the possibility of telephoning directly to other rooms.
Traffic between rooms/phone booths can be configured with the following measures:

- General configuration as the basic setup for all rooms.
- Specific configuration per room port for room status "Occupied".
  If the room status is "Free", the system automatically resets the default values.

The "room-to-room traffic" configuration can be modified by the System Attendant of the Office 45, e.g. to allocate special privileges to travel groups.

Secret code

The secret code feature (*34) allows barred room-to-room traffic and internal digit barring to be bypassed. If *34 is barred in the internal digit barring, "secret code" cannot be activated. The room-to-room configuration applies exclusively.
The secret code allows key hotel management staff, e.g., to make calls on barred subscriber extensions. If the secret code is disclosed to a group of guests, room-to-room traffic can also be enabled.
Note: This feature is not described in any of the operating instructions.

8.7.3 Room status

Each room is allocated "Free" or "Occupied". These room statuses can be assigned configurations individually. When the status changes, the following actions take place automatically:

Room status from "Free" to "Occupied":
(Log guest on)

- The name of the previous guest is deleted. (Only if "Delete name generally" is configured under "Attendant_Hotel_Settings").
- Call charge counter reading ICC is deleted.
- Subscriber is given exchange access.
- Call Forwarding, Call Forwarding on No Reply, wake-up orders, etc., are deleted.
- Room-to-room authorization is set to the default configured in the "Setup" menu. If no default is configured, the current "Room-to-room" setting is retained.

Room status from "Occupied" to "Free":
(Log guest off)

- Subscriber’s exchange access is barred.
- Call Forwarding, Call Forwarding on No Reply, etc., are deleted.
- The individual room report containing all the call data is printed out if the option "Yes" is configured in the menu "Attendant_Hotel_Setup_Receipt".

The wake-up orders and the ICC data are not automatically deleted during log-off as the guest may well settle his bill in the evening but leave only during the night or the following morning.

8.7.4 Wake-up call

Unanswered wake-up calls are normally displayed at Reception with "Wake-up call refused". This alarm can be switched off in the general hotel settings.

8.7.5 Phone booth

A hotel phone box allows guests to make external calls with charge recall and the hotel staff itself to make internal calls. It is also possible to pick up calls and to transfer calls (for instance pick up calls). This relieves the workload on the reception staff.

The Office 45 or the PC Operator can be used as an operator console.

Example:

Setting up a phone booth:

1. Subscriber configuration for No. 210:
   – Extension: Phone booth
   – Exchange access: No
   – Internal digit barring: 9
   – External digit barring: 10 (or no digit barring)
2. Internal digit barring 9:
   – All barred
   – Enabled list:
     0 (exchange access)
     *86 (call pick up)
     R (control key)
     5 (internal numbers beginning with 5)
3. External digit barring 10: (as required)
   – All enabled
4. The following macro is configured on one of the free keys of the terminal from which the charge recall is to be activated (normally at reception):

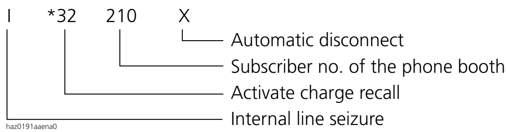


Fig. 2.216: Configuration of a key with charge recall

Phone booth operation, variant 1

Subscriber set No. 45 is defined as a hotline. When the receiver goes off-hook, "11" is dialled automatically and the operator console starts ringing.

```
I: Call from phone booth No. 45                      07:45
I: Call to phone booth No. 45                        07:46




-- Line key 1...5 _________________o__
Phone booth No. 45 031 885 23 12      THU  28. SEP 2000
Enquiry call     DTMF     Park      Message
```
haz1246aaena0

Fig. 2.217: Signalling on the Operator Console with variant 1 of phone booth operation

Operating sequence on the Operator Console

- Answer the internal call on the corresponding line key
- Press the phone box key for internal call (*3245 configured)
- Press hold key for enquiry call to Foxkey
- Press free line key
- Press the End key > phone box obtains dial tone and can dial out

When the call in the phone booth is completed the charge recall signal will ring on the Operator Console, and the call charge information is displayed (possibly with a delay, depending on the configuration).

```
I: Charge recall phone booth No. 45   CHF 001.40  07:49





-- Line key 1...5 _________________o__
Phone booth No. 45 031 885 23 12      THU  28. SEP 2000
Enquiry call     DTMF     Park      Message
```
haz1247aaena0

Fig. 2.218: Indication of charge recall

Phone booth operation, variant 2

Guest subscriber on set No. 45 notifies Reception that he wants to make a phone call.

```
I: Call from phone booth No. 45                     07:46
I: Call to phone booth No. 45                       07:46




-- Line key 1...5 _________________o__
*3245                                    THU 28. SEP 2000
Enquiry call     DTMF     Park      Message
```

haz1248aaena0

Fig. 2.219: Signalling on the Operator Console with variant 2 of phone booth operation

The guest in the phone box picks up the receiver within 2 minutes and obtains a dial tone. The line is signalled as "busy" on the Operator Console.

Operating sequence on the Operator Console:

- Press phone box key (*3245 configured)
- Press Enter key
- Press the End key

When the call in the phone box is completed the charge recall signal rings on the Operator Console and the call charge information is displayed in the same way as in variant 1 (possibly with a delay depending on the configuration).

Phone booth operation, variant 3

Subscriber 29, who does not have exchange access authorization, picks up the receiver and dials the operator’s number (11). He asks for a trunk line and puts down the receiver.

```
I: Call from subscriber 29                                  07:46
I: Function key charge recall (*32)                         07:47




-- Line key 1...5 _________________o__
*3245                                       THU 28. SEP 2000
Enquiry call      DTMF      Park      Message
```

haz1249aaena0

Fig. 2.220: Signalling on the Operator Console with variant 3 of phone booth operation

Operating sequence on the Operator Console:

- Press function key (*32 configured)
- Suffix dial 29 or wait 2 seconds
- Press the End key

As a call is signalled, the subscriber in the phone box picks up the receiver, obtains a dial tone and dials.

When the call in the phone box is completed the charge recall signal rings on the Operator Console and the call charge information is displayed in the same way as in variant 1 or 2 (possibly with a delay depending on the configuration).

8.7.6 Menu and operating concept with AIMS

Hotel features can be configured using AIMS subject to the following requirements:

- The PBX is connected to a PC on which AIMS has been installed (see "Access concept", page 151).
- The Hotel Manager is activated (see "AIMS Managers", page 146).

8.7.7 Hotel management systems

A PC capable of running functions in the PBX using command language can be connected to the V.24 interfaces of the basic system. A special command set is available for communicating with hotel management systems. These commands can then be used to carry out subscriber-specific polling operations (e.g. room status) and configurations (e.g. wake-up time). To be able to send the commands, you need to switch over to a special mode first.

For more detailed information on the mode, the command set, the connection possibilities and hotel management systems please contact Support or go directly to **open.interfaces@aastra.com**

8.8 Operation with external messaging, monitoring and alarm systems

External messages and external alarms are handled differently in the PBX.

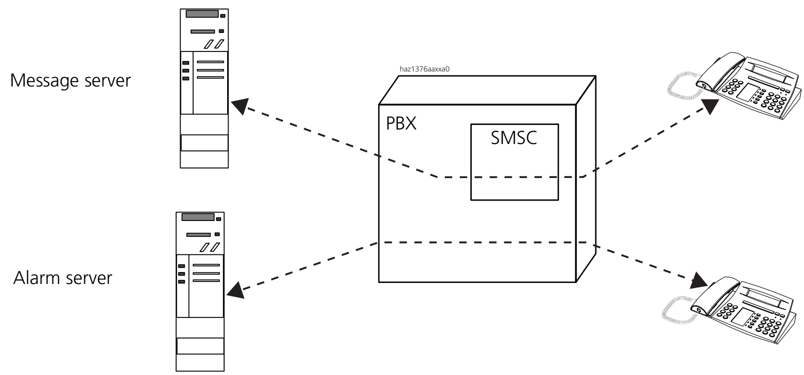


Fig. 2.221: Message and alarm handling

8.8.1 Message handling

External and internal messages are first sent to the SMSC (Short Message Service Centre), which then forwards the messages to the corresponding destination terminal. The SMSC is a software package integrated in the PBX that is responsible for the flow of messages within the PBX.

Up to 16 messages can be buffered for each terminal. Undeliverable messages (e.g. terminal memory full) are buffered in the SMSC (up to 400 messages). Overflow of the terminal memory is signalled accordingly on the display of the system terminal. A new attempt at delivering buffered internal messages is made after a configurable resend period. The messages are definitively deleted once the validity period, which is also configurable, has expired. With external messages the validity period is usually also transmitted. If not, the internal setting is also used. The resend period for external messages is always one quarter of the validity period.

AIMS can be used to delete all pending messages or messages that are more than three days old on all system terminals (see Tab. 2.213).

The SMSC is configured using the AIMS Configuration Manager:

Tab. 2.309: SMSC settings

Parameter	Remarks
SMSC port	The address is used for incoming messages of an external message server.
Retransmission period	Applies only to internal messages. For external messages the resend period is always one quarter of the validity period.
Validity period	It is only used for external messages if a validity period was not transmitted along with the message.

If the external message server is capable of handling short messages (SMS), then it is an ESME (External Short Message Entity). An ESME always communicates with the PBX via LAN. The PBX is capable of communicating with up to 10 different ESMEs. Examples of ESMEs include the IMS (Integrated Message Server) and Mailgate (both Ascom Wireless Solutions products).

The communication settings between the SMSC and the ESME can be configured in AIMS:

Tab. 2.310: ESME settings

Parameter	Remarks
IP address, port	IP address and port of the ESME for outgoing messages
Type	The ESME type is specified here (SM application, SM gateway or SM Service Centre).
Encryption	This parameter must match the setting on the external message server.
URL of ESME configuration	With the aid of this address a web-based configuration tool of an ESME can be retrieved directly via the AIMS Shell.

Parameter	Remarks
Default ESME	Messages without a destination address are sent to the default ESME.
Password	The password is checked every time a connection is set up between SMSC and ESME (minimum length = 4).
Response time	Max. amount of time the SMSC waits for an answer from the ESME [0...5 minutes]

Besides these settings the authorization to send short messages to an ESME can be enabled or disabled for each subscriber:

"Send SMS = Yes / No" (Inizialitation settings = Yes)

A web-based configuration is loaded into the browser via the AIMS Shell for the configuration of the ESME.

The use of the SMPP protocol to integrate an SMS server is subject to a licence (see "Licence-related System and Expansion Limits", page 593).

8.8.2 Alarm handling

External alarms from an alarm server are not handled by the SMSC but sent directly to the corresponding destination terminal. No more alarms can be sent to the corresponding terminal if the storage location is full. The alarm server is responsible for ensuring that alarms are delivered.

Other Properties and System Limits:

- Alarms take priority over messages.
- Max. length of alarm message texts is 160 characters.
- A maximum of 16 alarms can be stored for each subscriber. No more alarms can be delivered after that.
- Alarms are always routed to the destination defined in the send command; Call Forwarding Unconditional and Call Forwarding on No Reply operations have no effect.
- Several alarm sources can be connected to each PBX.

Warning:
In the case of applications designed for emergency calls and personal protection such as fire alarm systems, nurse light paging systems, alarm systems against attacks or hold-ups, etc., text messages may only be used to complement certified alarm systems. Text message alarming is only compatible with emergency operation if the PBX and the external alarm source are equipped with a UPS.

8.8.3 Connection via V.24 with ATPC1

External alarms, faults and messages from building energy management systems, nurse paging systems, security systems, etc., can be signalled as text messages to Office terminals via the V.24 interface of the Pocket Adapter. A licence is not required for connecting external alarm sources or messaging systems in this way.To ensure the maximum availability of the persons to be alarmed, Office DECT terminals should be used in preference for receiving the text messages.

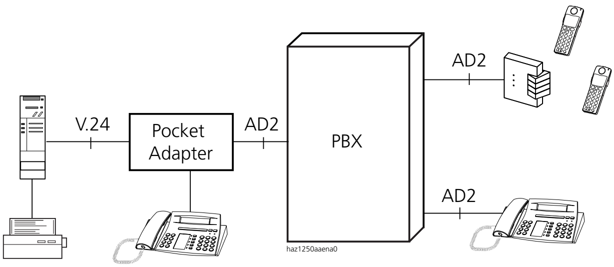


Fig. 2.222: Connection via Pocket Adapter

External alarm source

Any configurable systems equipped with a V.24 interface can be connected to the PBX as an external alarm source. These systems must also be capable of sending and receiving AT commands defined in the Ascotel standard ATPC1. A log printer can be provided on the external alarm source side to record the external alarms.

Connection to the PBX

To connect an external alarm source you need an AD2 user-network interface with an internal subscriber number already created and a corded Office terminal with a looped Pocket Adapter.

The following system terminals are capable of receiving external alarms:

- Office 25, Office 35, Office 45
- Office 30, Office 40
- Office 100, Office 135, Office 150, Office 155pro

External Alarms on Office System Terminals

Receiving external alarms

An external alarm is signalled on the system terminal either by continuous ringing or by a set ringing duration (0...3600 s) consisting of a cycle that alternates 4 s of ringing with a 1 s pause. The alarm text is indicated on the display at the same time. In this status the option to read, acknowledge, reject or delete the message is available.

Displaying an external alarm source

As long as the alarm has not been cleared, the Info key can be used to display the source of the alarm together with the subscriber's number and name.

Reading external alarms

The "Read" menu item stops the continuous ringing and the entire text message is then displayed (press Info key if necessary). The alarm server obtains a message to say that the alarm has been read.

Acknowledging or rejecting external alarms

Acknowledging or rejecting the message stops the continuous ringing, and the alarm server is notified of the user's response. The server can then send a confirmation to the system terminal, whose display then reads "Done". This tells the user that his response has been processed by the alarm server.

Deleting external alarm

The alarm can be cleared at any time either by the external alarm source or manually by the user. The alarm server is notified accordingly if the alarm is cleared by the user. If there is another alarm in the queue, it will be displayed once the previous alarm has been cleared.

8.8.4 Connection via V.24 / Ethernet with ATAS

Compared with ATPC1 the ATAS protocol provides additional possibilities for display on the system terminals (Fox menu) and allows an alarm to be triggered using the Redkey (see "Function Redkey", page 561). The connection is also monitored, and the connection set-up is password-protected. An ATAS licence is required for enabling the protocol (see "Licence-related System and Expansion Limits", page 593). This licence expands the possibilities for connecting external alarm and messaging sources to the PBX by providing the Ethernet interface and the V.24 interface of the system, in addition to the Pocket Adapter.

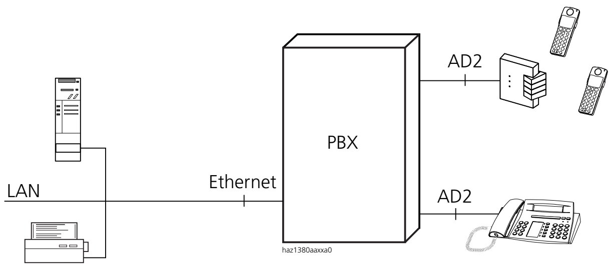


Fig. 2.223: Connection via Ethernet

Function Redkey

On each Office terminal one or more function keys can be configured as Redkeys. Depending on the application an alarm can then be triggered, a heating system switched on, a process controlled, etc., with the aid of the ATAS protocol on an ATAS server. The message sent contains the subscriber number and additional parameters (max. 32 characters/digits).

Other properties:

- The configuration is set for each subscriber and can only be made via AIMS.
- The function can be stored on any configurable keys of the Office terminals.
- Several keys on each terminal can be configured as Redkeys.
- The Redkey function is triggered by a single, double or long-click of the corresponding key.
- Once a Redkey has been configured, it can only be reconfigured via AIMS.
- The application on the ATAS server can acknowledge that a function has been triggered by a Redkey by sending a message to the terminal's display (with or without a prompt to acknowledge the message).

Hotkey modes on DECT system terminals

On DECT system terminals the Redkey function can be configured on the Hotkey. To ensure that again only one keystroke is needed to trigger the function, the Hotkey can be limited to one storage location (instead of 6) by using the parameter "1 Hotkey only = Yes" in the subscriber settings. In this mode the Hotkey is triggered even if the keypad lock is activated (requires a long click). This setting can be configured with AIMS for each DECT terminal.

8.8.5 Interface descriptions

The ATAS and ATPC1 protocols can be disclosed to interested manufacturers of messaging, monitoring and alarm equipment on request. Contact Support or go directly to "**open.interfaces@aastra.com**".

8.9 Handset as visitor

An Ascotel DECT subscriber can be logged on via AIMS or the System Assistant for the duration of a visit (see "Ascotel DECT configuration", page 857).

One hour before the visiting time expires, the system sends the message "<DECT SC No.> will be logged off in one hour" to a configurable internal subscriber.

When the visiting time expires, the handset is automatically logged off from the system.

9 Features Overview

Here is provided an alphabetical overview in table form of the features that can be operated on the terminals.

Tab. 2.311: Legend used in the table of features

	Menu or Foxkey-operated feature (also via */# procedures)
ISDN	Feature available as a standard ISDN service (ETSI signalling) and therefore menu-operated on commercially available ISDN terminals (also via */# procedures)
*/# procedure	Feature actuated only by using a */# procedure. For pulse dialling telephones without the * key, a substitute * can be defined in the numbering plan (e.g. "9")
R	Feature actuated using the control key
✓	Feature available on the terminal
TM	Feature supported by the PBX. Its availability depends on the terminal
Digit	Digit suffix dialling (activated DTMF)
–	Feature not supported on this terminal

Tab. 2.312: Features overview (several pages)

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Abbreviated dialling numbers, throughout the system	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Allocate cost centre before the call	see under "**Exchange Access**"									
Announcement										
• Answer within the group							–	✓	–	–
• Answer outside the group	*89	*89	*89	*89	*89	*89	*89	*89	*89	*89
• Initiate to a subscriber	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.	*7998 SC No.
• Initiate with autom. duplexmode			–			–	–	–	–	–
• Initiate to a group	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.	*79 Gr. No.
• Protect against	*04	*04	*04	*04	*04	*04	*04	*04	*04	*04
• Allow to own set	#04	#04	#04	#04	#04	#04	#04	#04	#04	#04
Answering general bell										
• Coded ringing	see under "**Coded ringing on generalcall**"									
• Ringing signal						*83	*83		*83	*83
Appointment call										
• Individual call order Activate	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm	*55 hhmm
• Permanent call order Activate	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm	*56 hhmm
• Clear	#55 or #56	#55 or #56	#55 or #56	#55 or #56	#55 or #56	#55 or #56	#55 or #56	#55 or #56	#55 or #56	#55 or #56

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Brokering										
• In an enquiry call									ISDN	R2
• with line key	✓	✓	–	✓	✓	–	–	–	–	–
Busy panel	✓	–	–	–	–	–	–	–	–	–
Call acceptance from connection	–	–	–	–	–	–	–	–	–	R1
Call charge display										
• For outgoing exchange calls	✓	✓	✓	✓	✓	✓	-	✓	ISDN	–
• For transferred exchange calls	✓	✓	✓	✓	✓	✓	-	✓	ISDN	–
Call charges										
• Charge management	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
• Transfer current call to another cost centre	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.	*78 CC No.
• Individual charge counting (ICC)	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
• Charge recall	*32 SC No.	*32 SC No.	*32 SC No.	*32 SC No.	*32 SC No.	*32 SC No.	–	*32 SC No.	*32 SC No.	–
Call deflection (CD)	see under "**Deflecting a call during the ringing phase (CD)**"									
Call door intercom	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Call Forwarding if Busy (CFB)										
• Activate		.		*67 Dest. No.	*67 Dest. No.	*67 Dest. No.	*67 Dest. No.	*67 Dest. No.	ISDN	*67 Dest. No.
• Activate to SC last configured				*67#	*67#	*67#	*67#	*67#	*67#	*67#
• Clear	#67	#67	#67	#67	#67	#67	#67	#67	#67	#67
Call Forwarding on No Reply (CFNR)										
• Activate						*61 Dest. No.	*61 Dest. No.		*61 Dest. No.	*61 Dest. No.
• To general bell with coded ringing						*68	*68		*68	*68
• Clear to general bell with coded ringing						#68	#68		#68	#68
• To SC last configured						*61#	*61#		*61#	*61#
• Clear to SC last configured						#61	#61		#61	#61
• To preconfigured SC	*62	*62	*62	*62	*62	*62	*62	*62	*62	*62
• Clear to preconfigured SC	#62	#62	#62	#62	#62	#62	#62	#62	#62	#62
• To pager						*68	*68		*68	*68
• Clear to pager						#68	#68		#68	#68
• Protect against	*02	*02	*02	*02	*02	*02	*02	*02	*02	*02
• Allow to own set	#02	#02	#02	#02	#02	#02	#02	#02	#02	#02

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Call Forwarding Unconditional (CFU)										
• Activate	🦊	🦊	🦊	🦊	🦊	*21 Dest. No.	*21 Dest. No.	🦊	*21 Dest. No.	*21 Dest. No.
• Activate to SC last configured	🦊	🦊	🦊	🦊	🦊	*21#	*21#	🦊	*21#	*21#
• Clear	🦊	🦊	🦊	🦊	🦊	#21	#21	🦊	#21	#21
• Activate to preconfigured SC	*22	*22	*22	*22	*22	*22	*22	*22	*22	*22
• Clear to preconfigured SC	🦊	🦊	🦊	🦊	🦊	#22	#22	🦊	#22	#22
• Activate to general bell with coded ringing / PS	🦊	🦊	🦊	🦊	🦊	*28	*28	🦊	*28	*28
• Clear to general bell with coded ringing / PS	🦊	🦊	🦊	🦊	🦊	#28	#28	🦊	#28	#28
• Activate to standard text	🦊	🦊	🦊	🦊	🦊	*24 Text No. Param.#	*24 Text No. Param.#	🦊	*24 Text No. Param.#	*24 Text No. Param.#
• Clear to standard text	🦊	🦊	🦊	🦊	🦊	#24	#24	🦊	#24	#24
• Protect against	*02	*02	*02	*02	*02	*02	*02	*02	*02	*02
• Allow to own set	#02	#02	#02	#02	#02	#02	#02	#02	#02	#02
Call pick-up (x: SC No. / UG No. / CDE No.)	🦊	🦊	🦊	🦊	🦊	*86 x	*86 x	🦊	*86 x	*86 x
Call transfer										
• after enquiry call	✓	✓	✓	✓	✓	✓	✓	✓	ISDN	✓
• without enquiry call	✓	✓	✓	✓	✓	✓	✓	✓	ISDN	✓
• Explicit call transfer (ECT)	–	–	–	–	–	–	–	–	ISDN	✓

Features	**Office 45**	**Office 35**	**Office 25**	**Office 40**	**Office 30**	**Office 20**	**Office 10**	**Office Handset**	**ISDN TM**	**Analogue TM (DTMF)**
Call waiting										
• Activate						*43 / 6	*43 / 6		*43 / 6	R*43 / R6
• Reject						End	End		ISDN	R0
• Answer with hold									ISDN	R2
• answer without hold						1	1		ISDN	R1
• answer with Conference	3	3	3	3	3	3	3	3	ISDN	R3
• Protect against	*04	*04	*04	*04	*04	*04	*04	*04	*04	*04
• Allow to own set	#04	#04	#04	#04	#04	#04	#04	#04	#04	#04
Callback if SC busy (CCBS) / available										
• Activate									ISDN	R9 or R*37
• Clear							#37		#37	#37
Change password x: old password y: new password							*47 x * y * y #		*47 x * y * y #	*47 x * y * y #
Clear configuration	*00	*00	*00	*00	*00	*00	*00	*00	*00	*00
Coded ringing on generalcall										
• Activate in prefix dialling				.		*81 SC No.	*81 SC No.		*81 SC No.	*81 SC No.
• Activate in suffix dialling						*81 / 8	*81 / 8		*81 / 8	R8 or R*81
• Answer						*82	*82		*82	*82
Conference										
• Set up (from connection)						3			ISDN	R3
• Set up (variable)	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.	*71 SC No.

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
• Expand (variable)	*SC No.	*SC No.	*SC No.	*SC No.	*SC No.	*SC No.	*SC No.	*SC No.	*SC No.	R*SC No.
• Terminate (variable)	#	#	#	#	#	#	#	#	#	#
• Exclude subscriber (internal)	#71	#71	#71	#71	#71	#71	#71	#71	#71	R#71
• Set up (predetermined)	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.	*70 Conf. No.
Courtesy Service (announcement prior to answering)										
• Activate	*93x	*93x	*93x	*93x	*93x	*93x	*93x	*93x	*93x	*93x
• Deactivate	#93x	#93x	#93x	#93x	#93x	#93x	#93x	#93x	#93x	#93x
• Record with handset	*91x y	*91x y	*91x y	*91x y	*91x y	*91x y	*91x y	*91x y	*91x y	*91x y
• Record from tape	*92x y	*92x y	*92x y	*92x y	*92x y	*92x y	*92x y	*92x y	*92x y	*92x y
• Check the recording	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y	*#91x y or *#92x y
• Delete the recording	#91x y	#91x y	#91x y	#91x y	#91x y	#91x y	#91x y	#91x y	#91x y	#91x y
• Activate wave files	*930	*930	*930	*930	*930	*930	*930	*930	*930	*930
x = Group [1..0.2] y = switch position [1...3]										
Deflecting a call during the ringing phase (CD)						only via function key	–		TM	–
Dialling by name	✓	✓	–	✓	✓	–	–	✓	–	–

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Discreet ringing										
• Activate	✓	✓	✓	✓	✓	–	–		–	–
• Deactivate	✓	✓	✓	✓	✓	–	–		–	–
Display caller's name (CNIP / CONP)	✓	✓	✓	✓	✓	–	–	✓	ISDN	–
Display caller's number (CLIP / COLP)	✓	✓	✓	✓	✓	✓	–	✓	ISDN	–
Do not disturb (Call protection)										
• Activate		*26	*26		*26	*26	*26	*26	*26	*26
• Clear		#26	#26		#26	#26	#26	#26	#26	#26
DTMF dialling	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Duplex mode	see under "**Announcement**"									
Emergency / priority exchange seizure	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Emergency number	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Enquiry call										
• To own system									ISDN	R SC No.
• To up-circuit system	*42 SC No.	*42 SC No.	*42 SC No.	*42 SC No.	*42 SC No.	*42 SC No.	*42 SC No.	*42 SC No.	*42 SC No.	R*42 SC No.
Exchange Access										
• Business (example CH)	0	0	0	0	0	0	0	0	0	0
• Least Cost Routing	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
• LCR (fallback)	*90	*90	*90	*90	*90	*90	*90	*90	*90	*90
• Private (example CH)	10	10	10	10	10	10	10	10	10	10
• With cost centre nn	13nn	13nn	13nn	13nn	13nn	13nn	13nn	13nn	13nn	13nn

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
• With charge recall	*32 SC No.	*32 SC No.	*32 SC No.	*32 SC No.	*32 SC No.	*32 SC No.	–	*32 SC No.	*32 SC No.	–
• Route selection, targeted (n depends on the system)	170 to n	170 to n	170 to n	170 to n	170 to n	170 to n	170 to n	170 to n	170 to n	170 to n
Features Remote control	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature proced.	*06 SC No. Feature pro-ced.
Follow me										
• Activate	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.	*23 SC No.
• Clear	#23	#23	#23	#23	#23	#23	#23	#23	#23	#23
Hold connection (HOLD)									ISDN	✓
• Connection from hold state	*94	*94	*94	–	–	–	–	–	–	–
• Connection nn from hold state	*95 nn	*95 nn	*95 nn	–	–	–	–	–	–	–
Intrusion										
• Activate		*44	*44	*44	*44	*44	*44	*44	–	R7 or R*44
• Reject						0	0		–	R0
• Answer with hold							2		–	R2
• Answer without hold						1	1		–	R1
• Answer with conference	3	3	3	3	3	3	3	3	–	R3
• Protect against	*04	*04	*04	*04	*04	*04	*04	*04	*04	*04
• Allow to own set	#04	#04	#04	#04	#04	#04	#04	#04	#04	#04

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Leave message										
• Standard							*24 Text No. Param.#		*24 Text No. Param.#	*24 Text No. Param.#
• Own	✓	✓	✓	✓	✓	–	–	✓	–	–
List of callers						–	–		TM	–
Making calls with your own settings on a third-party telephone										
• Business calls	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW
• Private calls	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW
MESSAGE LED										
• Activate (prefix dialling)							*38 SC No.		*38 SC No.	*38 SC No.
• Activate (suffix dialling)	–	–	–	–	–	–	–	–	–	R*38
• Answer (own set)							*#38		–	–
• Clear (own set)							#38#		–	–
• Clear (destination set)							#38 SC No.		–	#38 SC No.
Opening doors	*751…2	*751…2	*751…2	*751…2	*751…2	*751…2	*751…2	*751…2	*751…2	*751…2

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Pager functions										
• Search (prefix dialling)	🦊	🦊	🦊	🦊	🦊	*81 SC No.	*81 SC No.	🦊	*81 SC No.	*81 SC No.
• Search (suffix dialling)	🦊	🦊	🦊	🦊	🦊	*81	*81	🦊	*81	R8 or R*81
• Answer	🦊	🦊	🦊 .	🦊	🦊	*82	*82	🦊	*82	*82
Park										
• with line key	✓	✓	–	✓	✓	–	–	–	–	–
• with park key (local)	🦊	🦊	–	🦊	🦊	–	–	🦊	–	–
• Central parking	*76	*76	*76	*76	*76	*76	*76	*76	*76	*76
• Connect with centrally parked SC	#76	#76	#76	#76	#76	#76	#76	#76	#76	#76
Private calls with Password	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW	#46 SC No. PW
Private phone book (names / numbers)	✓	✓	✓	✓	✓	✓	✓	✓	–	–
Recording malicious calls (MCID)	✓	✓	✓	✓	✓	✓	✓	✓	ISDN	–
Reject call	🦊	🦊	🦊	🦊	🦊	–	–	🦊	ISDN	–
Relays										
• Activate (PBX)	*755...7	*755...7	*755...7	*755...7	*755...7	*755...7	*755...7	*755...7	*755...7	*755...7
• Deactivate (PBX)	#755...7	#755...7	#755...7	#755...7	#755...7	#755...7	#755...7	#755...7	#755...7	#755...7
• Switch group x in position y	*85 xy	*85 xy	*85 xy	*85 xy	*85 xy	*85 xy	*85 xy	*85 xy	*85 xy	*85 xy

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
Remote maintenance / Configuration										
• Enable / bar a one-off remotemaintenance access	*754 / #754	*754 / #754	*754 / #754	*754 / #754	*754 / #754	*754 / #754	*754 / #754	*754 / #754	*754 / #754	*754 / #754
• Enable / bar a repeated remotemaintenance access	*753 / #753	*753 / #753	*753 / #753	*753 / #753	*753 / #753	*753 / #753	*753 / #753	*753 / #753	*753 / #753	*753 / #753
Ringing relay with delay (line keys and team keys)	0, 10, 20, 30 s	0, 10, 20, 30 s	–	0, 10, 20, 30 s	0,10 s	–	–	–	–	–
Secret code (disable room-to-room barring)	*34	*34	*34	*34	*34	*34	*34	*34	*34	*34
Signalling from one user to another user (UUS-1)	–	–	–	–	–	–	–	–	ISDN	–
Subaddressing (SUB)	–	–	–	–	–	–	–	–	ISDN	–
Suppression of the call number display for each call (CLIR per call)	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.	*31 Dest. No.
Switch over switch groups 1...9 x = Group [1...9] y = switch position [1...3]	*85x y	*85x y	*85x y	*85x y	*85x y	*85x y	*85x y	*85x y	*85x y	*85x y
Taking over an active connection										
• Activate	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88	*88# or *87*88
• Preset authorisation call takeover from nn to mm	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#	*87 nn*mm#

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
• Preset authorization handover data connection from nn to mm	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#	*84 nn*mm#
• Clear authorization handover call active /passive	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.	#87 SC No.
• Clear authorization handover data connection from nn to mm	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.	#84 SC No.
Team keys	✓	✓	–	✓	✓	–	–	–	–	–
Telephone barring (Subscriber) PW = Password										
• Activate							*33#	*33#	*33#	*33#
• Activate with new password							*33 PW	*33 PW	*33 PW	*33 PW
• Deactivate							#33 PW	#33 PW	#33 PW	#33 PW
• Single call unlock	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW	#36 SC No. PW
Text messages										
• View						–	–		–	–
• Send (standard text with / without parameters) to SC						*3598 SC No. Text No.#	*3598 SC No. Text No.#		*3598 SC No. Text No.#	*3598 SC No. Text No.#
• Send (standard text with / without parameters) to group						*35 Gr. No. Text No.#	*35 Gr. No. Text No.#		*35 Gr. No. Text No.#	*35 Gr. No. Text No.#

Features	Office 45	Office 35	Office 25	Office 40	Office 30	Office 20	Office 10	Office Handset	ISDN TM	Analogue TM (DTMF)
• Send (standard text with / without parameters) to all						*3599 Text No.#	*3599 Text No.#		*3599 Text No.#	*3599 Text No.#
• Send user-definable message text	✓	✓	✓	✓	✓	–	–	✓	–	–
Transfer cost centre during the call	see under "**Call charges**"									
Two-company configuration	✓	–	–	–	–	–	–	–	–	–
User groups (UG) (optional)										
• Log into all user groups	*4800	*4800	*4800	*4800	*4800	*4800	*4800	*4800	*4800	*4800
• Log out of all user groups	#4800	#4800	#4800	#4800	#4800	#4800	#4800	#4800	#4800	#4800
• Log into specific user groups	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.	*48 UG No.
• Log out of specific user groups	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.	#48 UG No.

Part 3 Planning

1 Overview of Chapters

Expansion Stages

Basic systems can be expanded using expansion cards and licences. Chapter 2 describes which expansion options are available for which basic system and contains examples of component placement.

System and Expansion Limits

Chapter 3 provides an overview of the technical and licence-related limits to the expansion of basic systems. Here we need to differentiate between fixed technical system limits, e.g. the number of interfaces and slots, and power supply or memory-related limits, which depend on the number of terminals and their power requirements. This chapter contains the basics for manual calculation.

Planning a PBX

The first phase of any successful planning project is to determine the customer's actual communication situation, his requirements and what he has in mind for the expansion of the system. You can enter this customer-related information directly into the AIMS Project Manager. A rough guideline quotation is then drawn up on the basis of the information and adapted with the customer.

After this coordination process a definitive quotation is drawn up in phase 2; this is where you begin with the detailed planning of the communication system. Functions relevant to the system's operation are discussed with the customer and implemented in the Project Manager.

The AIMS Project Manager assists you in all the phases of system planning and automatically takes account of the system and expansion limits. With its help you can draw up a complete documentation package (list printouts) for the planned systems.

Planning DECT systems

Chapter 5 deals with the planning of cordless systems. With cordless systems you need not only to clarify the customer's needs and requirements but also to provide him beforehand with in-depth information on the performance of this complex technology, to ensure he is able to make the right decision with regard to the system's design. Complex measurements also have to be carried out on site, measurements which are not required with private leased-line networks.
The process of coordination between manufacturer, project manager and customer is far more intensive and, consequently, time-consuming than it is with leased-line networks.

Planning a private network

Additional planning aspects need to be taken into account when networking several PBXs: The location for the PINXs, the connections between the PINXs themselves and the public network, the clock synchronization, and routing matters in general. Special attention also has to be paid to the numbering plan over several PINXs to ensure that number ranges are grouped together in the best possible way. Chapter 6 provides the appropriate guideline, with instructions on how to set up a network step by step, and a full description of the pros and cons of various methods.

2 Expansion Stages

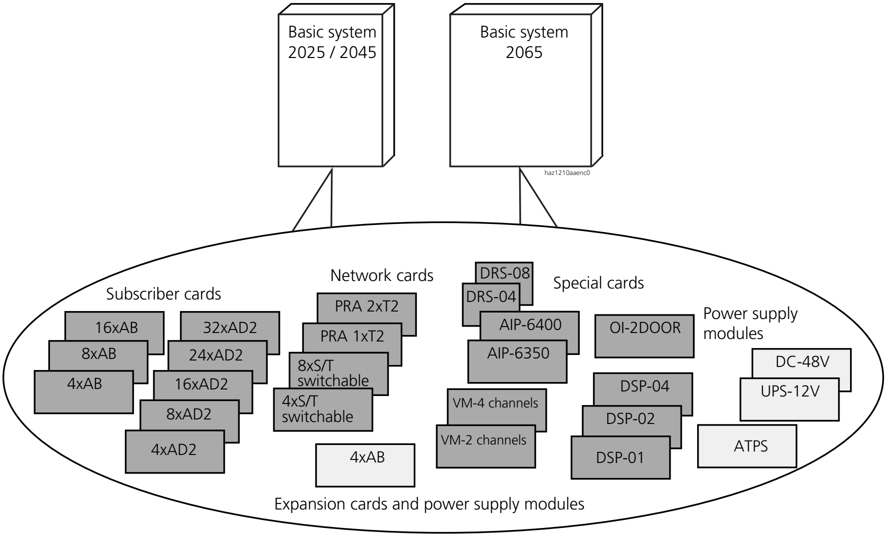


Fig. 3.1: System overview

Within the limits of the system the basic systems can be expanded using expansion cards and licences. To be able to adapt the overall concept of a PBX as best as possible to the customer's requirements, you need to know the available expansion cards and the system limits of the various basic systems and their expansion stages.

With the project data the optimum hardware configuration is easily determined using the AIMS Project Manager.

2.1 System Family

The system family consists of three systems:

- Ascotel IntelliGate 2025
- Ascotel IntelliGate 2045
- Ascotel IntelliGate 2065

The systems differ in the number of expansion slots and are based on two basic systems:

- 2025 / 2045 basic system
- 2065 basic system

Ascotel IntelliGate 2025 and Ascotel IntelliGate 2045 differ only through the expansion limits defined by the licences

Tab. 3.1: Basic systems and number of expansion slots per system

	Basic system	Expansion slots
Ascotel IntelliGate 2025	2025 / 2045	5
Ascotel IntelliGate 2045	2025 / 2045	5
Ascotel IntelliGate 2065	2065	14[1)]

1) An additional slot (No. 8) is reserved for the processor card.

Each system is available in a wall-mounting and rack-mounting variant. Unless otherwise indicated, all the information provided in this System Manual applies to both variants.

2.2 Basic Systems

Tab. 3.2: Interfaces of the basic systems

	a/b	S/T (individually switchable)	AD2	V.24	Ethernet interface	Audio input	Relay for General bell
2025 / 2045 basic system	3	3	4	2	1 (10 Base T)	1	1
2065 basic system	-	-	-	2	1 (10 / 100 Base T)[1)]	1	1

1) on processor card MPC-8260

Tab. 3.3: Resources of the basic systems

	Three-party conferences	Six-party conferences	Courtesy (announcement prior to answering)	DTMF transmitter	DTMF receiver	Internal music
2025 / 2045 basic system	3	1	approx. 1 minute	6	4	approx. 1 minute
2065 basic system	10	4	approx. 2 x 1 minute	26	16	approx. 1 minute

2.2.1 Basic system 2025 / 2045

The 2025 / 2045 basic system provides the basis for the systems Ascotel IntelliGate 2025 and Ascotel IntelliGate 2045. It consists of the following components:

- 2025 / 2045 housing
- Main board 2025 / 2045 incl. power supply
- Memory cards

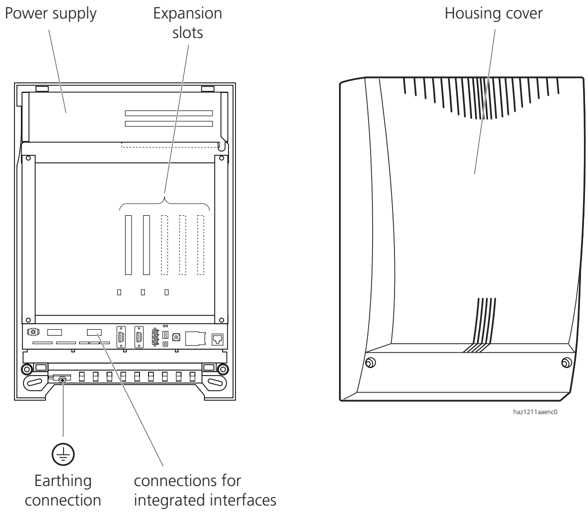


Fig. 3.2: 2025 / 2045 basic system

2.2.2 2065 basic system

The 2065 basic system provides the basis for the Ascotel IntelliGate 2065 system. It consists of the following components:

- 2065 housing
- Main board 2065 incl. power supply
- Processor card MPC-8260
- Memory cards

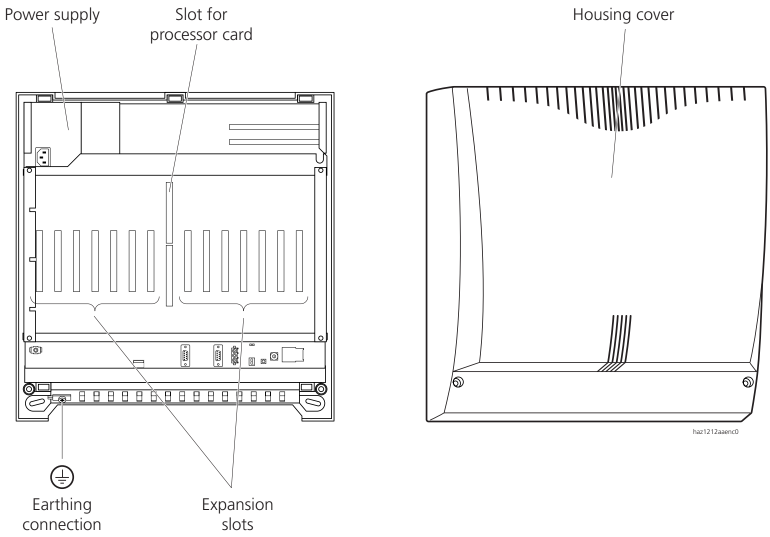


Fig. 3.3: 2065 basic system

2.2.3 System cards

System cards are integral parts of the basic systems and are supplied along with them. The systems are factory fitted with the system cards. System cards do not take up any expansion slots.

Processor card MPC-8260

Processor card MPC-8260 contains the system processor, the RAM card and the Ethernet interface for the Ascotel 2065 basic system.

Memory cards

Tab. 3.4: Memory cards per system

	2025 system	2045 system	2065 system
Flash card	1 x 21 MB	1 x 21 MB	1 x 25 MB
RAM card	1 x 32 MB	1 x 32 MB	1 x 64 MB
EIM card	1 x 256 bytes	1 x 256 bytes	1 x 256 bytes

Another slot is provided so the system can be subsequently expanded with a second Flash card.

2.3 Power Supplies

2.3.1 Power supply modules

Two optional DC power supply modules are available in addition to the power supply via the 230 V mains network:

- UPS-12V for the uninterruptible power supply from a 12V buffer or central battery
- DC-48V for the power supply from a 48V central battery

Each system can be complemented with a power supply module. This does not take up an expansion slot.

2.3.2 Auxiliary Terminal Power Supply (ATPS)

The external Auxiliary Terminal Power Supply (ATPS) is used to increase the supply power available for the terminals. It is required only for system expansions with a large number of terminals with high power requirements (e.g. a large numbe of DECT radio units without their own power supplies) (see also "Power supply-related limits per system", page 597).

The ATPS power supply unit is powered by the 230 V mains.

It is not possible to use ATPS and a DC supply module at the same time. An external UPS or an inverter is required to be able to ensure uninterruptible operation when using an ATPS system.

2.4 Expansion cards

The basic systems are expanded by fitting out the expansion slots. The 2025 / 2045 mainboard contains 5 of these slots. An upgrade licence (2025 → 2045) enables higher expansion limits (max. number of interfaces). The Ascotel 2065 mainboard contains 14 expansion slots.
The expansion cards are the same for all Ascotel IntelliGate systems. In principle any card can be used in any system (exception: The expansion card with two primary rate accesses (ISDN-02PRA) is not designed for Ascotel IntelliGate 2025). However the number of interfaces enabled is determined in each case by the system limits.

There are 3 types of expansion cards available for expanding the basic systems:

- Trunk cards
- Subscriber cards
- Special cards

2.4.1 Trunk cards

Trunk cards expand the number of network interfaces available to a system.

Tab. 3.5: Number of trunk cards of one particular type per PBX

Card type	Network interfaces per card	Max. number for Ascotel IntelliGate		
		2025	2045	2065
ISDN-04ST	4 x S/T[1)]	1	2	8
ISDN-08ST	8 x S/T[1)]	1[2)]	1	8
ISDN-01PRA	1 x T2[3)]	1	2	8
ISDN-02PRA	2 x T2[3)]	-	1	8
TC-04AB	4 x a/b	1	2	8

1) Interfaces are individually switchable

2) Due to the system limits not all the interfaces can be used (see Tab. 3.18)

3) Selectable protocol: DSS1 (network connection), QSIG (ETSI or ISO)

2.4.2 Subscriber cards

Subscriber cards expand the number of user-network interfaces.

Tab. 3.6: Number of subscriber cards of one particular type per PBX

Card type	User-network interfaces per card	Max. number for Ascotel IntelliGate		
		2025	2045	2065
SC-04AD2	4 x AD2	2	4	14
SC-08AD2	8 x AD2	1	2	14
SC-16AD2	16 x AD2	1[1)]	2	14
SC-24AD2	24 x AD2	1[1)]	1	13
SC-32AD2	32 x AD2	1[1)]	1	10
SC-04AB	4 x a/b	2	4	14
SC-08AB	8 x a/b	1	2	10

Card type	User-network interfaces per card	Max. number for Ascotel IntelliGate		
		2025	2045	2065
SC-16AB	16 x a/b	1[1)]	1	10
ISDN-04ST	4 x S/T[2)]	1	2	8
ISDN-08ST	8 x S/T[2)]	1[1)]	1	8

1) Due to the system limits not all the interfaces can be used (see Tab. 3.18)

2) Interfaces are individually switchable

2.4.3 Special cards

All cards that are not part of the families of trunk cards or subscriber cards belong to the category of special cards.

IP interface cards AIP-6400 and AIP-6350

These cards with a different functionality are both based on the card IPI-100BT. It features a 10/100 MBit/s Ethernet interface with automatic detection of the transmission speed. Depending on the software loaded, the IPI-100BT card becomes the AIP-6400 card or AIP-6350 card.

- With the AIP 6400 software the card functionally becomes an Ascotel IP gateway AIP 6400. The applications are VoIP (Voice over Internet Protocol) and QSIG networking in the IP network. The card makes it possible to network multiple PBXs in different locations via an existing voice-compatible IP network.
- With the AIP 6350 software the card functionally becomes an Ascotel IP gateway AIP 6350. Its application is to link up IP terminals via the AD2 protocol, enabling remote workstations to be integrated into the system.

It is possible to use the expansion cards AIP-6350 and AIP-6400 simultaneously in a single system. However the total number of cards must not exceed the limits in Tab. 3.13.

Tab. 3.7: Resources of the special cards AIP-6400 and AIP-6350 (basic card IPI-100BT)

AIP -6400	Number per IP interface card
Simultaneous VoIP / QSIG over IP connections	12
Ascotel systems networkable via IP	50
AIP -6350	
Call channels active simultaneously	12
Office 35IP / Office 1600IP terminals	32

DRS modules are needed for the real-time processing of voice data. They are AIP-specific modules mounted onto the IP interface card. One IP interface card contains 2 slots for DRS modules.

Tab. 3.8: Component placement variants for DRS modules and number of call channels

DRS-04	DRS-08	Max. number of call channels per IP interface card
1	-	4
2	-	8
-	1	8
1	1	12
	2	12[1)]

1) Determined by a software limit

16 AD2 interfaces are occupied on each AIP-6350 expansion card. The system limits can however restrict the maximum number of AD2 interfaces available for the IP terminals (see Tab. 3.18). It is also important to remember that on the 2025 / 2045 basic system the AD2 interfaces on the mainboard are enabled first. This results in the following values:

Tab. 3.9: IP system terminals and AD2 terminals

	2025 system	2045 system	2065 system
Max. number of IP system terminals	16	60	128
Max. number of AD2 interfaces available	8	32	64

A separate documentation is available with details of these special cards.

Voice Mail Cards VM-02P and VM-04P

Ascotel IntelliGate provides two card types which differ in the number of Voice Mail channels and recording time. Only one of these cards can be installed per system:

Tab. 3.10: Resources of the Voice Mail cards

Voice Mail card	Voice Mail channels[1)]	Number of mailboxes	Total recording time
VM-02P	2	128	4 h
VM-04P	4	128	8 h

1) One virtual a/b user-network interface is seized per channel

A separate documentation is available with details of this special card.

Voice Mail solutions with a large number of mailboxes and Voice Mail channels can be implemented using external Voice-Mail equipment. We recommend the product VME Office, with a maximum configuration of 500 mailboxes and 8 channels, and virtually the same functionality as the integrated Voice Mail system.

Options Card OI-2DOOR

The OI-2DOOR options card contains special interfaces for connecting door intercom systems, an interface for switch group 1 and relays.

Tab. 3.11: Special interfaces on the OI-2DOOR options card

Special interfaces	Number of entries
Door intercom system incl. door bell input, door opener, power supply relay	2
Interface for switching over switch group 1	1
Relays with two floating switchover contacts each	3

DSP Cards DSP-01, DSP-02 and DSP-04

DSP (Digital Signal Processor) cards are used if the system is to be equipped with cordless terminals (DECT). There are three card types, which differ in the number of available DECT voice channels. Pure DECT-DECT connections do not seize any DECT voice channels.
DSP-02 and DSP-04-type cards also enable full duplex hands-free operation on Office 45/45pro terminals. This increases the quality during hands-free mode. On all Office terminals with hands-free operation, half-duplex hands-free operation is also possible without a DSP card.

Tab. 3.12: Resources of the DSP cards

DSP card	Voice channels for DECT	Full duplex hands-free operation (Office 45/45pro)
DSP-01	12	-
DSP-02	18	6
DSP-04	36	9

Number of special cards per system

Tab. 3.13: Max. number of special cards of one particular type per PBX

Card type	Max. number for Ascotel IntelliGate		
	2025	2045	2065
AIP -6400 (basic card IPI-100BT)	1	2	4
AIP -6350 (basic card IPI-100BT)	1	2	4
Total AIP -6400 + AIP -6350	1	2	4
VM-02P/VM-04P	1	1	1
OI-2DOOR	1	1	1
DSP-01 / DSP-02	1	2	4
DSP-04	1	1	2
Total DSP-01 + DSP-02 + DSP-04	1	2	4

2.5 Component Mounting Rules and Examples

2.5.1 Component mounting rules

Besides the expansion limits specific to the cards, the following rules are to be observed with regard to fitting components:

- With the exception of the ISDN-02PRA card all the expansion cards can be used in all the systems. When the system is started up, however, only as many interfaces are enabled as are permitted by the expansion limits (see "System and Expansion Limits", page 592).
- On the basic system 2025 / 2045 the user-network interfaces on the mainboard are enabled before those on the expansion cards.
- The expansion cards are enabled "from left to right" and the interfaces on the expansion cards "from bottom to top". Rule: the lower designations come first (see also Fig. 3.4).
- If a limit value is reached when starting up the system or registering the cards, it is possible that not all the cards or not all the interfaces of the last card can be enabled.
- Voice Mail cards are considered as 2 or 4 analogue user-network interfaces during logon.

2.5.2 Examples of component placement

Tab. 3.14: Example of component placement with Ascotel IntelliGate 2025

Location	Card	Network interfaces	User-network interfaces	Note
2025 / 2045 basic system	(MBS)	2	8	(2xT, 1xS, 3xab, 4xAD2)
Expansion slot 1	OI-2DOOR	-	-	
Expansion slot 2	DSP-01	-	-	12 DECT channels
Expansion slot 3	SC-08AD2	-	8	of which 2 interfaces are used for radio units
Expansion slot 4	-	-	-	
Expansion slot 5	-	-	-	
	Total	2 (of max. 8)	16 (of max. 16)	max. 30 subscribers with their own number

Tab. 3.15: Example of component placement with Ascotel IntelliGate 2045

Location	Card	Network interfaces	User-network interfaces	Note
2025 / 2045 basic system	(MBS)	2	8	(2xT, 1xS, 3xab, 4xAD2)
Expansion slot 1	OI-2DOOR	-	-	
Expansion slot 2	VM-02P	-	2	
Expansion slot 3	SC-08AD2	-	8	of which 4 interfaces are used for radio units
Expansion slot 4	DSP-02	-	-	18 DECT channels
Expansion slot 5	AIP -6350	-	16	Max. 12 calls possible simultaneously
	Total	2 (of max. 19)	34 (of max. 40)	max. 60 subscribers with their own number

The figure below uses the complement placement example Ascotel IntelliGate 2045 to illustrate the sequence in which interfaces are enabled:

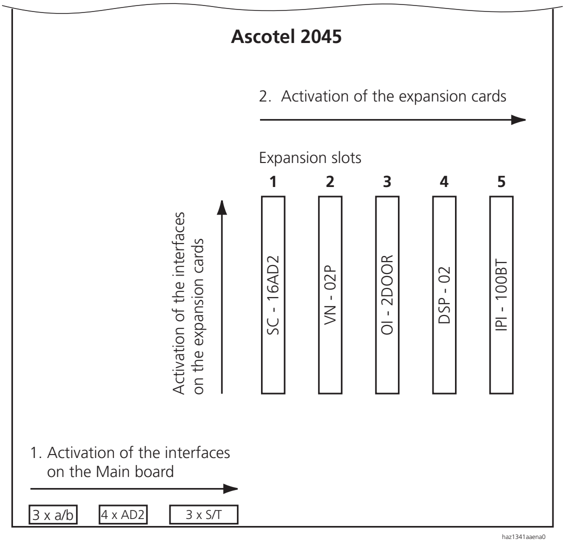


Fig. 3.4: Activation of the interfaces using the example of component placement on Ascotel IntelliGate 2045

Tab. 3.16: Example of component placement with Ascotel IntelliGate 2065

Slot	Expansion card	Network interfaces	User-network interfaces	Note
2065 basic system	(MBL)	-	-	
Expansion slot 1	SC-32AD2	-	32	
Expansion slot 2	SC-32AD2	-	32	
Expansion slot 3	SC-32AD2	-	32	
Expansion slot 4	SC-32AD2	-	32	
Expansion slot 5	-	-	-	unassigned
Expansion slot 6	SC-32AD2	-	32	32 interfaces for DECT radio stations
Expansion slot 7	SC-08AD2	-	8	
Slot 8	-	-	-	Processor card
Expansion slot 9	DSP-02	-	-	18 DECT channels
Expansion slot 10	VM-04P	-	4	
Expansion slot 11	AIP -6350	-	16	Max. 12 calls possible simultaneously

Slot	Expansion card	Network interfaces	User-network interfaces	Note
Expansion slot 12	AIP -6400	1	-	
Expansion slot 13	OI-2DOOR	-	-	
Expansion slot 14	ISDN-04ST	-	4	All S/T interfaces are connected as S interfaces
Expansion slot 15	ISDN-02PRA	2	-	
	Total	3 (of max. 64)	192 (of max. 320)	max. 400 subscribers with their own number

3 System and Expansion Limits

3.1 System Limits

Tab. 3.17: General System Limits

Max. number ...	Ascotel Intelli-Gate 2025	Ascotel Intelli-Gate 2045	Ascotel Intelli-Gate 2065
Expansion slots	5	5	14
Subscriber with own number[1)]	30	60	400
Virtual subscribers	30	40	250
Simultaneous connections [2)]	120	120	120
Simultaneous DECT to non-DECT connections	16	36	72
VoIP / QSIG over IP channels	1 x 12	2 x 12	4 x 12
Trunk groups	11	25	112
Network interfaces per trunk group	8	8	64
Routes	24	24	100
Trunk groups in route	8	8	8
Direct Dialling Plans	10	10	10
Total DDI numbers	500	500	4000
Call Distribution Elements	500	500	4000
User Groups	21	21	99
Subscribers per user group with global call distribution	16	16	16
Subscribers per user group without global call distribution	30	60	400
Abbreviated dialling numbers + PISN subscribers	1500	1500	4000
Line keys per key telephone	39	39	39
Line Keys	500	500	2000
External digit barring	8	8	16
Internal digit barring	8	8	16
Announcement / message groups	8	8	16
Data service tables	8	8	32
Call data memory internal (number of records)	300	300	1000
First-party CTI interfaces	12	36	160
Third-party CTI interfaces	1	1	1
Simultaneous third-party CTI interfaces users[3)]	30	60	400
Simultaneous ACD agents[4)]	10	30	150
Mailboxes	128	128	128
Total charge counters (subscribers, network interfaces, cost centres)	210	238	1428

Max. number ...	Ascotel IntelliGate 2025	Ascotel IntelliGate 2045	Ascotel IntelliGate 2065
Phonebook entries	8000	8000	8000
Call list entries	8000	8000	16000
Freely configurable keys	4000	4000	4000

1) Incl. pager, DECT subscribers, virtual subscribers, etc.
2) Corded subscribers to corded subscribers
3) With CTI Basic or CTI Standard licence
4) With CTI Professional licence

Tab. 3.18: User-network and network interfaces

Max. number ...	Ascotel IntelliGate 2025	Ascotel IntelliGate 2045	Ascotel IntelliGate 2065
Total user-network interfaces (AD2, a/b, S)	16	40	320
AD2 user-network interfaces	12	36	320
Analogue user-network interfaces (a/b) DTMF / PD (incl. voice mail ports)	12	23	168/56[1)]
S user-network interfaces	7	11	64
Network interfaces, total (a/b, T, T2, S external)	8	19	64
Analogue network interfaces (a/b) DTMF / PD	4	8	32/16[2)]
Basic accesses, total (T, S external)	7	11	64
Total primary rate accesses (T2, AIP 6400)	2	4	16
Interfaces S, total of S external + T	7	11	64
IP interfaces with up to AIP 6400 or AIP 6350	1	2	4

1) For performance reasons only 56 terminals with pulse dialling are permitted.
2) For performance reasons only 16 pulse-dialling interfaces are permitted.

3.2 Licence-related System and Expansion Limits

Certain system and expansion limits are scalable through licence acquisition. The following licences are available for Ascotel IntelliGate systems:

- Ascotel 2025 → 2045 system licence upgrade
  This licence upgrades an Ascotel IntelliGate 2025 to an 2045 by increasing the system limits. For both systems all the slots are available. Ascotel IntelliGate 2025 does not require a licence.

- QSIG licences
  These licences are used to implement a private leased-line network with QSIG by enabling a specific number of simultaneously outgoing QSIG B channels. A QSIG licence is not required for the QSIG over IP network.
- CTI licences
  These licences are used to enable the CTI features on the third-party CTI interface of the Ascotel IntelliGate. This means that a specific number of users (CTI clients) are able to monitor and control system terminals via a CTI application. No licence is required for the first-party CTI interface (see also System Manual Application Interfaces). Three licence levels are available:
  – CTI basic licence:
    The basic licence enables the CTI basic functions (e.g. for using a PC dial help) for all internal subscribers. The licence can be combined with CTI standard licences.
  – CTI standard licence:
    The standard licence authorises the use of a standard CTI application. The number of CTI Clients required is to be obtained accordingly. Different levels are available for this licence, and can be cumulated.
  – CTI Professional licence:
    This licence expands the functional scope of a specific number of existing Standard CTI clients by adding ACD features (ACD Queue, Login / Logout, Emergency Routing, etc.). Different levels are available for this licence, and can be cumulated. This requires a corresponding number of CTI Standard clients.

  For more detailed information on the CTI licences, please refer to the Application Interfaces System Manual.
- ATAS licence:
  This licence expands the possibilities for connecting external alarm and messaging sources to the PBX by providing the V.24 and Ethernet interface in addition to the Pocket Adapter. The ATAS protocol can also be used; compared with ATPC1 it provides additional possibilities for display on the system terminals (Fox menu) and allows an alarm to be triggered using the Redkey.
- Advanced Messaging licence:
  Enables the SMPP protocol to be used for integrating an SMS server and 9d handsets to be logged on as system terminals (Ascom Wireless Solution products). User-friendly messaging systems can then be implemented with Ascotel IntelliGate.

- AMI licence:
  For the implementation of very large DECT systems Ascotel IntelliGate 2065 can be connected to the DECT radio system DCT 1800 (Ascom Wireless Solutions product). The Ascotel Mobility Interface (AMI) is available for this purpose. Activation of the AMI functionality is subject to a licence.
- Office 1600IP licence:
  A licence is required to operate the IP-based PC phone Office 1600IP. One Office 1600IP client is enabled with each licence.
- Trial licence (Office 1600IP and CTI):
  This licence temporarily enables CTI clients and Office 1600IP clients. The number of clients is determined solely by the system limits. The licence is valid for 30 days once it has been generated on the licence server.

Tab. 3.19: Overview of licences

Licence	Licensed attributes	Without licence	With licence
System Upgrade 2025 → 2045	System limits	see "System Limits", page 592	
QSIG	QSIG B channels	0	2, 4, 8 or max. (up to the system limits)[1]
CTI Basic	CTI Clients with basic functions (e. g. use of a PC Dial Help)	0	all internal subscribers
CTI-Standard[2]	CTI Clients on standard CTI application	0	+5 +10 +20 +30 +100[3] (up to the system limits)
CTI Professional[4]	CTI Clients on ACD application (ACD queue, Login, Logout, etc.)	0	+10 +20 +30 +100[3] (up to the system limits)
ATAS	ATAS protocol; Ethernet and V.24 interface for connecting external alarm and messaging sources	unavailable	enabled
Advanced Messaging	SMPP protocol for integration of an SMS server and registration of 9d handsets as system terminals.	unavailable	enabled
AMI	Functionality of the Ascotel Mobility Interface for connecting the DECT system DCT 1800.	unavailable	enabled
Office 1600IP	Office 1600IP-Clients	0	1 additional licence per client
Trial (Office 1600IP and CTI)	CTI clients and Office 1600IP clients	0	all internal subscribers for a period of 30 days

1) Licence upgrades possible

2) Combinable with the CTI Basic licence

3) Licences can be cumulated

4) Upgrade to CTI standard licence

All the licences are offered in separate licence packages. Depending on the sales channels the packages may differ from the licences in Tab. 3.19.

The systems are factory supplied unlicensed. Back-licensing is not provided for. However, resetting to the factory setting is possible.

3.3 Terminals

Tab. 3.20: Maximum number of terminals per system and interface

<table>
<tr><th>Inter-face</th><th>Terminal type</th><th>System Terminals</th><th>2025 per system</th><th>2045 per system</th><th>2065 per system</th><th>per interface</th></tr>
<tr><td>Miscel-laneous</td><td colspan="2">Total terminals corded (incl. DECT RU)</td><td>30</td><td>60</td><td>400</td><td></td></tr>
<tr><td rowspan="5">AD2</td><td colspan="2">Terminals on AD2 (incl. DECT RU)</td><td>24</td><td>60</td><td>400</td><td></td></tr>
<tr><td>Featurephones / system terminals</td><td>Office 25, Office 35, Office 45, Office 10, Office 20, Office 30, Office 40</td><td>24</td><td>60</td><td>400</td><td>2</td></tr>
<tr><td>Operator Consoles</td><td>Office 45</td><td>3</td><td>6</td><td>12</td><td>2</td></tr>
<tr><td>V.24 interface</td><td>Office Pocket Adapter</td><td>12</td><td>36</td><td>160</td><td>1</td></tr>
<tr><td>Cordless System</td><td>Ascotel DECT Radio unit[1)]</td><td>4</td><td>32[2)]</td><td>64</td><td>1</td></tr>
<tr><td>DECT</td><td>Handsets</td><td>Office 100
Office 135
Office 150
Office 155pro</td><td>20</td><td>40</td><td>150</td><td></td></tr>
<tr><td>LAN</td><td>IP system handsets</td><td>Office 35IP,
Office 1600IP</td><td>16</td><td>60</td><td>128</td><td>32</td></tr>
<tr><td rowspan="3">S</td><td colspan="2">Terminals on S interfaces (total)</td><td>30</td><td>60</td><td>250</td><td>8[3)]</td></tr>
<tr><td>PC Operator Console</td><td>Office 1550</td><td>3</td><td>6</td><td>12</td><td>2</td></tr>
<tr><td>Paging system</td><td>teleCOURIER pager</td><td>30</td><td>60</td><td>400</td><td></td></tr>
</table>

Inter-face	Terminal type	System Terminals	2025 per system	2045 per system	2065 per system	per interface
	Terminals as per ETSI standard • ISDN terminals • ISDN PC cards • ISDN LAN routers • ISDN Terminal Adapters		30	60	250	8
a/b	Terminals on a/b interfaces (total)		12	23	168[4)]	1
	Analogue, nationally approved terminals • Pulse dialling (PUL) • Frequency dialling (DTMF) • Group 3 fax machines • Answering machines • Modems		12	23	168 [5)]	1

1) DSP cards required.

2) Maximum of 18 SB-8 radio units for operation on two AD2 interfaces in each case.

3) Maximum of 2 call connections possible simultaneously.

4) The Ascotel Voice Mail System occupies 2 or 4 a/b interfaces.

5) For performance reasons only 56 terminals with pulse dialling are permitted.

In addition to the general system limits and limits per user-network interface the maximum number of terminals connected to the system can also be limited by the supply power available to the system.

With the AIMS Project Manager all these factors are automatically taken into account, without any need for manual calculations. Manual calculation itself is subject to the following principles:

Power supply-related limits per system

The number of permissible terminals per system depends on the power requirements of the individual terminals: The total power requirements of all connected terminals must not exceed the power output of the power supply. The average power requirements of terminals as indicated in Tab. 3.22 are used for the calculations.

Tab. 3.21: Power output of the power supply (-40 VDC)

	2025 system	2045 system	2065 system
Available power output	28 Watt	28 Watt	70 Watt

Note:
The power available to supply the terminals can be increased to 250 W by using an external Auxiliary Terminal Power Supply (ATPS).

The table below shows the average power requirements of the terminals for a line resistance of 60 Ω (corresponds to approx. 330m at 0.5mm).

Tab. 3.22: Average power requirements of terminals

Terminals[1)]	Connection	Output P [mW] from PBX
Office 45pro[2)]	AD2 interface	< 10
Office 45	AD2 interface	560
Office 35	AD2 interface	170
Office 25	AD2 interface	290
Office 40	AD2 interface	390
Office 30	AD2 interface	240
Office 20	AD2 interface	170
Office 10	AD2 interface	270
Additional keypad (ZTF)	Office 30, Office 40	70
Expansion keypad (EKP)	Office 35, Office 45	70
Alpha keyboard (AKB)	Office 35, Office 45	20
Pocket Adapter	AD2 interface	310
Radio unit without power supply unitSB-4	AD2 interface	1400
Radio unit with power supply unitSB-4	AD2 interface	620
Radio unit without power supply unitSB-8	1 or 2 AD2 interfaces	1000
Radio unit with power supply unitSB-8	1 or 2 AD2 interfaces	< 10
ISDN terminal	S interface	approx. 450
Analogue terminals	a/b interface	approx. 400

1) The traffic volume assumption for Office terminals is: 0.38 Erlang (+ 0.02 Erlang ring), for the SB-4 radio unit: 2 Erlang (2 channels) and for the SB-8 radio unit: 4 Erlang (4 channels)

2) Power requirements from the PBX, when the terminal is provided power via a plug-in power supply unit (required for illuminating the display).

Memory-related limits per system

The maximum number of function keys, entries in private phone books and abbreviated dialling lists depends on the number and type of terminals. Memory management is dynamic. The maximum values can be found in the chart below.

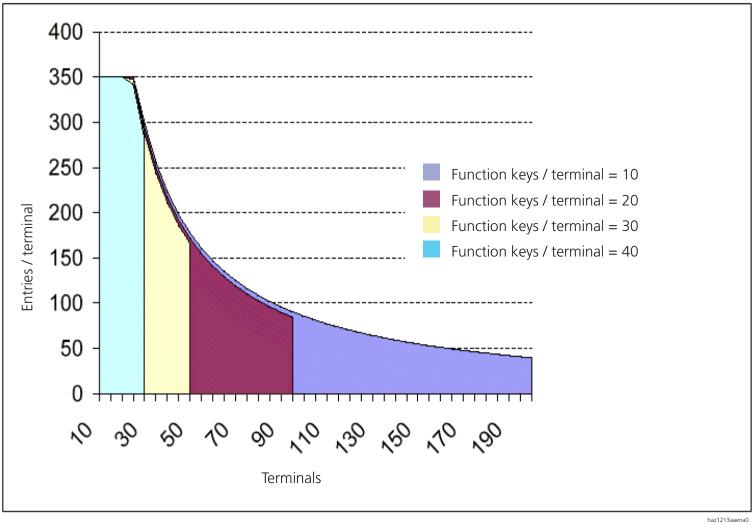


Fig. 3.5: Correlation between function keys and entries per terminal

Power supply-related limit per user-network interface

The power supply-related limit per user-network interface depends on the following variables:

- Terminals used
- Bus configuration
- Line length and conductor cross-section

For information on the calculations refer to "User-Network Interfaces", page 766.

4 Planning a PBX

4.1 Planning-related information

The professional planning of a PBX by a supplier requires extensive knowledge of the customer's needs and requirements. At the same time the customer must be familiarized with the vast array of possibilities available with the different systems so that customer ideas and wishes can be incorporated at a technical level.

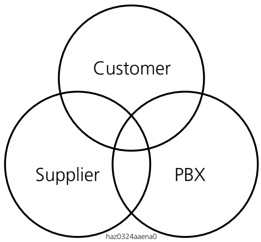


Fig. 3.6: Customer - Supplier - PBX relationship triangle

With existing customers the aim is usually to get to know their experience with the previous system to be able to suggest expansion options designed to solve current problems.

With new customers it is important first of all to find out in a general way precisely what they have in mind, their ideas and experience with communication systems. This also includes getting to know the infrastructure, process flows, organization, size and nature of the business operations.

Because the environment, nature and requirements of customers differ so much, it is not possible to offer any ready-made solutions. The following instructions therefore only provide general guidelines and recommendations and are intended essentially as a checklist.

Planning a system in several stages has proved particularly expedient.

Stage 1

Determining the size of the business and the nature of its activities, and ascertaining the customer's knowledge and requirements. This procedure is used to determine the scope of the system to be supplied and to draw up a rough guideline quotation.

Stage 2

Precision planning and defining detailed questions, expansion and option needs. This process is used to specify the operative scope of functions such as authorizations, direct dialling, organizational structure, definition of terminals, etc.

4.2 Project Manager

The Project Manager (PM) uses the number of terminals and network interfaces required to calculate and optimize the type of system, the expansion cards and the cables for the planned system (see PM User’s Guide in the AIMS Information Manager). In each case it ascertains the most cost-effective system.

The hardware is then correctly assembled, taking into account not only the compatibility between components but also memory restrictions, performance considerations and the power supply load. The results are documented in the form of a price and parts list, a block diagram and a system chart.

Finally the available information is used to generate the system's initial data:

- A subscriber directory, in which the terminals are allocated to the corresponding user-network interfaces, a numbering plan is created, and a set of basic authorizations is allocated to the subscribers based on the terminal types
- A list of network interfaces
- An abbreviated dialling directory

All three tables can be edited and then read in using AIMS and edited further.

4.3 Drawing up a rough guideline quotation (phase 1)

To draw up a guideline quotation all you have to do is enter the customer-related information into the masks of the Project Manager.

The following information can be entered and displayed in the Project Manager's masks:

- Administrative information about the customer
- Expansion parameters required
- Graphical representation of the planned system
- Overview of the network and user-network interfaces connected
- Unoccupied expansion slots and interfaces
- Overview of the components fitted to the planned system
- Overview of all the calculated items complete with prices

Note:
When drawing up a guideline quotation check whether the customer has other wishes or requirements that have not yet been taken into account (e.g. signalling options such as general bell, existing paging system (PS), remote premises, special workplaces, outside solution linkups, existing building management systems, etc.). The building management system and other services are not covered by the PM.

4.4 Definitive quotation (phase 2)

Once the guideline quotation is established, you can begin with the detailed planning of the system. Fine planning involves discussing with the customer the system's operational functions based on the specifications of the guideline quotation. Depending on the conceptual formulation the values are adapted once more in the Project Manager (PM). The system's definitive configuration is carried out at a later stage using the Configuration Manager in AIMS.

Note:
The topics listed in this chapter are intended as an aid (checklist) and are not exhaustive. The complete description of the scope of performance of Ascotel IntelliGate systems can be found in Part 2 of the System Manual.

4.5 Important planning information

4.5.1 Clock synchronization

The clock frequency of a PBX is provided (synchronized) by the public network via the basic accesses T and the primary rate accesses T2.

Should synchronization by the public network fail (due, for example, to exchange line interruptions), the PBX will use its own clock. This frequency deviates at most by 5 ppm from the nominal value, which ensures that the Ascotel DECT system also remains available.

In a private leased-line network, PINXs that are synchronized by the public network pass on the clock reference to PINXs that are not connected directly to the public network.

Synchronization in the private fixed network has to be carefully planned to ensure there are no synchronization loops (see "Synchronization", page 669).

All the private leased-line network connections and public exchange line circuits are automatically in a shared clock reference table when the PBX is configured for the first time.

If a PBX is not networked in a PISN, the clock reference table can be left as it is; only the initial reference may have to be assigned differently.

4.5.2 Emergency operation

In the event of a PBX failure in the PISN, telephony operations can be maintained by configuring alternative paths in all the neighbouring PINXs (see "Reliability aspects", page 667).

In the event of a 230 V mains failure, the uninterruptible power system – if fitted – takes over the power supply to the PBX (see "Uninterruptible power supply (UPS)", page 725).

4.5.3 Periodic Reactivation of Layer 2 on the T-Interface[1)]

Layer 2 of the T network interface can be reactivated periodically every three minutes so that incoming calls are not rejected already at the local exchange after potential temporary interruptions in the U-interface. To do so, configure the parameter "L2 reactivation" of the T network interface to "special".

Note:
In some countries T network interfaces are deactivated once a certain amount of time has elapsed without traffic, and are only reactivated when the PBX once again requests a connection.

4.5.4 Attenuation on analogue network interfaces

With analogue network connections you have a choice of four different attenuation settings:

- "Long" or
- "Long D" for long lines
- "Short" or
- "Short D" for short lines

On lines with a loop resistance of < 280 W, "Short" or "Short D" should be selected to avoid problems with echo or instability (feedback).

The "... D" settings are used to increase the volume in an "analogue exchange – digital subscriber" connection type by 3 dB in both directions as this type of connection is generally perceived as too quiet. The reference level is modified accordingly on the TC-04AB expansion card. Due to the restriction to the aforementioned connection type, the "... D" setting does not result in an increase if an analogue user-network interface is involved in a connection.

Restriction:
The "... D" setting should not be used (or only once the stability conditions have been thoroughly clarified) if the equipment (Terminal Adapter) operated on digital interfaces also features a four-wire to two-wire conversion, i.e. an analogue two-wire interface.

[1)] Only in Germany and Austria.

Initialization setting

Analogue network interfaces are set to "Long D".

See also:
"Analogue Network Interfaces", page 67

4.5.5 Voice and data terminals on the S interface

Both voice and data terminals can be connected to the same S interface. When designing the system, bear in mind that data terminals can also take up user information channels. ISDN routers and ISDN PC cards that support channel bundling can take up both user information channels.

In mixed operation the availability of the terminals has to be taken into account.

One call or data connection can be set up on each bearer channel (B channel) simultaneously and independently of the other B channel.

4.5.6 Down-circuit connection of a PBX with corded terminals

A PBX can be connected down-circuit from a large PBX as a key telephone or team system.[1)] All the features of the Ascotel remain available.

4.5.6.1 Analogue down-circuit connection

With an analogue down-circuit connection the features of the up-circuit PBX can also be utilized.

This results in the following special applications for the subscriber:

- Depending on the system configuration the subscriber makes phone calls in a complex PBX environment. The subscriber's disposal is a large number of features at two levels (subscriber's own system and the up-circuit system). A short induction course helps subscribers to familiarize themselves quickly with the PBX environment.

1) For this, the interface card TC-04AB (4 analogue network interfaces) has to be fitted.

- Practically all PBX types that can be used as an up-circuit PBX also feature the DTMF dialling method on the analogue subscriber line, in addition to pulse dialling. To prevent faulty dialling, the DTMF method should be used whenever possible.
- If the up-circuit PBX requires that subscribers wait for the exchange-free tone, all the entered abbreviated dialling numbers must be provided with a hyphen "-" (interdigit pause) after the digits for exchange access. At this point the PBX will again pause for the tone when dialling.

Example: Exchange access via exchange access prefix

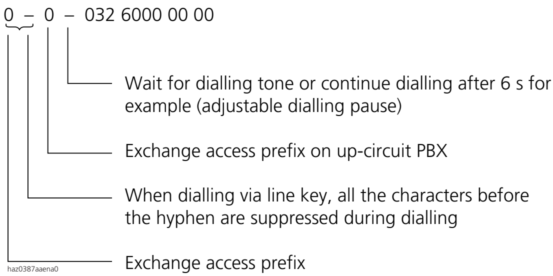


Fig. 3.7: Example of the exchange access prefix via up-circuit PBX

The following configuration steps are necessary:

1. The exchange access prefix of the up-circuit PBX must be entered in the exchange digit barring.
2. The corresponding analogue trunk lines are configured to "Down-circuit from the PBX ". Consequence:
   – Digit barring is switched off in general. The digit barring of the up-circuit PBX has to be used.
   – Incoming calls are forwarded transparently to the subscriber.
3. The corresponding analogue trunk lines are to be configured to the correct "dialling type".
   – If the up-circuit PBX provides DTMF and pulse dialling for internal subscriber
   → configure trunk lines to DTMF.
   – If the up-circuit PBX only provides pulse dialling for internal subscribers
   → configure trunk lines to pulse dialling; the system earth must also be installed for the earth key criterion.

Example: Enquiry call behind PBX

This feature is described in Part 2 under "Hold (enquiry call)", page 460. It can be used from both analogue terminals and system terminals.

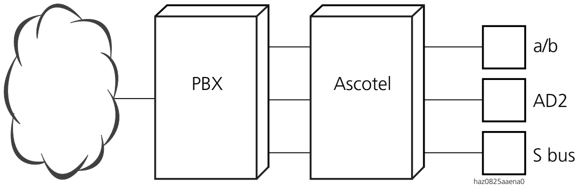


Fig. 3.8: Enquiry call behind PBX

Procedures with different terminals

Situation: The existing call connection of an Ascotel subscriber already seizes a trunk line to the up-circuit PBX. The procedure for setting up an inquiry call depends on the type of terminal:

- Analogue terminal
  - Flash: Dial tone of Ascotel
  - Flash *42: Dial of up-circuit PBX
- OfficeTerminals
  - Enquiry call menu: Dial tone of Ascotel
  - Key with macro ".*42": Dial of up-circuit PBX

Using the exchange's features

To activate features on the public network such as the exchange feature "Call Forwarding" from the system itself, you need to seize a trunk line. The feature can then be entered in accordance with the service provider's operating instructions.

See also:
System configuration:
– Down-circuit from the PBX; Analogue network interfaces
– Dialling type; Analogue network interfaces

4.5.6.2 Digital down-circuit connection of a PBX with QSIG

If a down-circuit Ascotel PBX is connected with an up-circuit PBX via digital lines (T, T2), all the features as per QSIG are available providing the up-circuit system supports the QSIG protocol (see "ISDN services supported by the system", page 424).

The down-circuit PBX is configured in accordance with the rules for networked PBXs.

The up-circuit ISDN PBX has a connection to the public network. It can also be an Ascotel or a third-party product, provided it supports the QSIG protocol.

As a rule the down-circuit PBX is connected with the up-circuit PBX via its own fixed lines. Depending on the PBX type the interfaces can be basic accesses (T) or primary rate accesses (T2). Connections on an S external-type interface are also possible instead of connections on a T interface, providing at least one T interface is available for synchronization via the ISDN network.

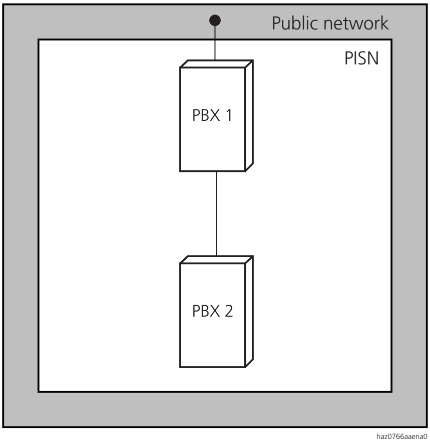


Fig. 3.9: PISN consisting of PBX 1 with a down-circuit PBX 2

4.5.6.3 Down-circuit connection of a PBX with cordless system

A PBX with cordless system can be connected down-circuit from another PBX in a number of ways. As a rule the up-circuit PBX is a larger system.

Digital down-circuit connection with QSIG

On systems with digital lines, common features as per QSIG can be used, provided the up-circuit system supports the QSIG protocol (see also "Digital down-circuit connection of a PBX with QSIG", page 608). This means the corresponding functions of the up-circuit PBX are also available from the cordless system.

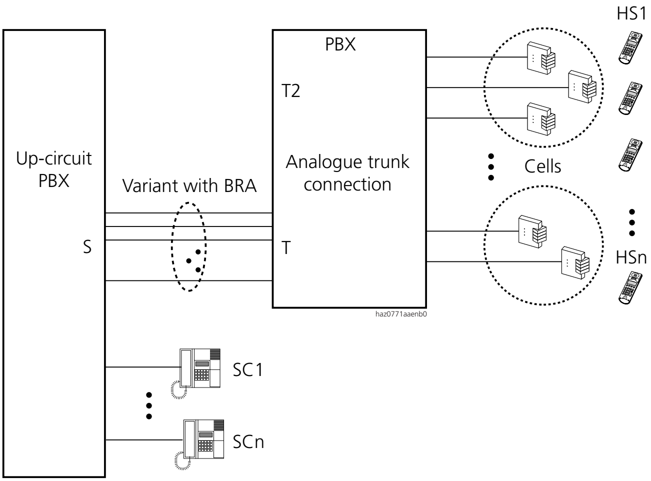


Fig. 3.10: Digital down-circuit connection with QSIG

The number of possible handsets with Ascotel DECT depends on the system limit.

4.5.6.4 Application of Direct Dialling Out (DDO)

If a fax server is connected to an S bus, individual fax receivers allocated a DDI number can be specifically addressed. In terms of routing technology, this corresponding to a DDO (Direct Dialling Out) function.

The fax server forwards the incoming faxes via e-mail to the relevant PC stations that are set up as fax receivers.

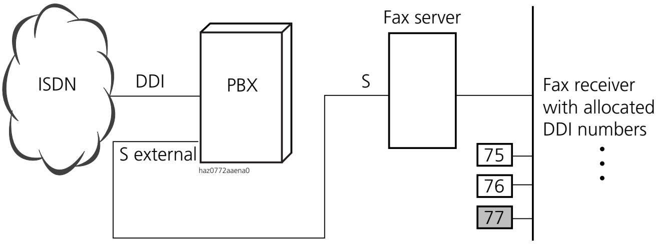


Fig. 3.11: Direct Dialling Out (DDO) to a fax server

Routing via trunk groups

Due to the configuration of the S interface as "EXTERNALS" and the use of the DSS1 protocol, the fax calls can be routed via routes and trunk groups. This means that all fax receivers that have been allocated a DDI number can be reached via a single S interface.

See also:
"Call to a DSS1 Terminal equipment on the S Bus (DDO)", page 302

5 Planning DECT systems

The following sections take you through the planning procedure for an Ascotel DECT system (DECT: Digital Enhanced Cordless Telecommunications). It contains recommendations on the project sequence and on handling the measuring equipment. Unless otherwise specified, these considerations apply to DECT radio units operated with the integrated antenna. The SB-8ANT radio unit with two external antenna connections is provided for special requirements or difficult topographical conditions (see "Using external antennas", page 621). For general information on the Ascotel DECT system and its features, see "Cordless Terminals", page 110.

New planning

A wide range of subjects have to be clarified in depth with the customer before planning a DECT system. DECT systems are very high-performing, which is why they need to be explained in detail. The better customers are informed, the more they will use the system’s features, which in turn means the application will be used more efficiently.

The following points are to be observed in general:

- Radio coverage area, capacity in terms of the number of connections
- Known problem areas and what to do about them
- Handover behaviour and what happens during a handover
- Cordless groups and other features

Project extension

There are different types of extension:

- Additional handsets without a notable increase in traffic
  – only new entries in the PBX numbering plan
- Additional handsets with an increase in traffic
  – entries in the PBX numbering plan
  – Existing radio cells, possibly needing to be reinforced with additional radio units (warning: radio cells become smaller, resulting in a shift in the handover limits)
  – Radio units possibly needing to be distributed to other location areas
- Coverage required over a greater area

System installation

Once determined, the locations must be observed precisely in order to obtain reproducible results. There are always surprising effects in practice, due for example to cells being set up with several radio units (rather than the one cell originally planned and surveyed). If for example masts are used as mounting structures and fitted on both sides with radio units, conditions on either side of the mast will be different.

For precise installation instructions, refer to Part 4.

Re-measuring the installed system

Before the system can be handed over to the customer, the measurements need to be verified and compared with the values of the original measurement.

More detailed information on installation and re-measurement is described in "Measuring Equipment", page 635 and "Measurements", page 656 as well as in, "General Checks", page 866.

Documentation

The system documentation is an essential requirement. This applies to both the planning and the execution phases. To this end all the data is recorded and set out in a final written version. Modifications and updates must also be documented.

This documentation can then be used as reference in the event of project extensions, enquiries or other matters.

5.1 Ascotel DECT and PBX

The Ascotel DECT system allows users to make and receive telephone calls anywhere on the company premises without being tied down by hard-wired sets. Several radio units form a network of coverage ranges within which a user is able to move freely.

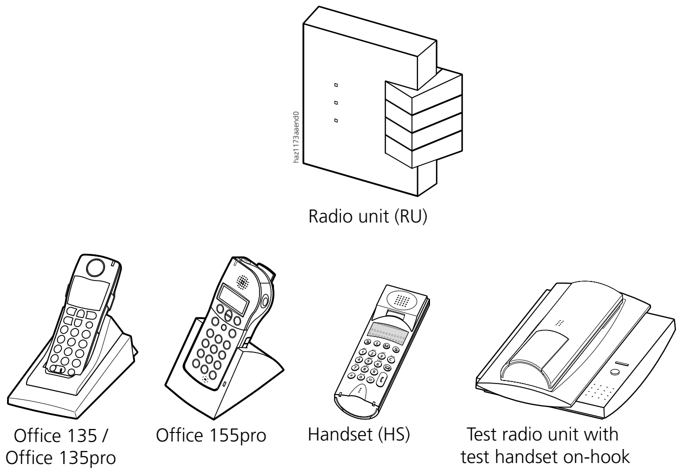


Fig. 3.12: The components of a Ascotel DECT system

The components of an Ascotel DECT system comply with the conditions of the "Digital Enhanced Cordless Telecommunications"standard (DECT). This standard regulates in particular the conditions on the high-frequency radio link and the signalling between handsets and radio units.

Besides system handsets it also possible to operate the installation using products that support the GAP (Generic Access Profile) standard. GAP defines a restricted functionality, thereby also supporting the handsets of other manufacturers. GAP handsets cannot be used for measuring purposes.

5.2 Coverage area

The supply range of a Ascotel DECT system can include many different types of geographic geometry. In most cases the bulk of the supply range is inside buildings.

A DECT system always relates to a PINX (Private Integrated Exchange) in a PISN (Private Integrated Services Network). Handover to a neighbouring DECT system in another PINX never occurs, even if the handset is logged on in both systems.

However, the planning can be carried out for both systems as the measuring equipment is used without PBX.

The following explanations describe the situation with a single DECT system.

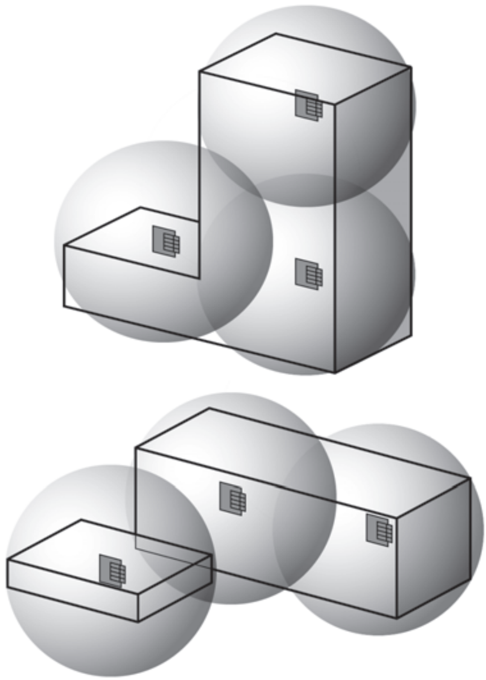


Fig. 3.13: Examples of the positioning of radio units inside buildings

Topology

A radio system supplies a specific area. This area is known as the supply range. In practice home users are able to receive a variety of autonomous DECT systems (e.g. Ascotel DECT, cordless DECT phones, etc.). An Office 100, Office 135, Office 150 or Office 155pro can be logged on in up to four autonomous DECT systems. The supply ranges for these independent DECT radio systems can also overlap.

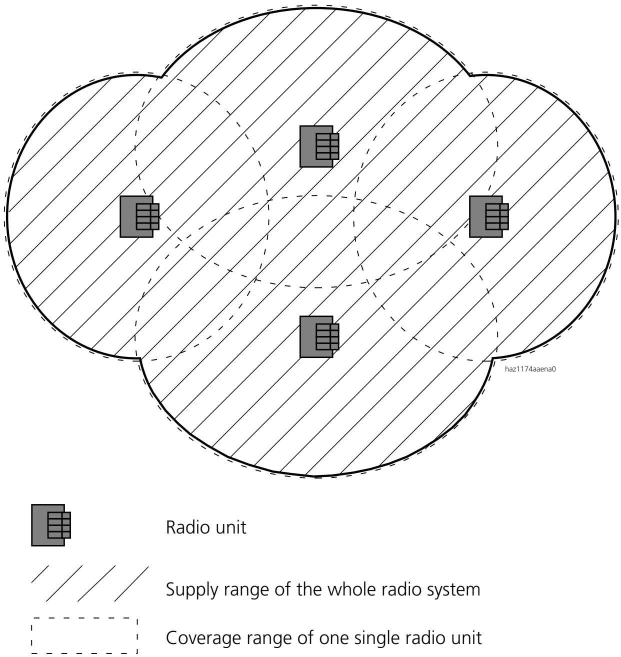


Fig. 3.14: Radio system

Location areas

Location areas are used to divide the radio area into as many as four locally subdivided supply ranges. The system knows the location area in which a handset is located and only needs to send messages to the handset within that location area. Location area partitioning helps to reduce the system load caused by the signalling of incoming calls.

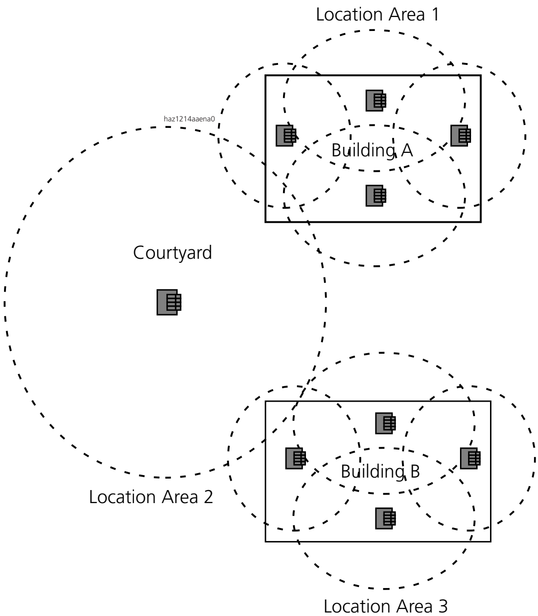


Fig. 3.15: Example of location areas

Distributing radio units among several location areas only makes sense as of a certain system size. The radio units are allocated by means of configuration. For a description of the best way to group the radio units of a system in location areas, see page 631 ff.

5.3 System characteristics

5.3.1 Radio characteristic of a radio unit

The coverage range of a radio unit depends among other things on the antenna through which the radio signal is transmitted. The SB-4 and SB-8 radio units transmit through two integrated antennas inside the housing. The radio characteristic of internal antennas is virtually spherical (with only transmission to the rear slightly diminished), in other words registered handsets are able to move away from the radio unit the same distance in every direction without the radio signal breaking up. This does not take account of the topology that attenuates the signal's propagation.

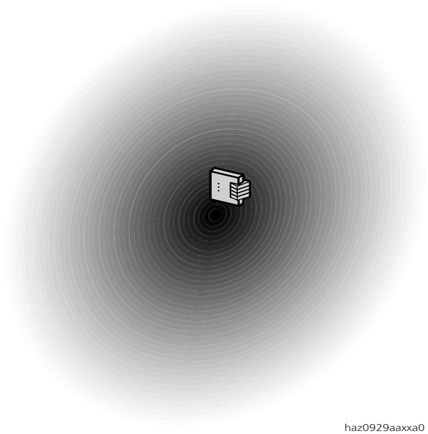


Fig. 3.16: **Spherical radio characteristic of a DECT radio unit**

The radio characteristic within the area to be covered is influenced by the objects and materials located in the buildings. The spherical radio characteristic is therefore deformed accordingly depending on local circumstances.

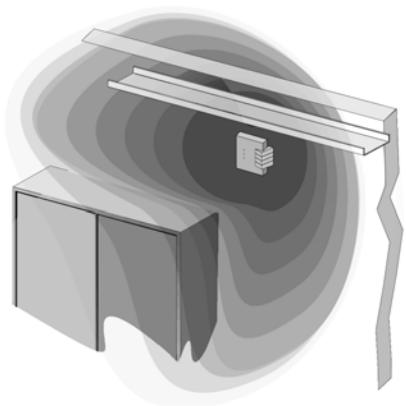


Fig. 3.17: **Radio characteristic through obstacles to radio propagation**

5.3.2 High-frequency propagation conditions

For a better understanding of the explanations on physical conditions given in the following sections, we first need to take a look at the basic principles of high-frequency propagation.

The method under discussion here is DECT standard. It operates in the frequency range of 1880...1900 MHz and provides 120 communication channels. All cordless telephony systems are subject in principle to the following explanations. The experience gained in planning any system will therefore be very useful when it comes to planning Ascotel DECT systems.

Interference factors

The knowledge of potential reasons for interference can raise the project engineer’s awareness to such an extent that many critical points can be avoided already at the design stage of the Ascotel DECT system by using the appropriate measures. In radio technology there are many interference factors that affect mainly the range and quality of the transmission.

In principle we need to differentiate between two types of interference factors:

- Interference by obstacles that attenuate and / or reflect radio propagation, causing dead spots
- Interference due to other radio signals, leading to transmission errors

The receive power of DECT signals can fluctuate a great deal, locally, within only a few centimetres (see Fig. 3.18). This means that signal interference can be reduced or eliminated simply by altering the position.

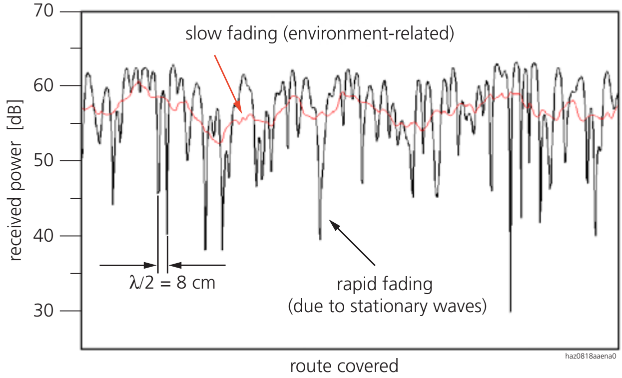


Fig. 3.18: Obstacle-induced attenuation and reflection of DECT signals

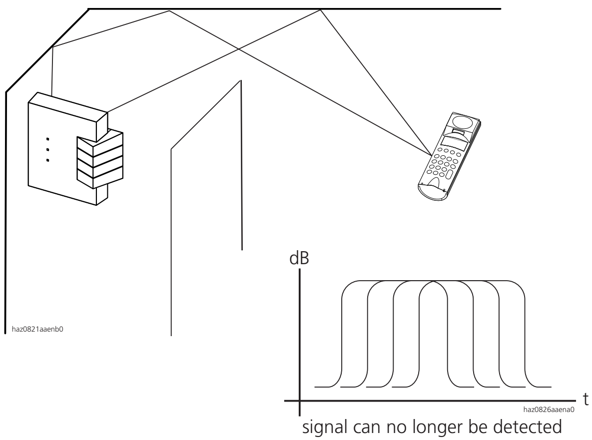


Fig. 3.19: Example of multipath propagation with DECT signals

Obstacles may include:

- Moving metal objects such as lifts, cranes, carriages, escalators, blinds, especially ones that are actuated automatically (the influence of such obstacles varies and is therefore difficult to assess)
- Metal-panelled rooms and large metal-clad objects such as air conditioners, computer rooms, metallized glassed areas (mirrored), fire protection walls, storage tank installations, refrigerating units, boilers

- Building structures and installations such as steel-reinforced concrete ceilings and walls, stairways, long corridors, rising mains, cable ducts
- Room furnishings such as metal shelves, file cabinets

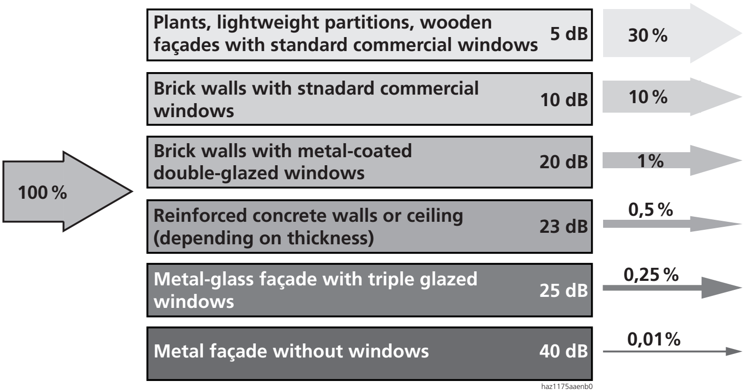


Fig. 3.20: Attenuation of RF signals due to walls, windows, plants

Due to their typical furnishings/fixtures projects in the following environments must be regarded as critical:

- Garages / repair workshops
- Scrap merchants
- Campsites
- Galvanizing plants
- Metal-working industry

Reception conditions

Optimizing range is a fundamental challenge of radio technology. The reception of information in marginal zones is patchy at best. Practical measurements are carried out on the premises to determine the range.

When instructing users, the following instructions can help to achieve optimum results:

- You can usually improve the connection quality through minor changes in location, e.g. by turning your head or your body.

- Avoid making phone calls in unsuitable places, for example in lifts. Users should be made aware of these zones during instruction.

5.3.3 Using external antennas

The SB-8ANT radio unit with 2 external antenna connections is ideally suited for use in difficult topographical conditions and for special coverage range requirements. External antennas are useful

- for rectifying radio signals, thereby achieving a greater range in one particular direction (e.g. to provide coverage to remote ancillary buildings).
- for providing coverage to an outside area without the shell of the building obstructing the propagation of the radio signals (this is achieved by mounting the radio unit within the building and the antennas outside it).

There are different types of antennas, each with highly specific radio characteristics for meeting individual coverage requirements; they are best illustrated by radiation patterns. The first example shows an antenna that radiates evenly horizontally but has a very restricted range vertically. This type is known as an omni-directional gain antenna and improves the horizontal range without increasing the radiation output; it is suitable for open, level premises.

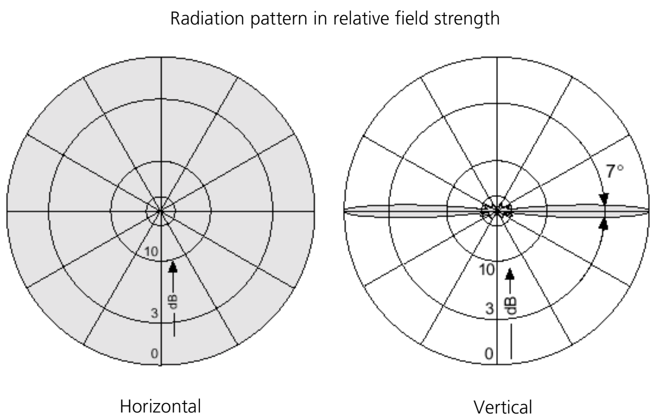


Fig. 3.21: Example 1: Omni-directional gain antenna

The second example of an antenna, which radiates directionally both horizontally **and** vertically, is called a corner reflector antenna. It is ideally suited for covering distant remote buildings or areas.

Radiation pattern in relative field strength

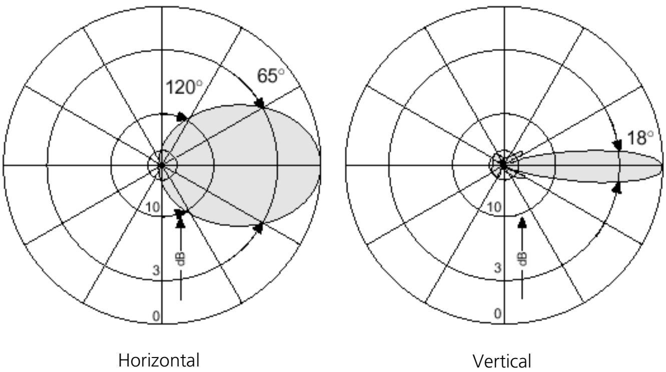


Fig. 3.22: Example 2: Corner reflector antenna

The same principle applies to all antenna types: The narrower the horizontal and vertical radiation range, the greater the distances that can be covered. In this connection we talk of antenna gain, which indicates the ratio of the energy radiated in a particular angle to the radiated energy of a standard antenna (usually a $\lambda/2$ dipole antenna). An extreme example of this is the radio relay antenna, which is used for transmitting radio signals in a targeted direction from one point to the next.

Fading effect

If an SB-8ANT radio unit is operated with external antennas, an antenna must always be connected to each of the two antenna connections. The antennas should have the same radio characteristic and also cover the same range. This reduces the occurrence of fading effects (caused by radio signals cancelling one another out through reflections), and considerably improves the connection quality. If such an effect occurs on one of the antennas, the radio unit automatically switches over to the second antenna, thereby preventing minor drop-outs. This is referred to as antenna diversity. Fading effects occur mainly inside and between buildings, which provide a great deal of reflection surface. Dual antennas inside a single housing, designed specifically with this effect in mind, are widely available in the market. That is why the radio units themselves are always equipped with two internal antennas.

5.3.4 Using repeaters (relays)

Repeaters are autonomous devices equipped with two antennas, which increase the range of the DECT system. They receive the signals from one radio unit over one antenna and send the signals back out again over the second antenna. When positioning a repeater you need to make sure not only that the coverage ranges overlap but also that the repeater is mounted in the coverage range of one radio unit and vice versa. Several repeaters can be allocated to one radio unit. Ring-form (see Fig. 3.23) as well as other arrangements around a radio unit are possible.

Most repeaters operate in "Blind-Slot mode". As they receive and send, only 2 call connections are possible simultaneously on each repeater.

Cascaded repeaters are not supported. However the handover between individual coverage ranges is guaranteed.

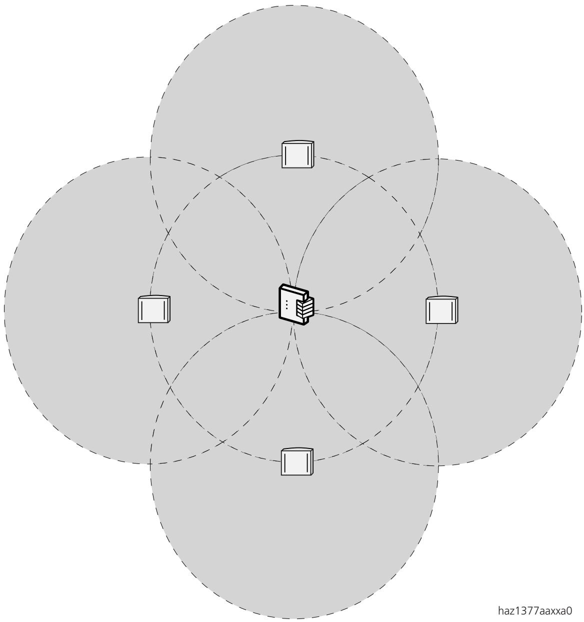


Fig. 3.23: Arrangement of repeaters around a radio unit

Even remote buildings can be covered with a repeater by using external antennas with rectification.

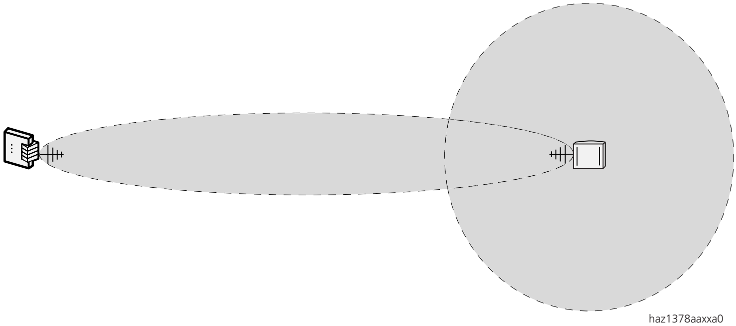


Fig. 3.24: Remote repeater with directional antenna

The use of repeaters is indicated particularly in cases where supply coverage conditions inside buildings are difficult and only a few handsets are being used.

Note:
Most repeaters do not support voice data encryption. For this reason the "Encryption" parameter should normally be configured to "No" in the DECT System Configuration.

5.4 Planning

In practice a regulated procedure for planning has proved sensible and, above all, efficient. A considerable increase in efficiency can be achieved through the consistent use of the aids available.

In the following, the planning procedure is described in the form of a "checklist":

1. Ascertain customer requirements.
2. Roughly determine the locations for the radio units.
3. Measure out the DECT system in situ.
4. Install and re-measure the system.

5.4.1 Ascertaining customer requirements

Since the aim of Ascotel DECT systems is to cover all kinds of different mobility requirements in a non-standardized environment, we need to determine and record precisely what sort of requirements the customer has in mind. Records avoid misunderstandings and can be used as a working paper (e.g. project progress report) or as specifications to be confirmed by the customer.

Important questions:

- Situation: Where are the calls to be made from – outdoors / indoors?
- Premises: What surface area and what height or depth (storeys, basement floors) form part of the supply range? Recommendation: Ask for a floor plan.
- Building structure: What sort of materials and types of construction are the buildings made of? What sort of structural changes are planned for the near future?
- Subscribers: How many handsets are required? What sort of phoning pattern do users have? Recommendation: Allocate in user groups.
- Traffic density: How are the handsets distributed throughout the premises and which users are where?
- Dynamics: How many handsets are expected where and at what times of the day? Recommendation: Take account of special infrastructure areas such as: Canteen: 9:00…10:00, meeting rooms.

5.4.2 Initial rough determination of the radio unit locations

Radio circumstances are difficult to estimate. For this reason, situations regarded as particularly critical need to be determined on site through measurements.

This will provide a reliable idea of the equipment required and the locations for the radio units.

The following rules of thumb may be of help:

- Good connections still possible in a horizontal direction behind 2...3 ordinary brick walls; barely any penetration through concrete floors and ceilings in a vertical direction and in ground floor or basement floors, i.e. each storey must be supplied separately. A certain amount of vertical penetration can be expected from the first floor upwards; generally speaking, radio propagation conditions improve as the distance from the ground increases.
- Openings in obstacles improve radio conditions.
- Subsequent furnishing: In empty buildings, the effect of the absence of furniture, machines, partitions, etc., and their room limitation needs to be taken into account. Subsequent extensions and conversions also have an influence.

- It is important to ensure sufficiently large overlap zones between neighbouring coverage ranges. The signal should not be too weak that it prevents automatic handover to the next radio unit. Here, it is necessary to find a solution between a large number of radio units and a reasonable coverage of the premises.
- Radio range (guideline values)
  – up to 30 m in buildings
  – outdoors up to 250 m.
- Observe the minimum distance between radio units (see "Installing the radio units", page 829).

5.4.3 On-site measuring

As soon as the concrete locations for the radio units have been planned, it is recommended to confirm the circumstances in keeping with the plan using on-site measurements.

A detailed description on how to use the test equipment can be found in "Measurements", page 656.

Remarks:

- Installation site for the test radio unit: To carry out the measurements do not place the radio unit on the ground / floor but position it in the location in which it is to be installed later.
- Measurements must be meticulously carried out (no compromises). The objective is optimum radio coverage.
- Documentation: It is advisable to keep a test log so results can be reproduced later. Record the values measured as well as the supply range on the ground plan, horizontal and vertical.
- The measurements provide a reliable idea of the equipment required and the locations for the radio units.
- Co-operation with the customer: As soon as you are able to make sufficiently binding statements about "problem areas", you should involve the customer for clarification purposes. It is imperative that the customer be informed of any areas where coverage is not optimum.

Optimum positioning of the DECT radio unit

The location of the DECT radio unit is determined by a number of different factors:

- Optimum radio coverage of the environment
- Conditions inside the building
- Installation possibilities
- Presence of supply leads and socket outlets
- Aesthetic aspects and wishes on the part of the customer
- Outside the buildings: weather protection (rain, sun), vandal-proof

DECT radio unit with outdoor supply

The following principles are to be observed when installing DECT radio units on outdoor premises:

- Choose a central position and avoid acute penetration angles
- Ensure the chosen location is protected from the weather
- Make sure the installation site is chosen sufficiently high to be protected from acts of vandalism

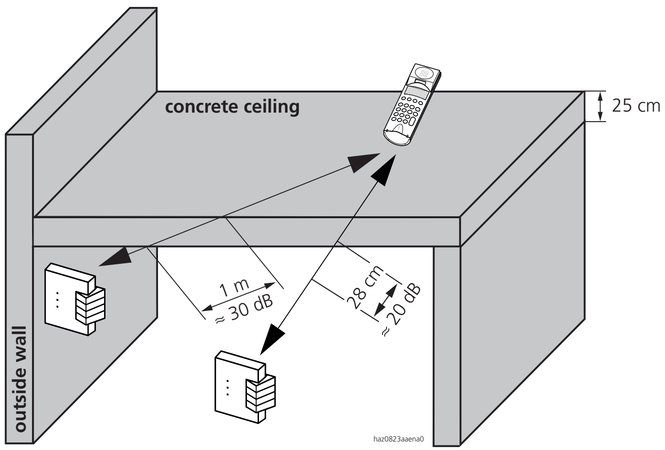


Fig. 3.25: Optimum positioning of the DECT radio unit on outside walls

DECT radio unit with indoor supply

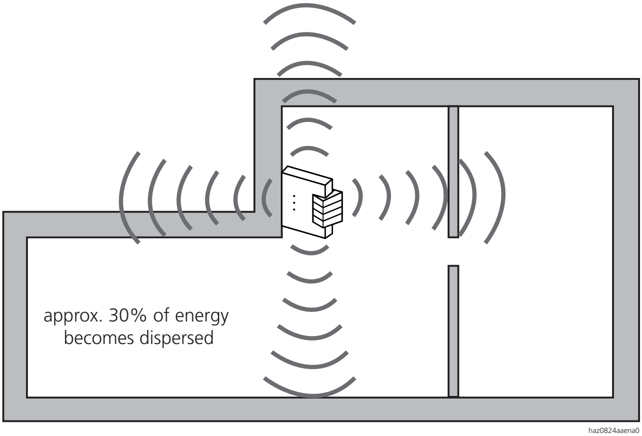


Fig. 3.26: Optimum positioning of the DECT radio unit indoors

The following principles are to be observed when installing DECT radio units indoors:

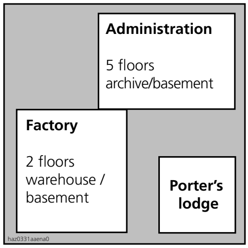
- In indoor supply install the radio units on inner walls rather than outdoor walls.
- Ceiling mounting can also be considered.
- Do not install in the immediate vicinity of cable ducts, metal cabinets and other large metal objects. They obstruct transmission and / or can result in crosstalk. Maintain a distance of > 50 cm!
- Connecting line between PBX and DECT radio unit:
  – Crosstalk can occur here if they are laid in parallel with mains feeder lines inside cable ducts (e.g. engineering workshops). This must be taken into account when choosing the cable and the cable route.
- Fluorescent lighting generates wide-band noise and can affect radio transmission.
- RF interference fields: PCs and other electronic equipment can affect radio transmission in the vicinity either intermittently or permanently.
- Radio units and handsets generate pulsed RF signals. These can affect sensitive electronic equipment (control systems, measuring sensors, sensors, diagnostics equipment on intensive-care wards in hospitals, etc.) permanently or temporarily.

Note:
Take note of safety distances and safety regulations. In areas subject to explosion hazards take note of the relevant rules and regulations.

This important phase of the project is to be explained in greater detail using an example.

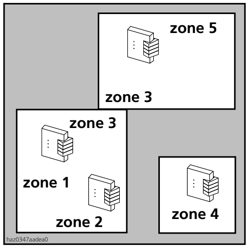
5.4.3.1 Preparing the measurements

Drawing up a situation plan:



- The situation plan records approximately the number and size of the coverage ranges required.
- The measurements provide the ideal locations for the radio units.

Radio supply and zone formation:



- Create a sensible distribution, e.g. several zones in the factory and administration (porter's lodge, basement, etc., in different zones).
- Make a note of the connection capacity required (number of simultaneous calls) for each zone to determine the number of radio units (see also under "Traffic density", page 633).

5.4.3.2 Global coverage in the supply range

First, ensure that the surface area of the planned supply range is satisfactorily covered.

The ideal locations for the radio units are determined using practical measurements.

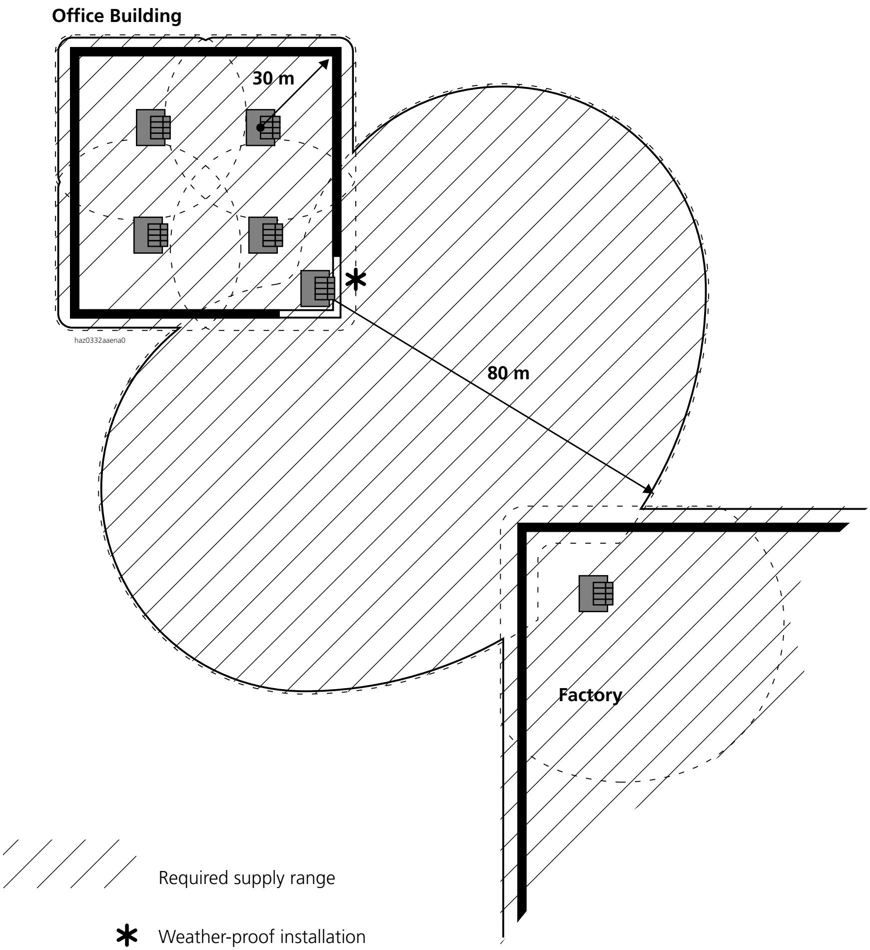
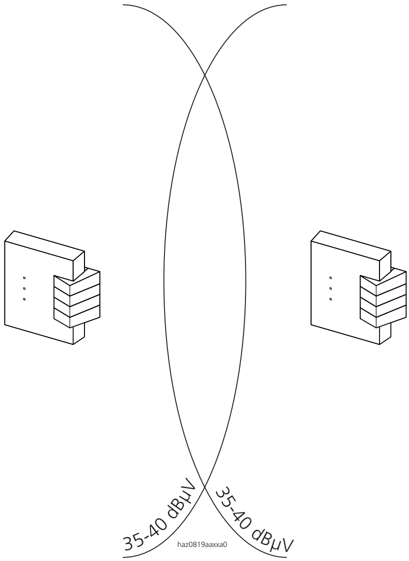


Fig. 3.27: Global coverage in the supply range

5.4.3.3 Handover overlap areas



- To guarantee a clean handover, the radio units should be fitted in such a way that the 35–40 dBµV limits are at least in contact.
- When conducting survey measurements in unfinished buildings, the limits should be increased by 10 dBµV.

Fig. 3.28: Planning handover overlap areas

How to evaluate the field strength measurement values and specify the overlap areas is described in "Measurements", page 656.

5.4.3.4 Location areas

With incoming calls each handset is called by the system simultaneously via all the radio units of the location area in which the handset is logged on. If a handset switches from the coverage range of one location to that of another, the handset will log into the new area from scratch. The logon procedure is automatic and does not affect the use of the handset. To prevent handsets from constantly logging themselves on in locations areas, plan the overlap areas of location areas in such a way that handsets are not permanently located there.

Distributing the radio units over several location areas ensures that the radio traffic involved in locating handsets for incoming calls is distributed across the location areas. This means that overall more calls can be processed simultaneously.

The optimum number of location areas depends on the number of handsets in the system and the traffic density (see page 633). The diagram below shows the minimum number of location areas to be created out of the maximum of four areas.

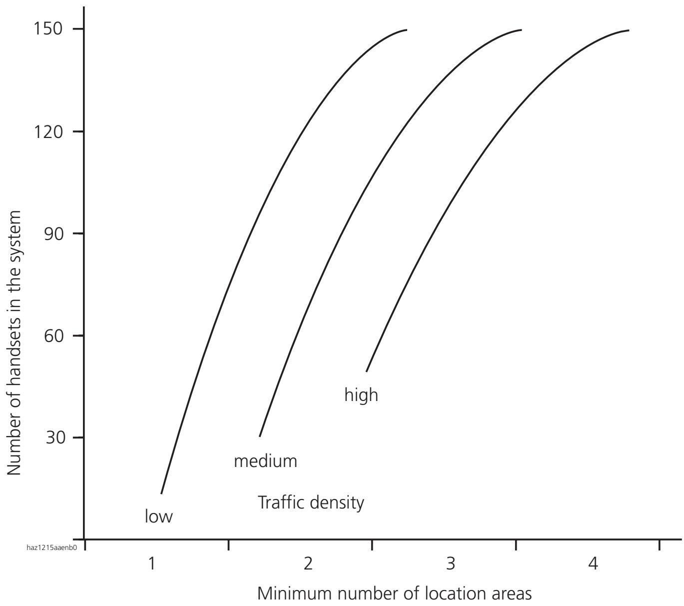


Fig. 3.29: Minimum number of location areas for different traffic densities

The following points need to be taken into account when dividing up radio units into location areas:

- As of approx. 60 handsets the radio units in the system have to be divided up into several location areas.
- If more than 9 incoming calls are to be signalled simultaneously on the handsets, the radio units of a system need to be spread over several location areas. No more than 9 calls should be made simultaneously in each location area.
- The radio units of any given location area should cover a contiguous area (see example on page 616).
- Handsets should be used mainly within the same location area.
- Handsets should not be continually used in the overlap zone between two location areas.

5.4.3.5 Traffic density

Once the supply range has been located and covered, the locations for the radio units are known.

The second phase now looks at other requirements: The traffic volume, determined by the number of handsets and the frequency and duration of connections in an area defined by the customer. In most cases such a zone does not match the coverage range of a radio unit.

For each zone you now need to determine how many radio units are currently being used to supply them. It may be necessary to boost the coverage with additional radio units for insufficiently supplied areas (in accordance with the customer's requirements). To do so we need to refer back to the radio unit locations that were determined for the area coverage. For instance, it is a good idea to place additional radio units at the centre of a zone with a higher traffic volume, even if those units end up between two already positioned radio units.

To determine the traffic volume, we assume three typical ranges: "low", "medium", "high". "High" means that approx. 50% of all handsets are making calls simultaneously. It is also important to note that the volume of traffic can vary considerably in the course of a day. The question is always whether or not to cover a traffic volume that may be higher only briefly in a particular zone.

The diagram below provides an overview of the recommended number of radio units that cover the zone in relation to the handsets in that zone. Radio units have a capacity of 4 channels (SB-4) and 8 channels (SB-8) respectively.

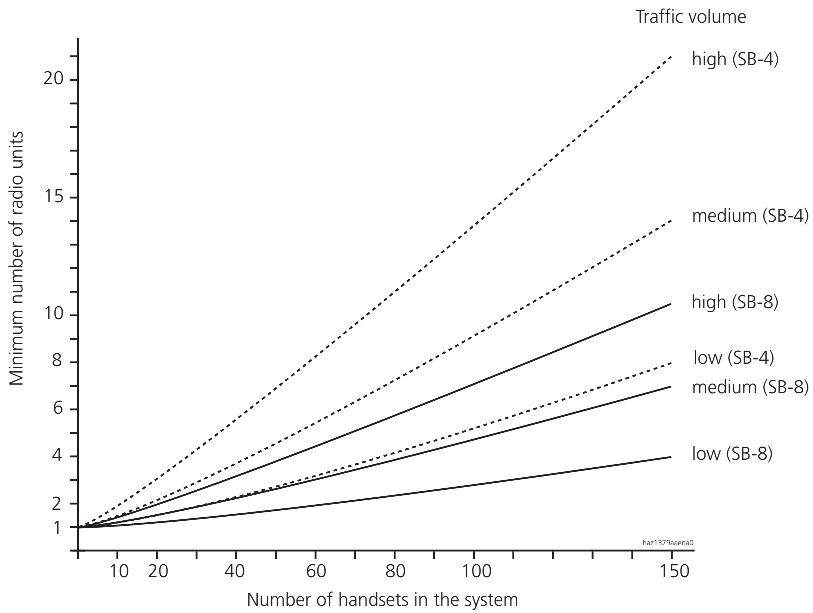


Fig. 3.30: Minimum number of radio units per zone in relation to the number of handsets and the traffic volume.

If the coverage ranges of radio units overlap, the resources of the neighbouring radio units in the shared area (zone) are cumulated.

Each radio unit is connected to an Ascotel DECT-compatible AD2 interface via one or two AD2 lines (prerequisite: at least one DSP card fitted). The maximum number of radio units per system is specified in Tab. 3.20.

The potential locations are determined on the situation plan and then definitively agreed on-site through the use of measurements.

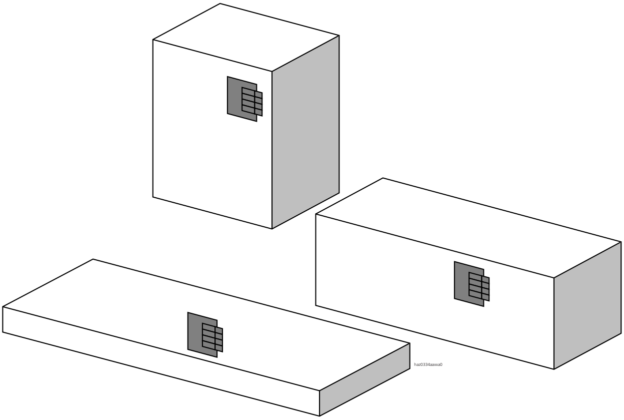


Fig. 3.31: **Determining locations**

5.4.4 Installing and re-measuring the definitive system

For more detailed information, please refer to Part 4.

As soon as the definitive system is installed, test measurements should be carried out in handover overlap border areas and compared with the results of the planning (see "On-site measuring", page 626). Critical areas should be discussed with the customer.

5.5 Measuring Equipment

The test kit contains the following equipment:

- Two test radio units with integrated charging bays for the handsets
- 2 test handsets for the test radio units
- 2 standard handsets (with Foxkeys)
- 1 set of Operating Instructions - Planning
- 1 set of Operating Instructions for the standard handset
- 1 set of Operating Instructions for the test handset
- 40 batteries
- 2 charging units
- 2 battery boxes for the radio units
- Miscellaneous cables

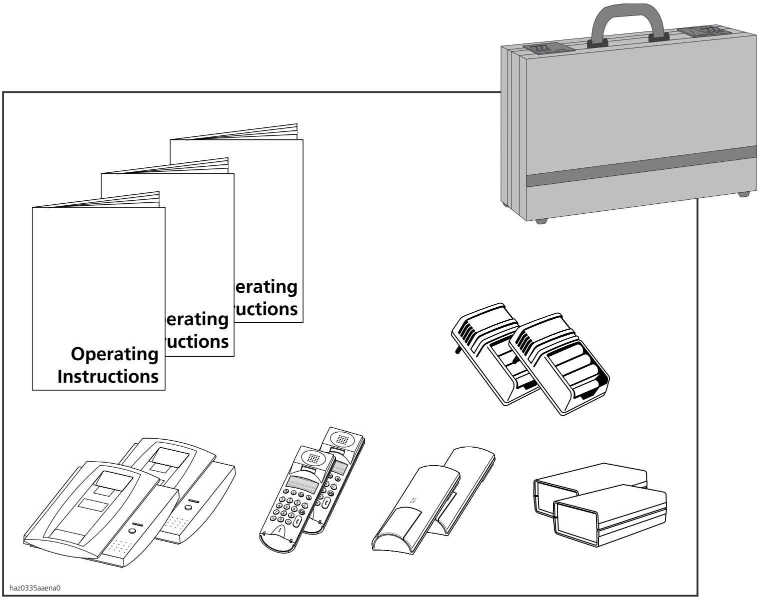


Fig. 3.32: Test kit and its contents

In addition to the field strength measurements the speech quality can also be tested using standard handsets. Additional handsets can be connected to the system for this purpose. The radio unit used for testing is a special radio unit that cannot be used on the system.

5.5.1 Handset types

2 handset types are used for planning:

- The test handset is used for coordinating call connections among the persons involved in the measurements.
- The standard handset is used purely for radio measuring purposes.

Only the characteristics necessary for planning are discussed here. For more detailed information, see the handset operating instructions.

Operating conditions

The ambient temperature during operation should range only within the limits of 5 to 40 °C (non-condensing).

The operating time with batteries is around 5 hours at room temperature; the lower the temperature, the shorter the operating time.

General / Function

Always remove the batteries from the handsets before placing the handsets in the case. (Batteries are discharged even when the handset is switched off.)

Configuration with 2 radio units

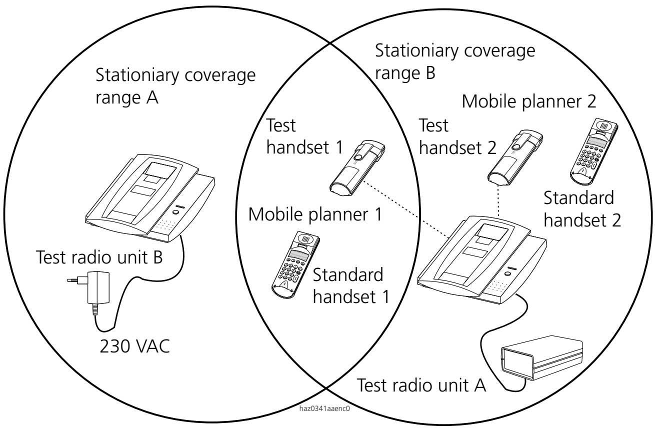


Fig. 3.33: Configuration with 2 radio units

Used for:

- Determining the optimum overlap for the two coverage ranges
- Increasing efficiency by measuring simultaneously in 2 coverage ranges
  - Planning work carried out by 2 people
  - Calls with the test handsets are possible (only via the same test radio unit)

All the equipment is always duplicated. This makes for more efficient planning within the team. The standard handsets are used for determining the field strength. They indicate the field strength of one or more radio units in the range from 0 to 65 dBµV. For this, the equipment switches to a look-around mode in which the handset provides a selection of all the radio units received. Up to 4 radio units can now be selected and measured with one another. For measurement-related reasons this is however not recommended.

To be able to determine the transitions, it is useful to have 2 radio units in operation at the same time. In some cases you may also want to prevent the changeover from being signalled. The handover behaviour of the standard handset is selectable.

A second handset type is also available to each planner: the test handset for setting up call connections. For a call connection we need to select test radio unit A or B.

Configuration with 1 radio unit

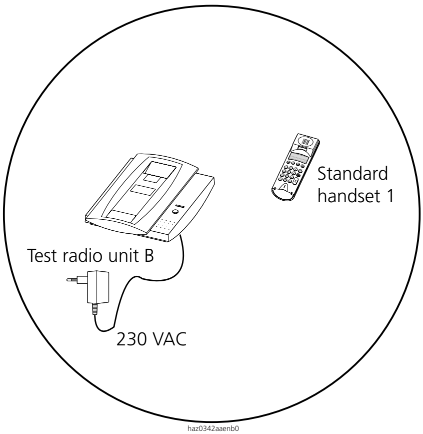


Fig. 3.34: Configuration with 1 radio unit

Used for:

- Terminating area at the limit of the planned supply ranges
- Miniature system
- Re-measuring the system already in operation (planning usually carried out by 1 person)

5.5.2 Default settings on the test handsets

The following log-on procedures need to be carried out so that the handsets can later be used anywhere in the test itinerary. These procedures ensure that both test handsets can be called on both test radio units using in each case their internal number. One test radio unit is designated as "A", the other as "B".

Supplied condition of the test kit test handsets

- Handset 1 is logged on to test radio unit "A and B" as subscriber 1.
- Handset 2 is logged on to test radio unit "A and B" as subscriber 2.

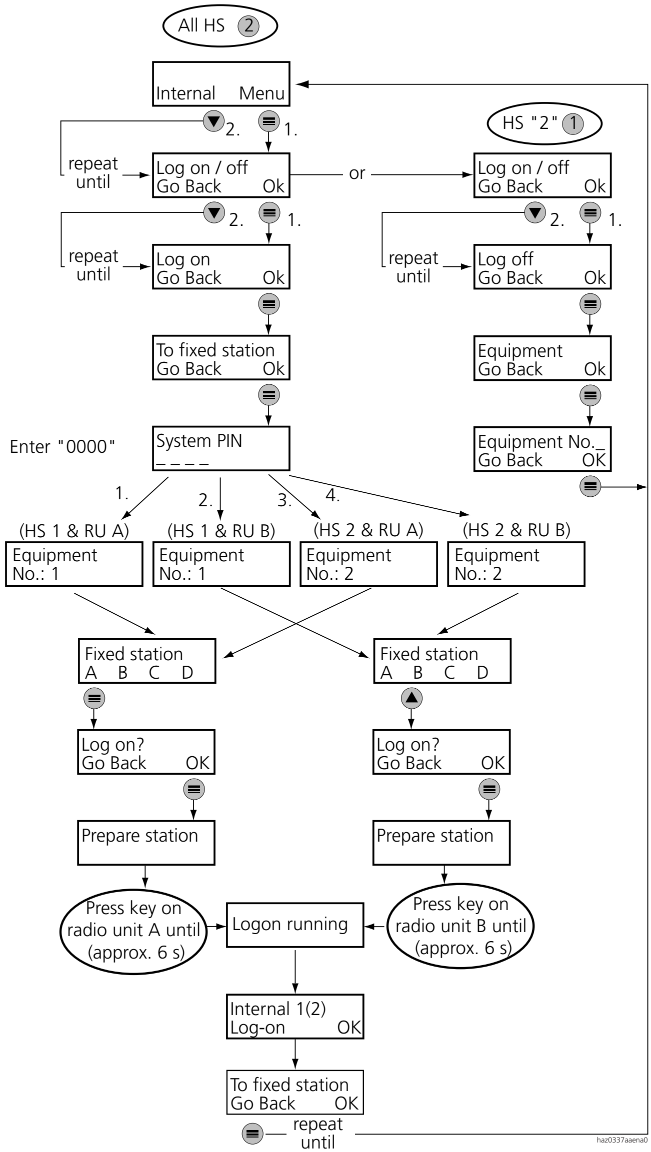


Fig. 3.35: Logging the test handsets on to the test radio units

The handsets are already logged on in the supplied condition.

5.5.3 Switching radio units

Since call connections between test handsets can only be made via the same test radio unit, it is important to know how to use the test handset to switch to another test radio unit. The "Automatic" mode lets the handset choose the radio unit and is therefore not recommended.

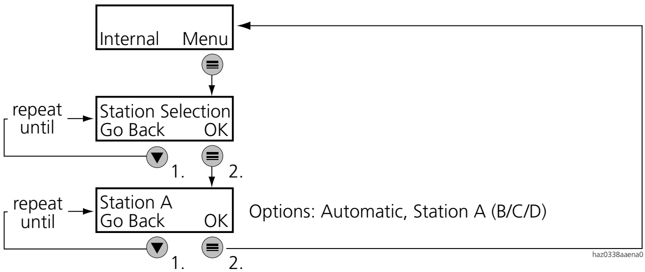


Fig. 3.36: Setting the switchover to another test radio unit on the test handset

Standard handset operation

The standard handset corresponds to the normal system handset and can also be used for planning purposes.
Here we shall only be discussing the procedures required for planning.

To switch between normal mode and test mode ("TEST" displayed), proceed as follows:

- From normal mode to test mode:
  - Press the "M" key repeatedly until "PRG1" appears.
  - Now press the "M" key repeatedly until "Test" appears.
  - Press the Foxkey when "Test" appears.
- From test mode to normal mode
  - Press the "C" key repeatedly until you exit the test menu.

Menus in the test mode:

Two menus are available when you access the test mode (see menu structure in Fig. 3.38 to Fig. 3.42).

Menu 1: "Look"

This menu takes you to the menus for measuring the coverage ranges of radio units.

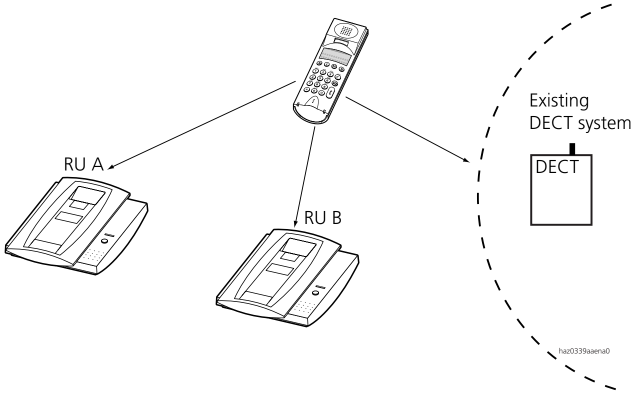


Fig. 3.37: Look-around mode

In this mode the handset recognizes up to 10 radio units in whose coverage area it is situated.

- The right-hand Foxkey can be used to browse through the radio units (Display 1, see Fig. 3.44 and Tab. 3.23).
- The radio unit to be measured out can be selected with the M key and viewed on the handset's display (display 2, see Fig. 3.46).
- The full RFPI of the currently displayed radio unit is always displayed with the "i" key.

The field strength of 4 radio units simultaneously can be shown on the handset's display. For measurement-related reasons this is however not recommended! The Foxkey "Tick" is used to select which radio units are to be displayed together (Tab. 3.24). With the M key you can branch to "Idle Lock" (Tab. 3.26) and with the Foxkey further to "Locked" (Tab. 3.27)

Menu 2: "Show HO"

This menu indicates the handover behaviour when switching over to another radio unit (page 654).

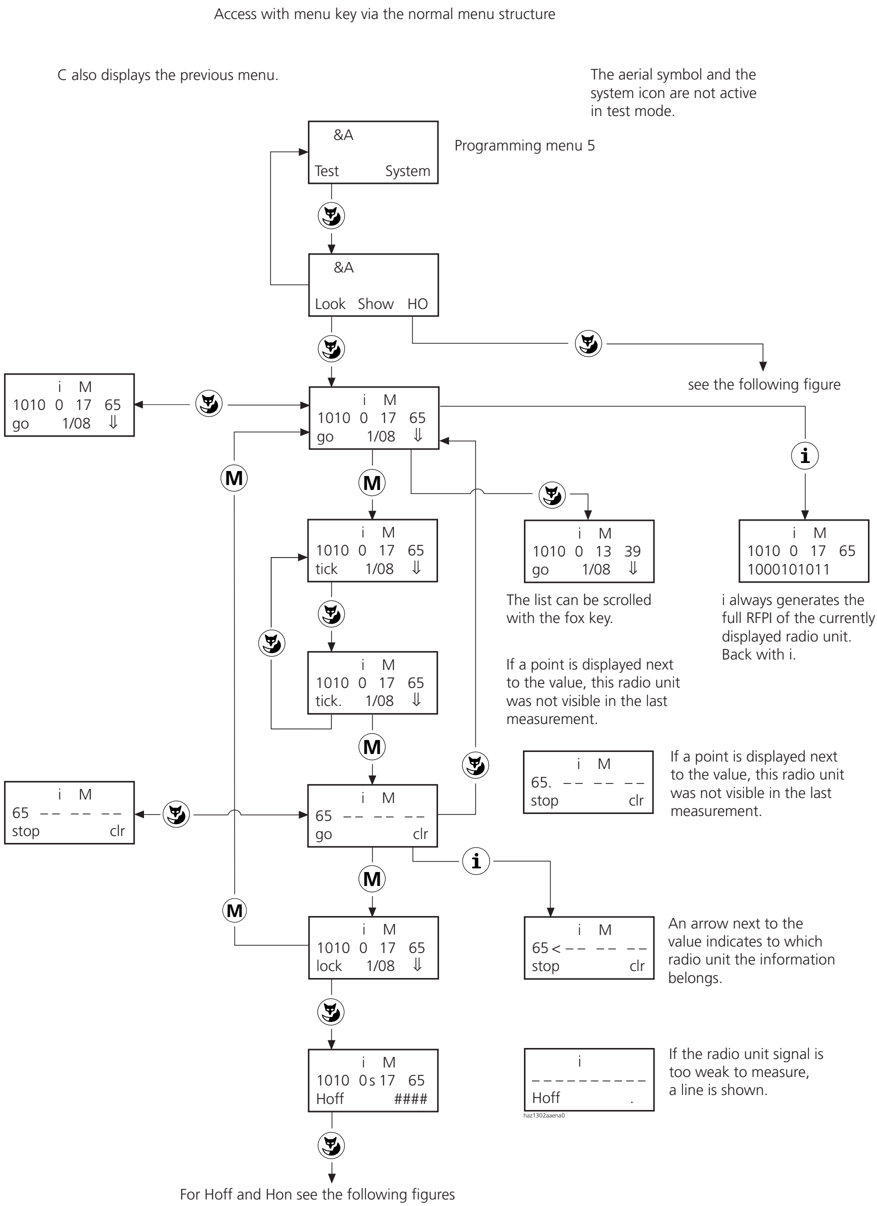
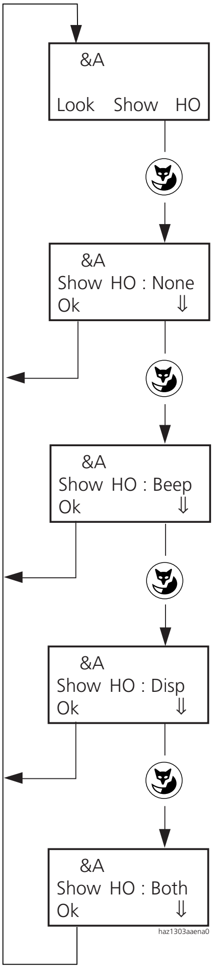


Fig. 3.38: Test modes the Office 100 handset



For look mode and normal mode a selection can be made as to whether handovers are to be displayed.

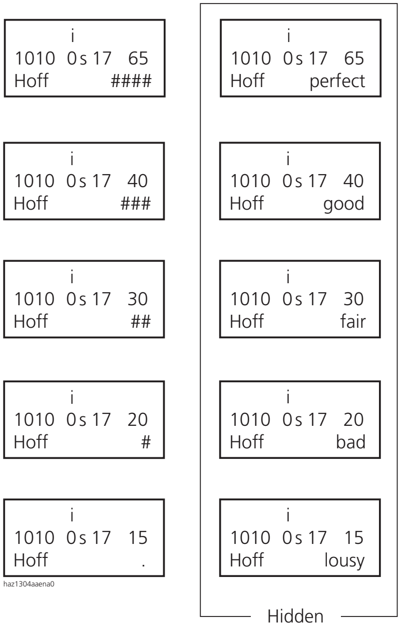
The beep setting specifies that each handover is acoustically indicated by a beep tone: high-pitched tone for connection HO to a new radio unit, low-pitched tone for bearer HO to the same radio unit.
The beep is output only in normal mode.

The "Disp" setting specifies that each handover in normal mode is indicated on the display.

Before the HO:
The bar indicates that the signal strength of the current radio unit (No. 17) is bad and that HO should be attempted.
HA attempts HO connection to a new radio unit (No. 18).

After the HO:
The bar indicates that the signal strength is very good. The new radio unit No. is visible in the display (No. 18).

Fig. 3.39: Selection of the handover display for Look mode and Normal mode



The signal strength in look mode is indicated ba 0 to 4 black fields by default.
The more active fields there are, the better the signal.
The number of fields means:

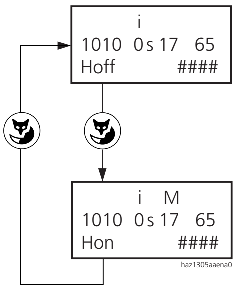
4 fields:	perfect signal
3 fields:	good signal, no handover required
2 fields:	fair signal, handover is required, otherwise the connection may be lost!
No field:	no usable signal is receivable.

If a channel is disturbed (e.g. by another DECT or a radio), the disturbance affects the signal strength display: the number of fields is reduced by 1 or 2 as long as the disturbance continues.

A corresponding standard text display is also possible. The fields are then replaced by words.
Field-to-text correspondences:

4 fields:	perfect
3 fields:	good
2 fields:	fair
1 field:	bad
No field:	lousy

Fig. 3.40: Display of the signal strength in Look mode



Hoff deactivates roaming:
The HA then remains locked on the selected radio unit and no roaming to a new radio unit will be made.

Hon activates roaming and permits HA to synchronize to a new radio unit and lock.
This makes it possible to see to which radio unit an HA would handover. Although no actual handover takes place, the process can be interpreted as such.
The HA is just not logged on to system.

Default setting:
Handover permitted, thus Hoff in the display.

Fig. 3.41: Hoff and Hon in Lock mode

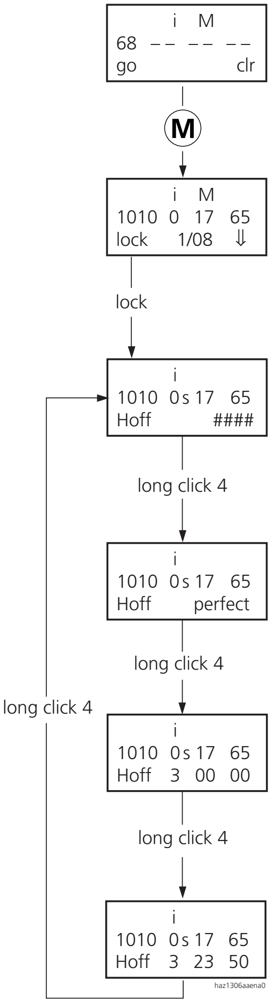


Fig. 3.42: Switching with long click on 4 in locked mode

In the previous menu trees Fig. 3.38 to Fig. 3.42 the displays are represented with substitute symbols. The symbol line of the display field which is not shown here is described in the handset operating instructions. Below are explanations of the displays using practical examples.

Procedure for an individual measurement

Display in Look-around mode

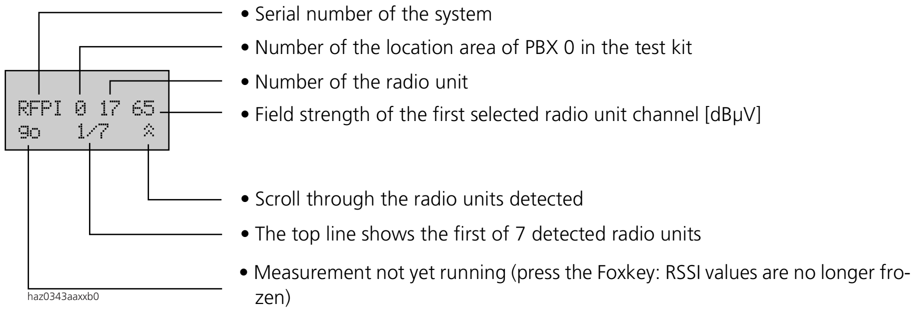


Fig. 3.43: Measure / Look-around

Selection of an individual radio unit to measure (display 1)

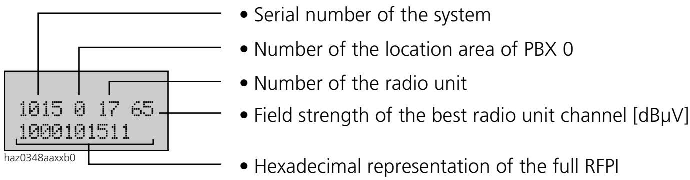


Fig. 3.44: Measure / Look-around state

Selection of the radio units to measure

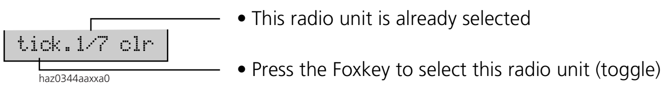
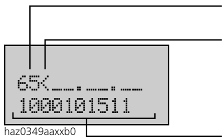


Fig. 3.45: Measure / Tick

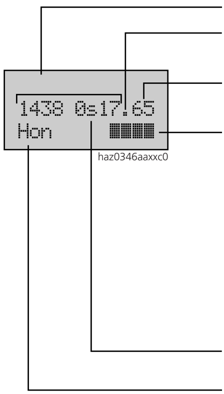
Selection of multiple radio units to measure (display 2)



- Field strength value of the first radio unit in the group
- This radio unit information is shown in the status line.
  When the signal is too weak, a line is shown instead of the radio unit number
- Hexadecimal representation of the full RFPI

Fig. 3.46: Measure / Group state

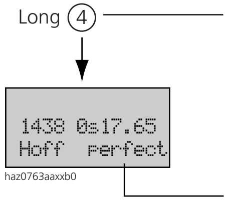
Display of the voice quality (bar indicator = standard indicator)



- RFPI: Radio Fixed Part Identity
- Handover notification (dot)
  (if handover is successful, new RFPI appears without a dot)
- Field strength of the RU (dBµV)
- Quality bar indicator (CRC errors and field strength)
  – ▮▮▮▮ Very good
  – ▮▮▮ Good
  – ▮▮ Sufficient
  – ▮ Poor
  – . No signal
- Handset is synchronized to this radio unit
- Handover behaviour
  – Hon = Enable handover by pressing the Foxkey
  – Hoff = Bar handover by pressing the Foxkey

Fig. 3.47: Measure and Re-Measure / Locked

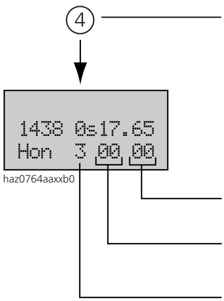
Display of the voice quality (text indicator)



- First long-click on 4 after accessing the ”Measure and Re-Measure / locked state
- Indication of quality values in text form
  – perfect
  – good
  – fair
  – bad
  – ousy (no signal)

Fig. 3.48: Quality indication in text form in "Measure and Re-Measure / Locked" mode

Display of transmission errors (reflections)



- First long-click on 4 after accessing the ”Measure and Re-Measure / locked” state
- CRC-Counter in total (CRC = Cycle Redundancy Check)
- CRC-Counter language only
- Voice quality: (3 = good, 7 = mute)

Fig. 3.49: CRC counter indication in "Measure and Re-Measure / Locked" state

Note:
With reflections the CRC counter > is zero.

Display of min. / max. values

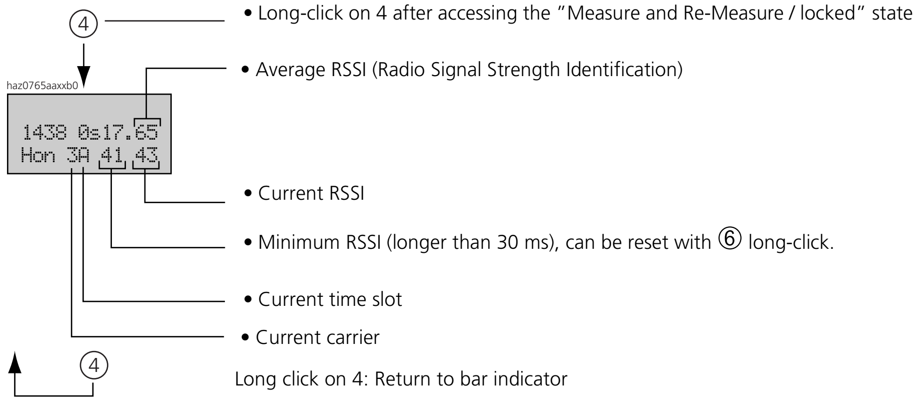


Fig. 3.50: Minimum and maximum RSSI values in the Measure and Re-Measure / Locked" state

Procedures for a multiple measurement

The field strength of 4 radio units simultaneously can be shown on the display of the handset. For measurement-related reasons such a multiple measurement is however not recommended.

Display in Look-around mode

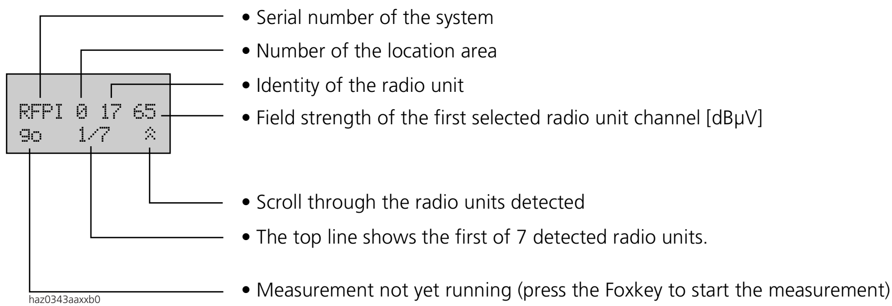
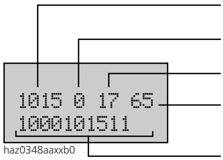


Fig. 3.51: Measure / Look-around

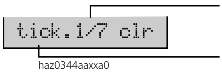
Selection of an individual radio unit to measure (display 1)



- Serial number of the system
- Number of the location area
- Number of the radio unit
- Field strength of the best radio unit channel [dBµV]
- Hexadecimal representation of the full RFPI

Fig. 3.52: Measure / Look-around state

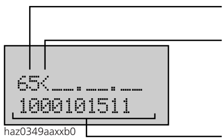
Selection of the radio units to measure



- This radio unit is already selected
- Press the Foxkey to select this radio unit (toggle)

Fig. 3.53: Measure / Tick

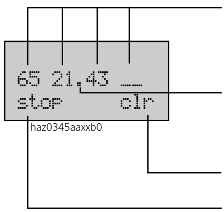
Selection of multiple radio units to measure (display 2)



- Field strength value of the first radio unit in the group
- This radio unit information is shown in the status line.
  When the signal is too weak, a line is shown instead of the radio unit number.
- Hexadecimal representation of the full RFPI

Fig. 3.54: Measure / Group state

Measuring multiple radio units



- A maximum of 4 radio units can be selected to be displayed simultaneously with their field strength value
- This radio unit is not recognized at present (last measured value and dot are signalled)
- Clears all ticks
- Measurement in progress (press the Foxkey to halt the measurement)

Fig. 3.55: Measure / Group

Note:
The simultaneous measurement of radio units is not recommended for measurement-related reasons.

Measuring the coverage ranges of the test radio units

This method is used for planning a new system with the aid of the equipment in the test kit, i.e. there is no PBX already in operation.

The "Look Around" menu is used for listing, selecting and measuring radio units "detected" by the handset:

Tab. 3.23: Indications in the "Look Around" menu

Display	Function
"Look"	Access from the "Test" menu
"go"/"stop"	Starts, stops the continuous measurement
"⁂"	Presents one radio unit in each case from the list of all radio units received
"M"	Access the selection menu
"C"	Goes back to normal menu

The "Tick State" menu defines the composition of the group of radio units selected:

Tab. 3.24: Indications in the "Tick" menu

Display	Function
"tick"	Includes the radio unit shown in the measurement
"⁂"	Presents one radio unit in each case from the list of all radio units received
"M"	Display of single or multiple radio units
"C"	Goes back to "Look Around"

Tab. 3.25: Indications in the "Group" menu

Display	Function
"go" "stop"	Starts / stops the continuous measurement
"clr"	Deletes single or multiple radio units
"M"	Selects a specific radio unit / go to "Idle Lock" menu
"C"	Goes back to "Tick"

The handset’s normal system response is now switched off, i.e. it remains on this radio unit and cannot take any calls. Outgoing calls can be made to all system subscribers.

The "Idle Lock State" menu is used for re-measuring an installed system radio unit in operation:

Tab. 3.26: Indications in the "Idle Lock" menu

Display	Function
"M"	Display of single or multiple radio units to measure
"lock"	Access the ”Call quality” menu
"C"	Goes one step back

The "Locked State" menu is used for measuring the selected radio unit:

Tab. 3.27: Indications in the "Locked" menu

Display	Function
Long click on 4	Access to further submenus
"Hon"	Enables the switch to another radio unit (at present not permitted before this key is actuated)
"Hoff"	Bars any switch to another radio unit (at present not permitted before this key is actuated)
"C"	Goes back to normal menu

Long-Clicks

By pressing the following keys you can access additional functions directly. These indications apply to Office 100, Office 135, Office 150 and Office 155pro. Deviations are indicated by footnotes.

Tab. 3.28: Long-clicks on the handsets

Key	Function
0[1]	The handset is switched off to save power. It is switched back on with the Connect key and the Foxkey. Once the handset is switched off, it **cannot** receive any more calls!
1[2]	Switches over to the next radio system.
2	Indicates the radio system parameters (handset IPEI and radio system PARK). With each additional call the next radio system is indicated in each case if there are other logons.
3	Indicates the handset's internal diagnostics.
4	Indicates the data of the valid radio unit. Continually updated during connection. One-shot display if no connection.
5	Indicates the handset's software version.
6[3]	Indicates battery charge status and the type.
7[4]	Indicates the handset's internal diagnostics.
8	Activates "semi" key lock". See Operating Instructions for details.
9	Activates key lock. See Operating Instructions for details.
5 + 3	Switch error messages on/off (first start value: off). Messages relating to the following errors cannot be switched on/off: HS logon error, incorrect location registration, no locatable radio unit, network, PBX or radio unit overload.
5 + 4[5]	Switch on/off warning tone on exceeding radio range (first start value: on).
5 + 7[5]	Swtch on/off tone accompanying error message for overloaded radio unit (system busy) (first start value: off).

1) Office 135 can also be switched on/off using the C-key

2) Office 135 and Office 155pro temporary switchover only.

3) Not available for Office 150 / Office 155pro.

4) Office 135 / Office 155pro: Indicates the PBX's software version.

5) Office 135 / Office 155pro: Long-click on the #-key.

See also:
Descriptions of other key long-clicks (to simplify function operation) can be found in the Office User's Guide.

Re-measuring the handover response of the installed system radio units

This method is used for checking a system that is already in operation. Both the system and the standard handset can be used.

Handover setting

The "Handover Setting" menu is used for setting the operating mode required for signalling the switch to another radio unit:

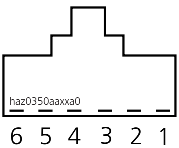
- "Show HO": from "Test" menu
- "∨": presents one selection in each case (in succession)
- "None": no indication / signalling for a switch to another radio unit
- "Beep": indication / signalling for switch activated with acoustic signal
- "Disp": indication / signalling for switch activated with indication of the destination radio unit
- "Both": indication / signalling for switch over activated, with acoustic signal and indication of the destination radio unit
- "Ok": accept choice and go back to "Test" menu

5.5.4 Radio units

Testing the radio unit voltage supply

The power supply is ensured by the plug-in power supply unit, which is permanently connected to the radio unit. If a 230 VAC connection is not available, the radio unit can be operated with a battery box.

Tab. 3.29: Connecting a battery to SUPPLY

SUPPLY	Pin	Function
	1	Supply +
	2	Supply -
	3	–
	4	–
	5	–
	6	–

The voltage must be between 9 and 15 V (DC voltage). The current consumption is 230 mA. This means that with a capacity of 1.1 Ah the battery box can provide up to 5 hours of measuring time.

Radio unit distances to the PBX

When planning a system it is important to note that with a SB-4 radio unit the power supply can be provided from the AD2 bus for distances of up to 660 m (wire diameter 0.5 mm). From 660 m up to the maximum for the AD2 bus of 1200 m, a local power supply has to be provided with the recommended plug-in power supply unit, d. h. you need to provide a 230 VAC connection.
A local power supply is not required for the SB-8 radio unit as it can be powered via the AD2 bus up to 1200 m.

Tip:
The SB-8 radio unit can also be powered locally to relieve the terminal power supply. This can also prevent the use of an auxiliary terminal power supply.

5.6 Measurements

We are assuming that the project planner is familiar with the information contained in the previous sections of this document. The equipment should be set in accordance with "General / Function", page 637.

A word about handling batteries

The batteries supplied with the test kit are state of the art. Nonetheless a few points need to be observed to ensure perfect operation:

- It takes a few charge cycles for the batteries to reach their nominal capacity. Therefore discharge / charge the batteries several times before use; the charging unit inside the test kit provide the discharge function.
- Self-discharge amounts to around 1 % per day. Re-charge before use; this process takes only a short while with the charging unit.
- Always remove the batteries from the equipment as soon as the measurements are completed and recharge them immediately.
- The batteries need to be discharged / recharged from time to time if they not used for a longer period of time.

5.6.1 Functional test prior to use

Prior to any field use, we recommend a functional test in accordance with the instructions below:

1. Recharge the batteries
   – Fit the handsets and battery boxes with batteries.
   – Put the equipment into operation
2. Test handsets: Handset 1 dials handset 2
   → Connection is established via one of the two radio units.
3. Switch the handsets over to the other radio unit
   (see "Switching radio units", page 641).
4. Handset 1 dials handset 2 via the other radio unit
   → Connection is established
5. Standard handsets: Using both handsets, call up the list of radio units present using the "Test" – "Look" menus. Use the "⁂" key to scroll through the list.
   → The two radio units appear on the list.
6. Use the "⁂" key to determine the field strength value of the two radio units.
   → The measured value should be 60 hin the immediate vicinity of the test radio unit.

5.6.2 Typical measurement process

The following work instructions have been compiled as a summary of the detailed explanations above to enable efficient work on site. We are assuming that the equipment has been correctly prepared and tested.

We shall be looking here at the case of a configuration with 2 radio units.

Typical procedure

- Install test radio unit A in the first position according to the plan and connect the plug-in power unit, provided a 230 VAC connection is available (if not, use the battery box).
- Set the standard handset to the Look-Around mode:
  – Press the "M" key repeatedly until "Test" appears
  – Press the Foxkey at "Look"

A list of radio units appears:

- Top line: Identification, last digit – field strength in dBµV
- Bottom line: "go" (Foxkey), starts the measurement
  Selected radio units / number of visible radio units "⁂", selection of radio units (scrolling).
  This display is used to provide an overview of the radio units to be received.
- To display only the field strength of radio unit A, you need to select A from the list:
  – Press the "M" key
  – Press the Foxkey repeatedly at "⁂" until the identification corresponding to radio unit A appears on line 1.
  – Then press "Tick"; a dot appears next to "Tick".
  – Press the "M" key
    The field strength of radio unit A appears
  – Press the Foxkey at "go": The measurement starts.

Check:
The measured value should be >60 in the immediate vicinity of the test radio unit.

- Display radio units A and B:
  – Press "M" -- back to the overview
  – Press "M" -- Tick mode
  – Tick radio unit B
  – Press "M" and then the Foxkey at "go" -- both radio units are displayed.

The field strengths of A and B can now be measured simultaneously and entered on the plan.

Tab. 3.30: The measured values and what they mean

Value	Meaning
> 60	Very good quality
40 to 60	Good quality
40 to 60	Mostly good quality. Crackle interference is possible depending on the position, especially when moving. A check with the test handset is recommended
25 to 30	Still accessible but stronger interference is possible, check with test handset. A good call quality will be possible during operation provided the user stands still.
below 25	No stable connection possible.

A value of at least 40 should be measurable in the overlap area of the two radio units.

Once the coverage range of the two radio units has been measured out, the radio units are repositioned. It may also make sense to measure out only part of the supply range and then to shift only one of the two radio units.

Please note

- Make sure the test handset is logged on / re-registered with the correct radio unit.
- Make sure there is only one test handset in the critical area.

Tip:
To increase efficiency, 2 people can measure simultaneously with the 2 standard handsets. In this case the test handsets are used for communicating with each other.
If you are unsure which value belongs to which radio unit, you can retrieve the information using the "i" key. The top line then displays the identification of the radio unit marked with "<". Press "i" a second time to display all 4 radio units once again.

6 Planning a private network

Information on networking concepts and characteristics can be found in Part 1 and Part 2. For information on Networking over IP, please refer to the System Manual AIP 6400.

The following guide on how to plan private networks is designed to help you create a small, simple network. As the number of systems in the network increases, the requirements and potential sources of error also increase exponentially. For larger networks, experience and the assistance of specialists are invaluable.

When it comes to implementing a specific network there are always several solutions and configuration options. That is why you should regard the explanations below merely as tips and advice for one possible solution. Alternatives are always possible and, depending on the problem to be resolved, more appropriate.

This chapter consists of:

- a planning aid for converting the customer's ideas into a concrete project (as of page 660)
- instructions for planning a simple specimen network (as off page 681)
- and other topics such as:
  - virtual networking (networking via the public network, as of page 697)
  - networking with a virtual PBX (Centrex, page 700)
  - networking with third-party systems (page 702)
  - networking via the Ethernet interface (see AIP 6400 System Manual)
  - other aspects of networking (page 702)

6.1 Planning aid

A customer has specific requirements for a telephony and data traffic infrastructure. The task is to determine how to convert these requirements into the best possible solution. The entire decision process is iterative. In other words, the items listed are not to be processed simply one after the other; instead, you need to return to previously defined parameters and, possibly, re-define them all over again.

You need to determine the following fundamental network properties:

- The number and types of systems to be networked
- Traffic Volume

- The routing in a private network
- The type of connection between two systems
- Accesses to the public network
- Dialling through from the public network
- Incorporating virtual subscribers
- Incorporating Centrex
- Numbering requirements
- Features to be supported

6.1.1 Number and type of systems to be networked

Determine the number of nodes and ascertain whether the use of a virtual PINX (Centrex) is appropriate:

- What locations are there?
- Which locations are to be networked?
- Is a separate system required for each location?
- What is the optimum number of nodes for each location? Would it make sense to integrate a down-circuit cordless system?
- What types of system are the best suited?
- Which systems are to be networked and how (virtual, fixed)?
- Would it make sense to integrate a Centrex solution? (A Centrex solution may make sense if, for example, some of the subscribers are separated by long distances.)

6.1.2 Past traffic volume

Over a representative period analyse the call logging values for the traffic volume between the locations using the previous solution.

- How high is the traffic volume?
- How is spread out over time?
- Are there bottlenecks or restrictions?

Estimate together with the customer how much the traffic volume is set to change in general with the new solution.

Use the results to determine the potential savings likely to be achieved with a private network.

6.1.3 Routing in a private network

Determining the routing in a private network depends on the following factors:

- Connections between the nodes
- Accesses to the public network
- DDI requirements
- Overflow routing
- Volume of traffic in individual cases

Once the routing has been specified, you can estimate how many B channels are required between two systems.

6.1.3.1 Connections between nodes

Determine the nodes to be connected with one another:

- The topology can be star-shaped, meshed or a combination of both.
- For reasons of connection reliability it is advisable to ensure that each system can be reached via at least 2 independent routes (requirement for overflow routing). See also "Reliability aspects", page 667.
- If possible specify which connections are best implemented virtually via the public network (e.g. also the integration of GSM subscribers into the private network).

6.1.3.2 Accesses to the public network

Determine which systems are to have accesses to the public network:

- Which systems are to have virtual networking via the public network? This question is to be addressed in connection with the type of connection between two systems. Virtual networking is particularly appropriate for greater distances with a relatively small volume of traffic.

- On which systems should calls be routed to the public network?
  The traffic volume on a system on the public network (gateway PINX) is the sum of the volume of traffic of the system's own traffic and the volume of traffic of all the systems in the network that telephone out into the public network using that system. The same applies accordingly for systems that forward calls from one system to another (transit PINX).
- To which system are calls with the main number routed?
- On which systems are calls from the public network to be routed directly to subscribers in the private network (direct dialling in)?
  – Should the subscribers be obtainable via several DDI numbers?
  – Should incoming calls be routed in accordance with break-in criteria? This requires other direct dialling numbers for the correct CLIP display (see "Break-In (Direct Dialling in for Virtual PISN Subscribers)", page 100).
- Where are additional connections to the public network required for overflow routing compatible with emergency operation? (See also "Reliability aspects", page 667.).

6.1.3.3 Traffic volume in the private network

With the results from the previous section and the previously estimated general traffic volume you can roughly estimate the volume of traffic for each node and for each connection between two nodes.

- How high the traffic volume is for the following types:
  – Internal traffic
  – Transit traffic
  – Transferred calls
    (e.g. calls transferred via a main number)
  – Overflow routing
    (within the private network or via the public network)?
- How is the volume of traffic spread out over time?

6.1.3.4 Determining the B channels

With the traffic volume estimate you can determine the number of B channels required.

6.1.4 Connection types between two permanently networked systems

Each connection between two nodes in the private network can be implemented differently if required. The choice of the physical connection type depends on

- Distance and line length between two systems
- Volume of traffic
- Existing infrastructure
- Financial resources

6.1.4.1 Connections with primary rate accesses

Two systems can be connected with one or several primary rate accesses. A transmission facility can be connected between the systems to bridge greater distances.

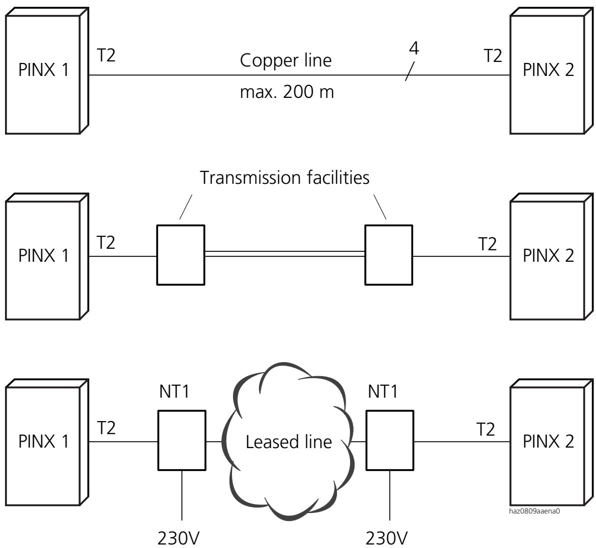


Fig. 3.56: Two systems connected via a primary rate access

Connection types for primary rate accesses:

- Copper lines without transmission equipment
- Unstructured leased lines with 30 B + D as per G.703
- Structured leased lines with (n x B) + G0.704D as per G.704
- Copper lines with transmission facilities
- Fibre optic cables with transmission facilities

The type of leased lines available depends on the service provider.

Transmission facilities for primary rate accesses:

- HDSL modems for primary rate accesses, 2 or 4-wire variants
- HDSL modems with multiplexer for combined telephony and data networks
- Transmission facilities for fibre optic cables

6.1.4.2 Connections with basic accesses

Systems can also be connected with one or several basic accesses.

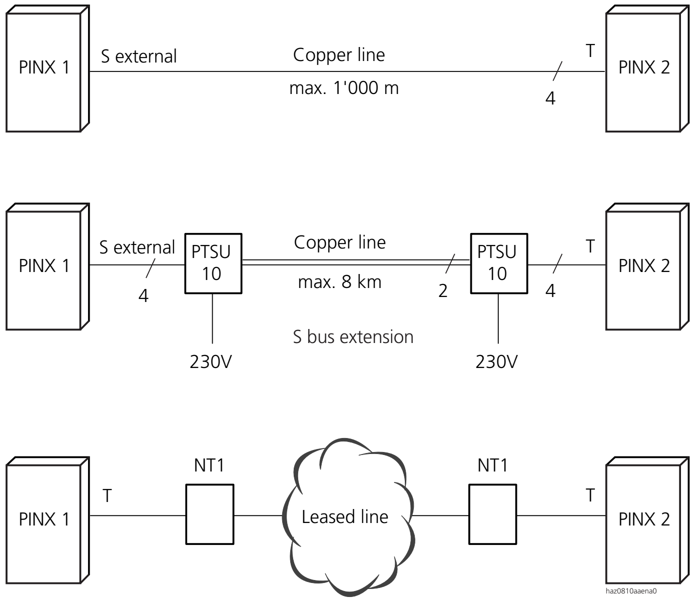


Fig. 3.57: Two systems connected via a basic access

It is important that an S bus be used on the one side (to which terminals are normally connected) and a T interface on the other side as for the public network.

The user-network interface with the S bus is converted to "S-bus protocol = EXTERNS" in the interface configuration. As a result the user-network interface acts like a network interface. This means no more terminals can be operated there and no numbers allocated. In return the network interface can now be incorporated into a trunk group.

The system with access to the public network should contain the S interface to ensure the systems are synchronized (see also "Synchronization", page 669).

Several basic accesses between two systems are grouped together in a trunk group.

Connection types for basic accesses:

- Copper lines without transmission equipment
- Leased lines
- Copper lines with transmission facilities

- Fibre optic cables with transmission facilities

The type of leased lines available depends on the service provider.

Transmission facilities for basic accesses:

- PT 10 S bus extension up to 8 km line length (see "S Bus Extension PT 10", page 138)
- HDSL modem for basic accesses, 2 or 4-wire variants
- HDSL modem for multiplex for combined telephony and data networks

6.1.5 Protocols and licences

The PSS1 (QSIG) protocol is generally used in the private leased-line network. The appropriate licence is required (see also "Licence-related System and Expansion Limits", page 593). Connections to the public network normally use the DSS1 protocol.
DSS1 should not be used on S external as otherwise only Base Call is supported.

6.1.6 Reliability aspects

Connection reliability in a private leased-line network

If a network consists of several systems, connection reliability can be increased if there are two or more paths in each case between two systems.

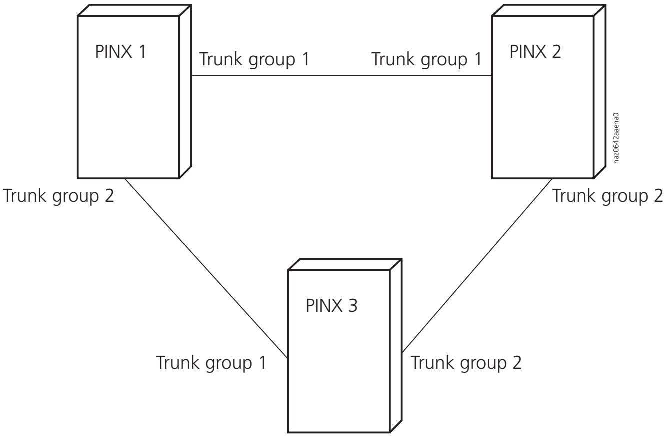


Fig. 3.58: Each system can be reached in two ways

Routes are used for outgoing routing, as described in "Example of networking", page 681.

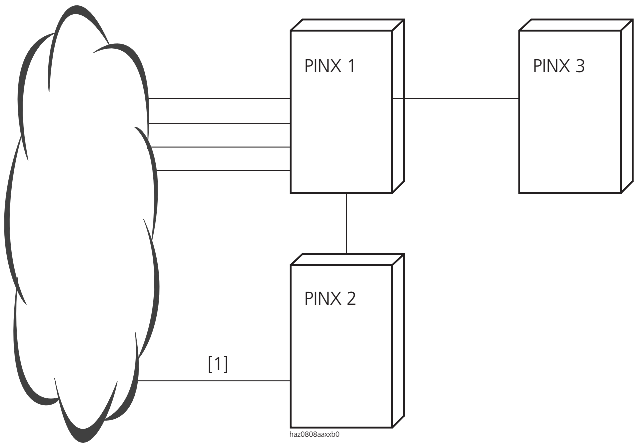
Route 2 in system 1 for example is used for connections from system 1 to system 2. In this route trunk group 1 is now added in the first position and trunk group 2 in the second.

Under normal circumstances calls will be routed directly from system 1 to system 2 via trunk group 1. If, however, the lines of trunk group 1 are faulty or if they are all busy, calls will be routed over trunk group 2 via system 3.

Connection reliability with the public network

To guarantee a high level of connection reliability also with the public network, it is useful to connect at least two systems with the public network.

Overflow routing can then be implemented via the public network using Least Cost Routing.



[1] Basic access used to increase connection reliability

Fig. 3.59: Connection reliability in the public network

6.1.7 Synchronization

A PBX’s clock frequency is predefined (synchronized) by the public network via basic accesses T and primary rate accesses T2.

If the synchronization provided by the public network fails (due, for instance, to interrupted trunk lines), the PBX will use its own clock frequency.

In the private leased-line network, systems synchronized by the public network pass on the clock reference to systems that are not directly connected to the public network.

6.1.7.1 Clock propagation diagram

Synchronization in a private leased-line network has to be carefully planned to ensure that synchronization loops never occur. A synchronization loop occurs whenever two systems synchronize each other.

The best solution is to sketch out a clock propagation diagram based on the following pattern:

1. Enter the network systems.
2. Draw the connecting lines between the systems.
3. Determine where the clock is forwarded from and to:
   – Add direction arrows to the connecting lines. Make sure no loops are created.
   – To check the arrangement, follow the path in the direction of the arrows: If you never encounter the same system twice on the same path, you can assume that there are no loops.
4. Define network interfaces (T2, T or S external).
5. For each system configure the network interfaces of all the trunk lines with an incoming arrow in the clock reference table. One of these will then be chosen as the original reference.

Example of a clock propagation diagram: Net without loops

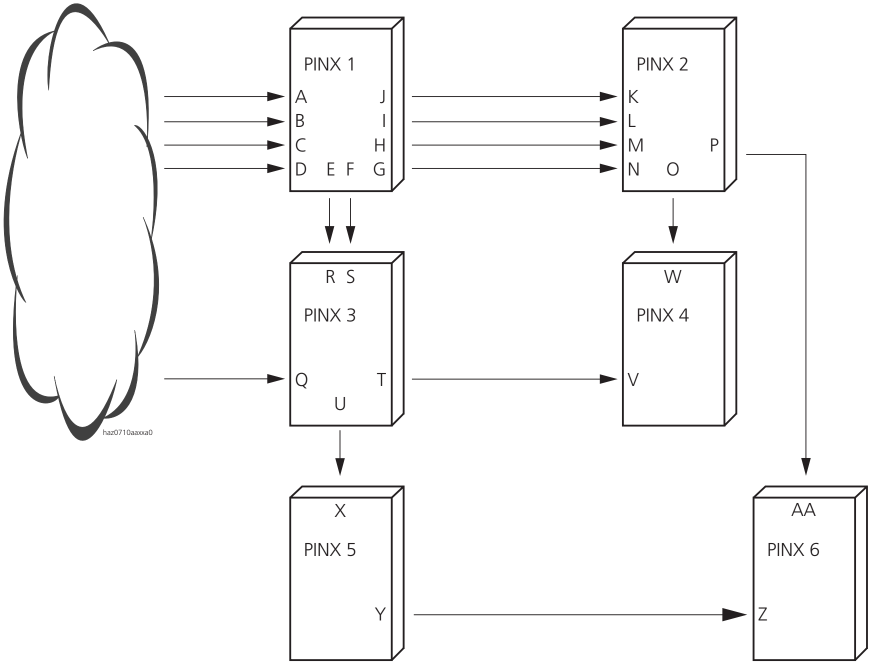


Fig. 3.60: Example of a clock propagation diagram (A to AA indicate the network network interfaces)

Tab. 3.31: Configuration for the above example

PINX	Original reference	Clock reference table	Remarks
1	A (or B or C or D)	A B C D	E and J propagate the timing and are therefore not entered in the clock reference table
2	K (or L or M or N)	K L M N	O and P propagate the clock and are therefore not entered in the clock reference table
3	Q	Q R S	T and U propagate the timing and are therefore not entered in the clock reference table
4	V	V W	
5	X	X	Y propagates the timing and is therefore not entered in the clock reference table.
6	AA	AA Z	AA is better than Z because more lines to the public network are available.

Negative example of a clock propagation diagram: Net with loops

In this example PINX 1 might synchronize with PINX 3 while at the same time as PINX 3 synchronizes with PINX 1. The systems will run out of sync.

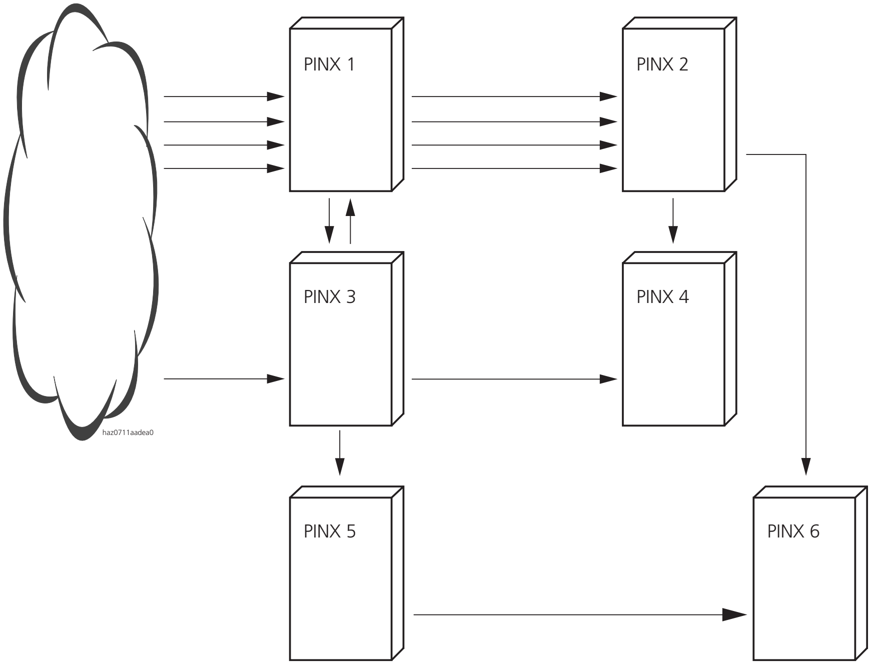


Fig. 3.61: Negative example: Net with synchronization loops

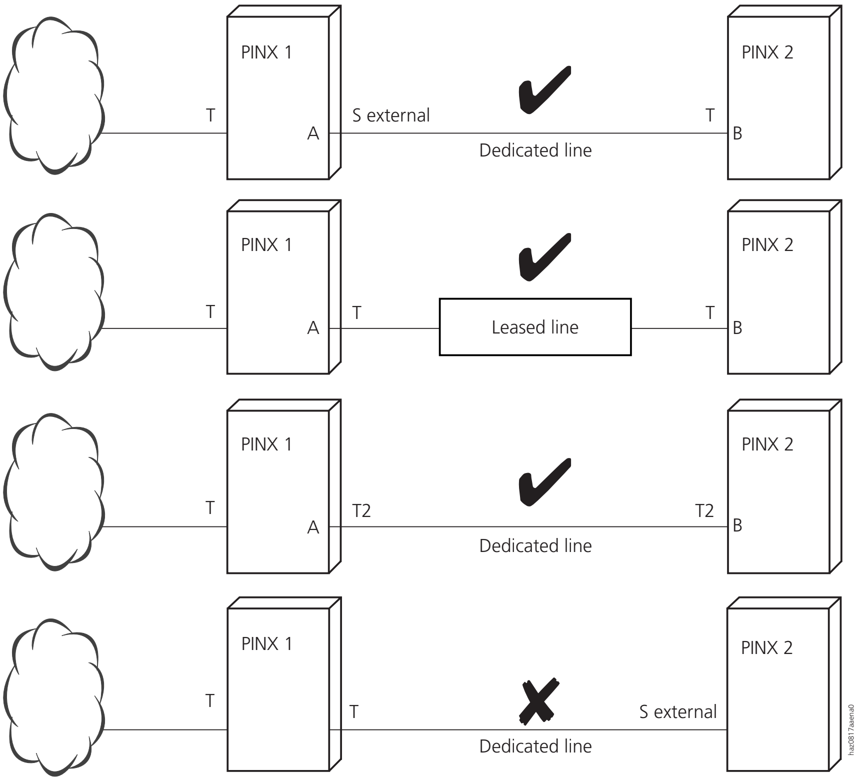
6.1.7.2 Planning rules

The object of the planning is as follows:

- To use the clock propagation source to draw up the list of all the possible clock sources for each PINX (see Fig. 3.60).
- To determine the best possible clock source for each system, It is then configured as the initial reference.

Follow the following rules when creating the clock propagation diagram:

- Each T or T2 interface of a system can act as the clock source for that system and be entered in the clock reference table.
- A T2 interface is preferable to a T interface.
- A connection to the public network is preferable to a connection to the private network.
- The original reference should always consist of the connection closest to the public network.
- Each T2 or S external interface is capable of passing the synchronization on to another system. However, such an interface must not be entered in the clock reference table or else synchronization loops may occur.
- An S external interface cannot be used as a clock source; it can, however, transfer the clock to another PINX. This should be taken into account when planning the network.
- A T interface is not suitable for passing on the timing as in fixed networking it can only be connected either to an S external interface or to a leased line. However, an S external interface cannot receive a clock timing, and the timing from the public network is always supplied via a leased line.



A should not be in the clock reference table
B should be in the clock reference table

Fig. 3.62: Possible and prohibited connections

Initialization setting

All T interfaces are entered in the clock reference table. Therefore, if a system is integrated in a network, certain network interfaces will have to be excluded from the clock reference table.

6.1.8 Numbering

There are two methods for setting up the numbering plan of a private network:

- Numbering with blocks (shared numbering plan)
- Numbering with regions

6.1.8.1 Numbering with blocks (shared numbering plan)

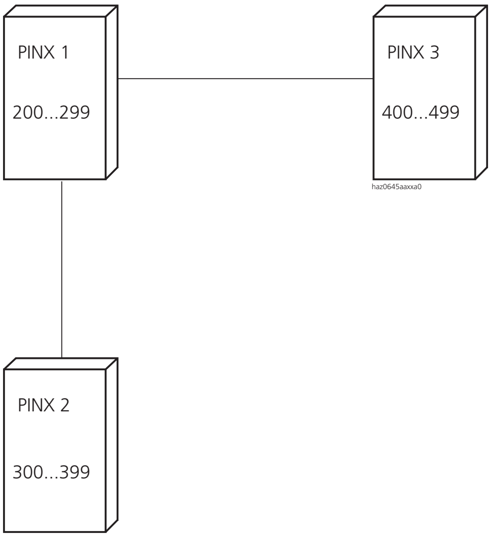


Fig. 3.63: Numbering with blocks (shared numbering plan)

The numbering range is divided into blocks. Which are distributed among the systems. This method is preferable as the user does not have to know the network topology. He can reach any subscriber in the network simply by dialling the internal number, regardless of which system he is connected to. Drawbacks of the method:

- When networking existing systems the existing numbering plans sometimes have to be adapted. Subscribers do have to be allocated new numbers.
- The number of network subscribers is limited by the number range available.

The page 689 contains an implementation example for this method.

6.1.8.2 Numbering with regions

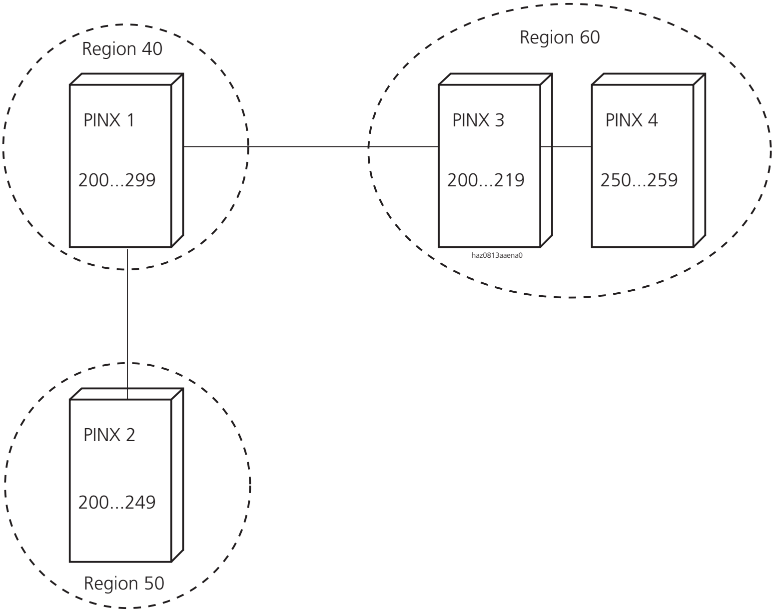


Fig. 3.64: Numbering with regions

The network is subdivided into regions. One or more systems belong to each region. Each region has its own numbering plan. The systems within a region share a numbering plan, with the number range divided up into blocks (see "Numbering with blocks (shared numbering plan)", page 675). Each system is allocated to a region using the setting "Own Regional Prefix".

Advantages of this method:

- When networking existing systems the existing numbering plans may have to be adapted. Subscribers do not have to be allocated new numbers.
- The number of network subscribers is not restricted by the number range of a single numbering plan as there is a separate numbering plan available for each region.

Numbering with regions can be implemented in two ways:

- Region selection via PISN subscriber
- Region selection via local area network

Region selection via PISN subscriber

In each system a PISN subscriber is created for each region. For example, in system 1 a PISN subscriber 60xxxx is created for Region 60.

Advantage: individual subscribers can also be allocated uniquely and are therefore obtainable through dialling by name.

Drawback: Only numbers with the same digit length can be reached using the entry with wildcard characters.

The page 693 contains an implementation example for this method.

Region selection via local area network

In each system a route is created with the region's call number for each region in the numbering plan. For example, in system 1 a route is provided with call number 60.

Advantage: All the numbers in Region 60 are obtainable, regardless of how many digits they have.

Drawback: Subscribers in the network cannot be dialled by name.

Note:
This method cannot be used in gateway systems as an incoming call from the public network cannot be routed to the destination system.

The page 694 contains an implementation example for this method.

6.1.9 Link-up to a public network

In a private network any number of systems can be provided for the link-up to the public network. It may involve one or more systems. Systems that route a call from a different system to the public network are referred to as gateway PINXs.

Calls from systems that are not directly connected to the public network can be routed via several systems to the gateway PINX. These systems perform a transit function. They are thus called transit PINX.

The system in which a call is set up is called the originating PINX. The system to which the call's destination subscriber is connected is called the destination PINX.

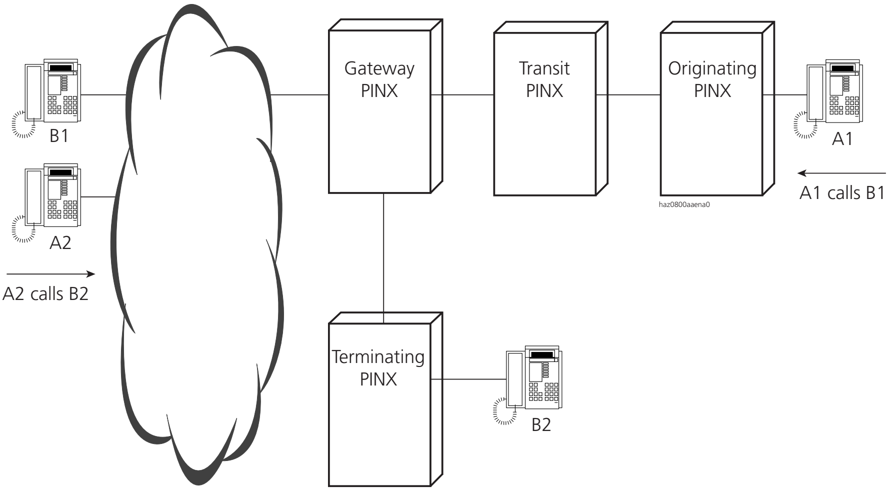


Fig. 3.65: The role of the individual systems in dealing with public network traffic

For incoming traffic, DDI numbers have to be created at a gateway PINX for all the subscribers in the private leased-line network who are to be directly reachable from the public network via the system (see "Direct dialling in at the gateway PINX", page 679).

The following points need to be taken into account for outgoing traffic:

- Ensure on the originating PINX that a call to the public network is recognisable as such (see "Identification of calls to the public network", page 680).
- Configure the transit and gateway PINX so that they forward a call to the public network (see "Definition of the transit route", page 681).

6.1.9.1 Direct dialling in at the gateway PINX

In a gateway PINX DDI numbers are created for all the subscribers of the private network who are to be obtainable directly from the public network, via the system. The destination subscribers in their own system and in the other systems are entered in the relevant call distribution elements for each switch position of the allocated switch group.

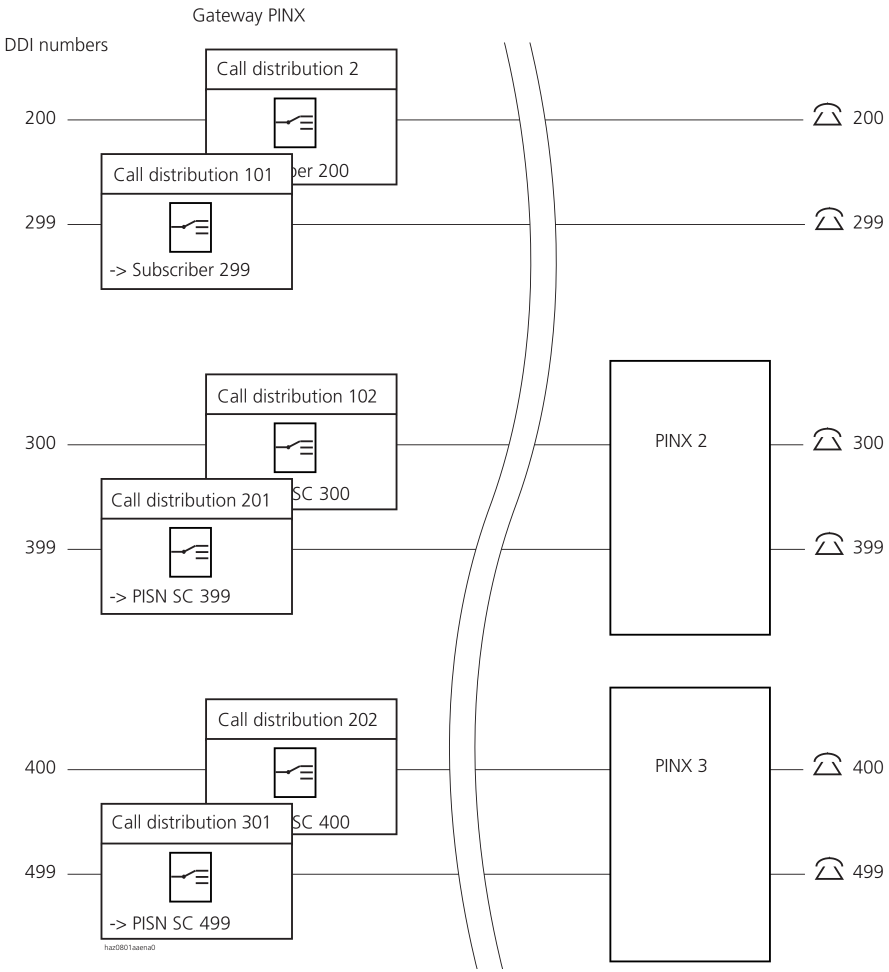


Fig. 3.66: In the gateway PINX DDI numbers are created for all the subscribers in the network

6.1.9.2 Identification of calls to the public network

If a transit PINX or a gateway PINX is to recognize whether or not it should forward an incoming call to the public network, the call number must be an external number. As such it must

- either comply with numbering plan identifier (NPI) E.164 or
- be preceded by an exchange access prefix.

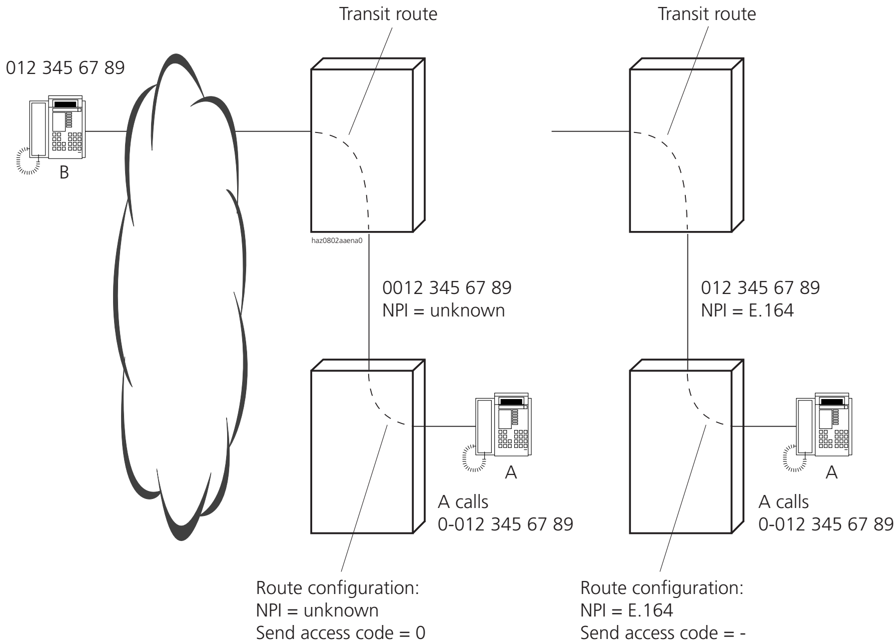


Fig. 3.67: Call number attributes for outgoing calls to the public network (2 variants)

In the system both attributes are set in the route configuration. This means that a separate route is always reserved for calls to the public network.

The page 687 contains an implementation example.

6.1.9.3 Definition of the transit route

On the transit PINX and gateway PINX you need to define the route via which calls to the public network are to be forwarded. In each system this is done with the aid of the "Transit Route" setting (under the PISN settings).

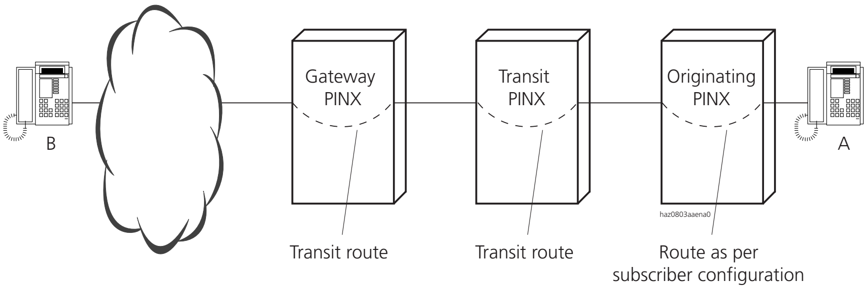


Fig. 3.68: A call to the public network is routed via the transit route in a Transit and gateway PINX

The page 684 contains an implementation example.

6.2 Example of networking

A small network will be used to describe a planning procedure. The system designation always precedes the systems so that the configurations of the various systems can be differentiated. For example , trunk group 5 of system 1 is designated as trunk group 1-5.

The following assumptions are made:

- System 1 is connected with system 3 via a primary rate access.
- System 1 is connected with system 2 with two basic accesses.
- System 1 is connected with the public network via a primary rate access (system 1 is the gateway system).

Steps in the procedure:

1. Create the routes (as of page 682).
2. Create the trunk groups (as of page 684).
3. Configure the routes (as of page 686).

4. Create the numbering plan (as of page 688).
5. Set up direct dialling in (as of page 696).

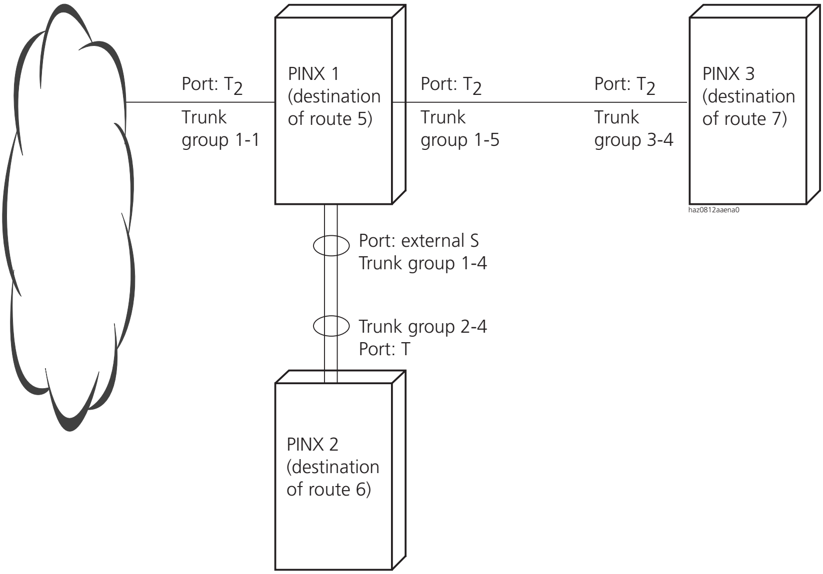


Fig. 3.69: The specimen network

6.2.1 Creating the routes

6.2.1.1 Replicating the systems on routes

A route always defines a destination. A destination is either a system in the network or a connection to the public network. You need to create as many routes as there are destinations. To ensure a clearer overview, the same route in each system is always used for the same destination, i.e. a route is reserved for each destination.

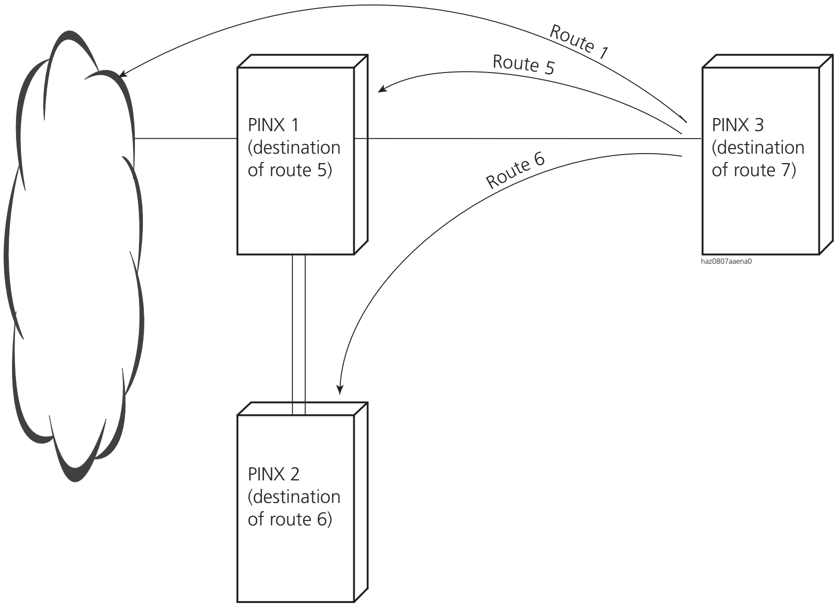


Fig. 3.70: One route is reserved for each destination

In each system create one route for each of the other systems.

1. Create a route for each system:
   – For system 1: Route 5
   – For system 2: Route 6
   – For system 3: Route 7
2. In each system, name the routes you have just defined:
   – Route 5: for PINX 1
   – Route 6: for PINX 2
   – Route 7: for PINX 3
3. In system 1 create one route for system 3 (route 1-7) and one route for system 2 (route 1-6).
4. In system 2 create one route for system 1 (route 2-5) and one route for system 3 (route 2-7).
5. In system 3 create one route for system 1 (route 3-5) and one route for system 2 (route 3-6).

6.2.1.2 Defining the routes to the public network

In each system create a route to the public network. The simplest way is to reserve one route number for the public network.

System 1 is the gateway PINX, i.e. all the calls between the private leased-line network and the public network are routed via this system. The route that assumes this task must be specified.

1. Reserve route 1 for the public network.
2. In each system create the route to the public network:
   - System 1: Route 1 (route 1-1)
   - System 2: Route 1 (route 2-1)
   - System 3: Route 1 (route 3-1)
3. In all the systems, specify route 1 as "to the public network".
4. In system 1 specify route 1 for routing calls from other systems to the public network:
   - Transit route = route 1

6.2.2 Creating the trunk groups

6.2.2.1 Creating the trunk groups between the systems

Create a trunk group with all the lines between two systems.

1. In system 1 create a trunk group with all the lines to system 2 (trunk group 1-4) and a trunk group with all the lines to system 3 (trunk group 1-5).
2. In system 2 create a trunk group with all the lines to system 1 (trunk group 2-4). As there are no lines leading directly to system 3 and there are no other lines, you do not need to create any other trunk groups.
3. In system 3 create a trunk group with all the lines to system 1 (trunk group 3-4).

You need to set the trunk groups for a private network:

4. Select the following settings in the trunk group configuration for trunk groups 4 and 5 of all the systems:
   – Network type = private
   – Protocol = PSS1

Name the trunk groups as an orientation aid:

5. Name the trunk groups of system 1:
   – Trunk group 1-4: Name = PINX 2
   – Trunk group 1-5: Name = PINX 3
6. Name the trunk groups of systems 2 and 3:
   – Trunk group 2-4: Name = PINX 1
   – Trunk group 3-4: Name = PINX 1

6.2.2.2 Creating the exchange trunk group

An exchange trunk group is created in system 1 for the lines leading to the public network:

1. Create a trunk group in system 1 with the line to the public network (trunk group 1-1).

You need to set the exchange trunk group for the public network:

2. Select the following settings in the trunk group configuration of trunk group 1-1:
   – Name = public network
   – Network type = public
   – Protocol = DSS1

6.2.3 Route configuration

Once you have created the trunk groups, configure the routes.

Allocating the trunk groups

Three routes are provided for in each system (one for each system in the network and one for the public network. You do not need one for your own system). The trunk groups are now allocated to the routes:

1. In system 1 allocate the trunk groups to the routes:
   – Route 1-1: trunk group 1-1 (trunk group to the public network)
   – Route 1-6: Trunk group 1-4 (trunk group to system 2)
   – Route 1-7: Trunk group 1-5 (trunk group to system 3)
2. In system 2 allocate trunk group 2-4 to all the routes (all the routes use the same trunk group as all the calls go via system 1):
   – Route 2-1: Trunk group 2-4 (trunk group to system 1)
   – Route 2-5: Trunk group 2-4 (trunk group to system 1)
   – Route 2-7: Trunk group 2-4 (trunk group to system 1)
3. In system 3 allocate trunk group 3-4 to all the routes. (All the routes use the same trunk group as all the calls go via system 1):
   – Route 3-1: Trunk group 3-4 (trunk group to system 1)
   – Route 3-5: Trunk group 3-4 (trunk group to system 1)
   – Route 3-6: Trunk group 3-4 (trunk group to system 1)

While systems 2 and 3 have three routes, they contain the same trunk group. This is because calls with system 1, system 2 (or 3) and the public network as their respective destinations are routed via the same lines. If at a later date systems 2 and 3 are connected with their own lines, simply create new trunk groups and re-assign the routes accordingly. All the rest remains the same; more importantly, you do not need to change the configuration of the network subscribers (PISN subscribers).

Settings for the routes to the other systems

All the call numbers of calls within the network are network-internal (PISN-internal) numbers. As such they have two special properties:

- They must comply with numbering plan identifier PNP.
- They do not have to be checked by an external digit barring.

These properties are set in the configuration to the routes:

1. In system 1 select the settings for routes 1-6 and 1-7:
   – Numbering plan identifier (NPI) = PNP
   – Digit barring = no
2. In system 2 select the settings for routes 1-5 and 1-7:
   – Numbering plan identifier (NPI) = PNP
   – Digit barring = no
3. In system 3 select the settings for routes 1-5 and 1-6:
   – Numbering plan identifier (NPI) = PNP
   – Digit barring = no

Settings for the routes to the public network

All the call numbers of calls to the public network are external numbers. As such they must

- either comply with numbering plan identifier E.164 or
- be preceded by an exchange access prefix.

In addition they should be checked by an external digit barring if required.

These properties are set in the configuration for the routes.

Variant 1:

1. In system 1 select the settings for route 1-1:
   – Numbering plan identifier (NPI) = E.164
   – Digit barring = yes
2. In system 2 select the settings for route 2-1:
   – Numbering plan identifier (NPI) = E.164
   – Digit barring = yes
3. In system 3 select the settings for route 3-1:
   – Numbering plan identifier (NPI) = E.164
   – Digit barring = yes

Variant 2:

1. In system 1 select the settings for route 1-1:
   - Numbering plan identifier (NPI) = E.164
   - Digit barring = yes
2. In system 2 select the settings for route 2-1:
   - Numbering plan identifier (NPI) = PNP
   - Send access code = 0 (as the exchange access prefix)
   - Digit barring = yes
3. In system 3 select the settings for route 3-1:
   - Numbering plan identifier (NPI) = PNP
   - Send access code = 0 (as the exchange access prefix)
   - Digit barring = yes

6.2.4 Creating the numbering plan

Now that the network is defined through the definition of the routes and trunk groups, we need to specify the numbering within the network.

In the following we shall be looking at both methods described on page 675.

6.2.4.1 Numbering with blocks

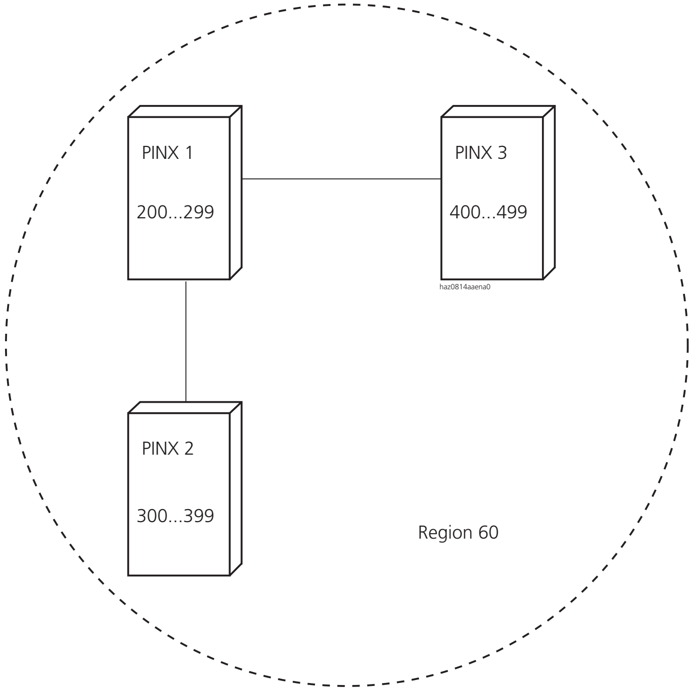


Fig. 3.71: Numbering with blocks (shared numbering plan)

Each system is assigned one (or more) number ranges. For example:

- System 1: Number range 200..0.299
- System 2: Number range 300..0.399
- System 3: Number range 400...499

With this method all the systems in the same region can be reached under the same regional prefix, which is why subscribers no longer have to dial it.

Define a separate regional prefix nonetheless for all three systems (e.g. prefix 60). In each system configure the own region prefix under the basic settings:

1. Create the internal subscribers:
   – System 1: Internal subscribers 200 to 299
   – System 2: Internal subscribers 300 to 399
   – System 3: Internal subscribers 400 to 499
2. Specify the regional prefix:
   – In System 1: Own region prefix = 60
   – In System 2: Own region prefix = 60
   – In System 3: Own region prefix = 60

With this method local area network (see page 682) is not used. The number ranges of the other systems have to be defined. To do so create numbers in the PISN subscribers category in the numbering plan. There you can create either individual numbers or number ranges by specifying the range with "XX". Each "X" corresponds to one digit place.

1. Create the PISN subscribers in system 1:
   – One PISN subscriber 3XX (range of system 2)
   – One PISN subscriber 4XX (range of system 3)
2. Create the PISN subscribers in system 2:
   – One PISN subscriber 2XX (range of system 1)
   – One PISN subscriber 4XX (range of system 3)
3. Create the PISN subscribers in system 3:
   – One PISN subscriber 2XX (range of system 1)
   – One PISN subscriber 3XX (range of system 2)

Under the subscriber configuration you will find the PISN subscribers you have just created, with the numbers 2XX, 3XX and 4XX.
The setting options for PISN subscribers differ from those of ordinary internal subscribers. Next use the "route" setting to allocate routes to the PISN subscribers:

1. Assign the routes for the PISN subscribers of system 1:
   – PISN subscriber 3XX: Route 6
   – PISN subscriber 4XX: route 7
2. Assign the routes for the PISN subscribers of system 2:
   – PISN subscriber 2XX: Route 5
   – PISN subscriber 4XX: route 7
3. Assign the routes for the PISN subscribers of system 3:
   – PISN subscriber 2XX: Route 5
   – PISN subscriber 3XX: route 6

Make sure the "Number" is left blank. It is used, for example , if the subscribers are to be linked up in a virtual network.

Complete numbers can also be created as PISN subscribers. This allows you to define exceptions in blocks. These numbers can be assigned names, which are then available for dialling by name. In this way you can implement dialling by name throughout the network.

Assuming Mr Newton is a subscriber in system 3, with call number 420. He is to be obtainable using dialling by name from the other systems, too. To this end, we need to create in the other systems a PISN subscriber with Mr Newton's complete call number:

1. In system 1 and system 2 create a PISN subscriber with call number 420.
2. In the subscriber configuration of both systems, assign route 7 to this PISN subscriber.
3. In the subscriber configuration of both systems, assign the name to this PISN subscriber.

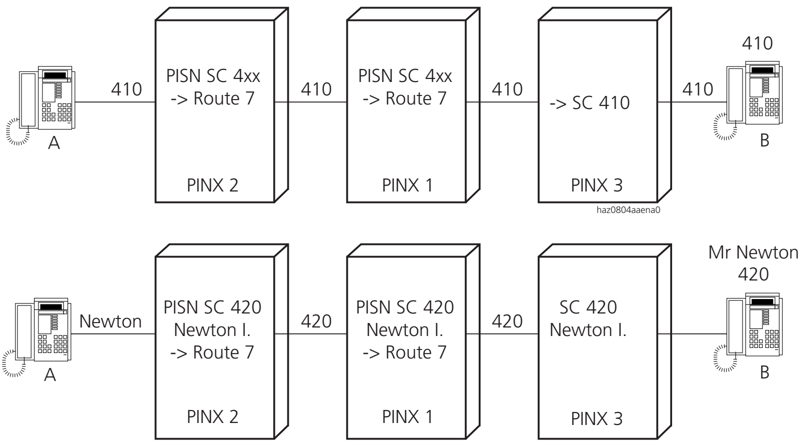


Fig. 3.72: Entering a separate PISN subscriber (example)

6.2.4.2 Numbering with regions

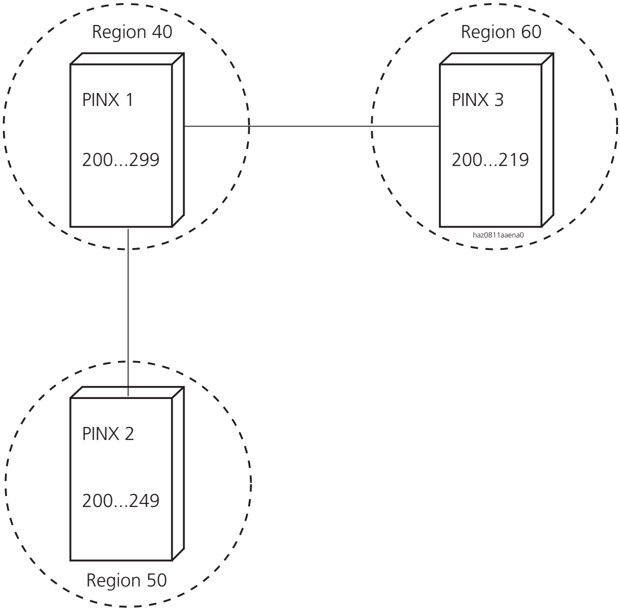


Fig. 3.73: Numbering with regions

Each system has its own numbering plan. The numbers always start with 200.

If the subscribers of one system want to call a subscriber in another system, they dial the corresponding prefix followed by the subscriber number. For example, a subscriber of system 2 wans to obtain the number 210 in system 3. He dials 60210.

A separate region has to be created for each system (similarly to the regions in the public network). To this end we need to specify the regional prefix for each system:

1. Specify the region for each system:
   – System 1: Region 40
   – System 2: Region 50
   – System 3: Region 60

2. In each system configure its own region prefix under the basic settings:
   - In System 1: Own regional prefix: 40
   - In System 2: Own regional prefix: 50
   - In System 3: Own regional prefix: 60

The further procedure depends on whether region selection is implemented via PISN subscriber or via local area network (see also the explanations on page 675). In the following we shall be looking at both types.

Region selection via PISN subscriber

The regions of all the other systems are now entered in each system. To do so create numbers in the PISN subscribers category in the numbering plan. There you can create either individual numbers or number ranges by specifying the range with "XX". Each "X" corresponds to one digit place.

1. In system 1 create the following PISN subscribers:
   - One PISN subscriber with the number 50XXX (region of system 2)
   - One PISN subscriber with the number 60XXX (region of system 3)
2. In system 2 create the following PISN subscribers:
   - One PISN subscriber with the number 40XXX (region of system 1)
   - One PISN subscriber with the number 60XXX (region of system 3)
3. In system 3 create the following PISN subscribers:
   - One PISN subscriber with the number 40XXX (region of system 1)
   - One PISN subscriber with the number 50XXX (region of system 2)

Under the subscriber configuration are the PISN subscribers you have just created, with the numbers 40XXX, 50XXX and 60XXX.

The setting options for PISN subscribers differ from those of ordinary internal subscribers. Next use the "route" setting to allocate routes to the PISN subscribers:

1. Assign the routes for the PISN subscribers of system 1:
   - PISN subscriber 50XXX: Route 6
   - PISN subscriber 60XXX: route 7
2. Assign the routes for the PISN subscribers of system 2:
   - PISN subscriber 40XXX: Route 5
   - PISN subscriber 60XXX: route 7
3. Assign the routes for the PISN subscribers of system 3:
   - PISN subscriber 40XXX: Route 5
   - PISN subscriber 50XXX: route 6

Make sure the "Number" is left blank. It is used, for example , if the subscribers are to be linked up in a virtual network.

As with the numbering method using blocks you can also create complete numbers as PISN subscribers for dialling by name.

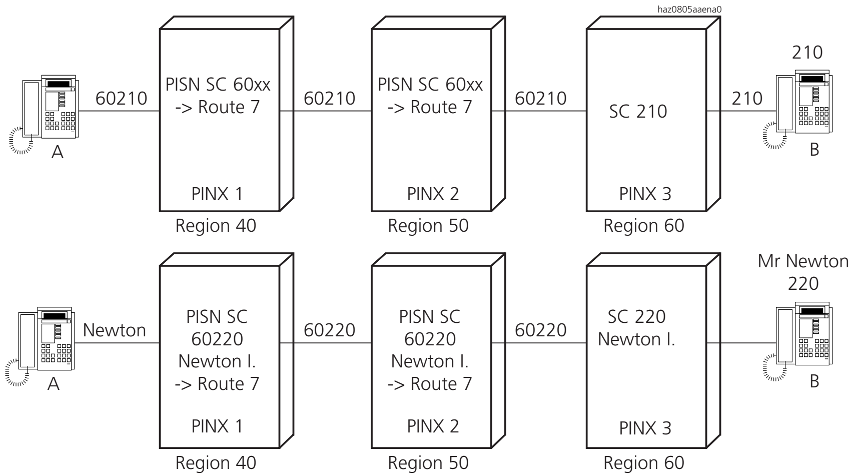


Fig. 3.74: Entering a separate PISN subscriber (example)

Region selection via local area network

With this variant, too, the regions of the other systems are entered in each system. In this case, however, a route with the region's call number is provided for each region in the numbering plan. Simply change the call number of the routes in all three systems:

1. Change the call number of the routes in system 1:
   – Route 1-6: Call number 50
   – Route 1-7: Call number 60
2. Change the call number of the routes in system 2:
   – Route 2-5: Call number 40
   – Route 2-7: Call number 60
3. Change the call number of the routes in system 3:
   – Route 3-5: Call number 40
   – Route 3-6: Call number 50

A route number prefix (exchange access prefix for the local area network) is always truncated before the call is forwarded. As with this variant the route number prefix is also the region prefix, the latter is lost. To ensure that a call can nonetheless be routed to its destination via several transit systems, the regional prefix has to be added once again.

For this enter the regional prefixes in the route configuration under "Send Access Code":

1. Enter the regional prefixes of system 1:
   – Route 1-6: Send access code = 50
   – Route 1-7: Send access code = 60
2. Enter the regional prefixes of system 2:
   – Route 2-5: Send access code = 40
   – Route 2-7: Send access code = 60
3. Enter the regional prefixes of system 3:
   – Route 3-5: Send access code = 40
   – Route 3-6: Send access code = 50

With the route selection variant you cannot create complete numbers as PISN subscribers. Dialling by name would have to be organized using abbreviated dialling. You can, however, reach any internal number of any other system, regardless of how long that number is. Assuming Mr Newton in system 3 with call number 220 has a handset with call number 5220. He can be reached by subscribers in other systems either directly on call number 60220 or on 605220.

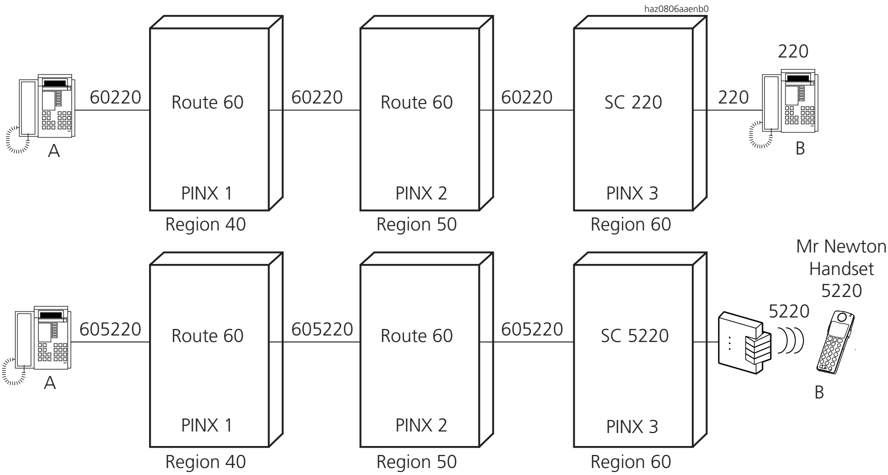


Fig. 3.75: Route selection with long call number (example)

Note:
With the local area network variant, PISN subscribers cannot be entered as destinations in the call distribution elements. That is why it cannot be used in gateway systems in which direct dialling to the public network has been organized.

6.2.5 Setting up direct dialling in

In system 1 DDI numbers are created for all the subscribers of the private network. In the corresponding call distribution elements the destination subscribers are entered for each switch position. For subscribers of systems 2 and 3, precisely the same numbers are entered as those that would be dialled by a subscriber of system 1 to call the subscribers of those systems.

The following example relates to numbering with blocks.

1. In system 1 create the DDI number range provided by your network provider. Each newly created DDI number is automatically assigned a new call distribution element.

2. In the corresponding call distribution elements (CDE) specify the destinations of the DDI numbers for the subscribers of your particular system (system 1), for example:
   - CDE 2, switch position 1 to 3 = subscriber 200
     to
   - CDE 101, switch position 1 to 3 = subscriber 299
3. In the corresponding call distribution elements (CDE) specify the destinations of the DDI numbers for the subscribers of system 2, for example:
   - CDE 102, switch position 1 to 3 = subscriber 300
     to
   - CDE 201, switch position 1 to 3 = subscriber 399
4. In the corresponding call distribution elements (CDE) specify the destinations of the DDI numbers for the subscribers of system 3, for example:
   - CDE 202, switch position 1 to 3 = subscriber 400
     to
   - CDE 301, switch position 1 to 3 = subscriber 499

See also Fig. 3.66

6.3 Networking via a public network

Virtual networking, making use of exchange-to-exchange connections, is supported by the system. This means that subscribers can still divert to a destination in the public network, forward calls or set up conferences.

In addition to these options, it is also possible to obtain that subscribers in the public network appear and are handled as network-internal subscribers. For this characteristic to function it is a requirement that the subscribers in the public network use CLIP to identify themselves. The system must use digital network interfaces to link up with the public network.

6.3.1 Tying-in an individual subscriber

A remote subscriber with one (or fewer) number(s) is to become part of a private network.

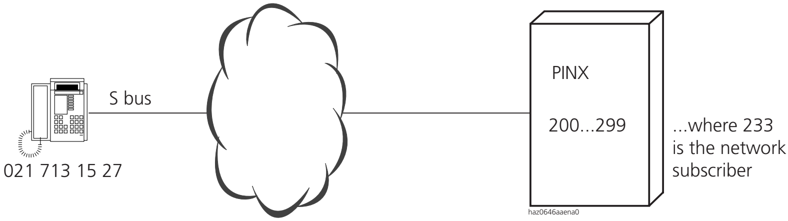


Fig. 3.76: Virtual network subscriber

The following assumptions are made:

The system's number range is 200...299 and the number 233 should not be an internal number but the remote subscriber. In other words, a call made to number 233 should ring at the remote subscriber. When the remote subscriber himself makes a call, the call should be displayed as an internal call from number 233.

Route 1 contains the lines for calls to the public network.

Configuration steps:

1. In the system, the number 233 is created as a network subscriber (category PISN subscribers in the numbering plan).
2. The subscriber's name is configured under the subscriber configuration for number 233, along with route 1 and the number 0217131527. The number must be configured in precisely the same way as the public network supplies the number when the remote subscriber makes a call.

The configuration procedure is now completed.

If an internal subscriber dials the number 233, a line from route 1 is seized and the number is dialled.

Conversely, the transmitted CLIP is compared with the configured CLIP number if the remote subscriber dials a number in the system (normally a DDI number). If the numbers match up, the number 233 and the caller's name will be displayed to the called subscriber as the CLIP.

Note:
The incoming CLIP analysis only works once you have exited the configuration.

6.3.2 Networking two systems

The procedure described above can also be used for networking two systems via the public network.

The subscribers of a system need to have the appropriate DDI numbers so they can be reached directly.

For the correct CLIP to be displayed, each subscriber of the other system has to be entered individually as a PISN subscriber.

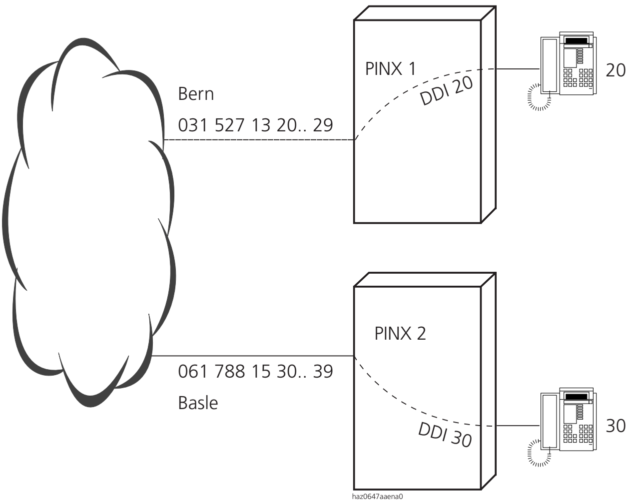


Fig. 3.77: Two systems in a virtual network

System 1 with the DDI range 031 52713 20...29 is located in Bern. The internal numbers 20...29 can be reached with DDI.
System 2 with the DDI range 061 788 15 30...39 is located in Basle. The internal numbers 30...39 can also be reached with DDI.
The subscribers in Bern want to be able to reach the subscribers in Basle as if they were internal subscribers, using the numbers 30...39; vice versa, the subscribers in Basle want to be able to reach those in Bern with the numbers 20...29.

The direct dialling plan for both systems and the internal subscribers are configured. The following configurations are used for this purpose:

1. In System 1: in the numbering plan create the number 3x as PISN subscriber.
2. For PISN subscriber 3x enter the route and the number 061788153x.
3. In System 2: in the numbering plan create the number 2x as PISN subscriber.
4. For PISN subscriber 2x enter the route and the number 031527132x.

If subscriber 20 now calls the number 30, the call is set up via the public network using network number 061 788 15 30 and routed to subscriber 30 in system 2 via DDI. Subscriber 30 see the number 20 as the CLIP as subscriber 20 has identified himself in the public network with CLIP 031 527 13 20. This number is implemented in system 2 and displayed as internal subscriber 20, including name and internal ringing.

Note:
The incoming CLIP analysis only works once you have exited the configuration.
If individual subscribers are to be reached using dialling by name, the corresponding PISN numbers need to be created individually and provided with names.

6.4 Networking with a virtual PBX (Centrex)

The link-up of a virtual PBX in the public network is implemented in exactly the same way as the networking of two systems.

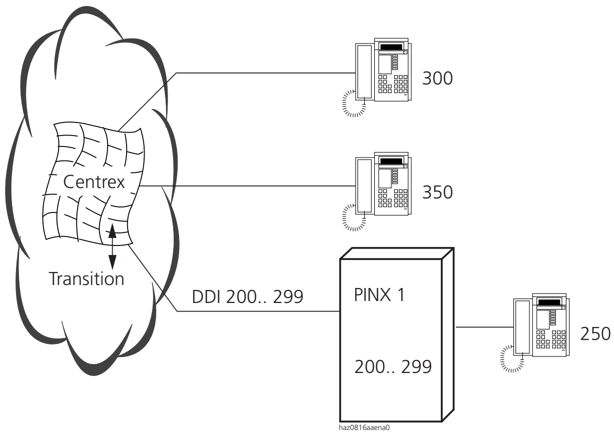


Fig. 3.78: Networking with Centrex

System 1 is connected with Centrex. A DDI range 200...299 is defined there.

I. e., Centrex regards all numbers 200...299 as internal numbers in the Centrex range in system 1.

The Centrex range also contains other internal numbers, e.g. 300 and 350.

If subscriber 300 wants to reach subscriber 250, he simply dials 250.

If subscriber 300 wants to reach a subscriber in the public network, he dials 0 (exchange access prefix) followed by the subscriber number, e.g. 032 6241399. For the operation to be exactly the same for a user of system 1, you need to set up the following configurations.

1. All the subscriber numbers in the Centrex range, i.e. 300 and 350, are created as PISN subscribers in system 1.
2. A route is defined (route 2) with the trunk group (trunk group 1) that contains all the exchange lines to the Centrex. PISN subscribers 300 and 350 are allocated the corresponding route (route 2). The protocol in the trunk group must be configured to DSS1 and the type to "Public".
3. An additional route (route 1) is defined for calls to the public network. This is the route configured as the route for exchange output for internal subscribers. Trunk group 1 is also configured in this route.

4. In addition, however, the parameter "Send Access Code" is configured with digit "0" in route 1. This is the exchange access prefix for the Centrex.

If a subscriber in system 1 dials the exchange access prefix 0 followed by the public number, e.g. 032 6241399, system 1 will seize a line from route 1 and immediately dial the digits entered under "Send Access Code", i.e. in this case "0". Only then does the public number follow. Centrex sets up a connection to the public network.

Note:
Calls from the public network reach system 1 via Centrex. The CLIP number is preceded by an exchange access prefix as system 1 is simply an internal subscriber from Centrex's point of view. For the CLIP to be correctly displayed on system 1, the exchange access prefix needs to be deleted again from the PBX. To this end, "Truncate CLIP = 0" is entered in the corresponding trunk group. This means that when a CLIP number beginning with "0" is received, the "0" is truncated.

6.5 Networking with third-party systems

In principle all network-compatible third-party systems that support the QSIG standard can be networked with the system. However, some restrictions in the features available are possible. A number of points need to be taken into account for a third-party system to communicate correctly with the system.

6.5.1 Compatible QSIG protocol

In practice two different QSIG protocols are used, QSIG and QSIG / PSS1 ISO. The two variants differ only with regard to the communication technology, not the scope of performance. The system supports both versions. They can be selected under the "Protocol" trunk group setting. Make sure you always use the same version between two systems.

6.5.2 Outgoing calls via a third-party system

There are instances where third-party systems do not recognize a call to be forwarded to the public network based on its numbering plan identifier. For this reason an exchange access prefix for the third-party system should be sent with the call (see variant 2 on page 687).

6.5.3 Incoming calls via a third-party system

The system uses the call number’s numbering plan identifier to determine whether an incoming call is to be forwarded to the public network (NPI = E.164) or within the internal network (NPI = PNP). However, some third-party systems use only NPI = E.164 in general. To ensure that the system is nonetheless able to route the call correctly you need to proceed as follows:

1. Set up the third-party system in such a way that outgoing calls to the public network are not routed via the same lines as outgoing calls that remain within the private network.
2. In the system set up two trunk groups for the lines to the third-party system:
   - In the first trunk group combine all the lines via which calls to be forwarded on to the public network are received.
   - In the second trunk group combine all the lines via which calls to be forwarded on within the private network are received.
3. Set the parameters of the first trunk group as follows:
   - Protocol = PSS1
   - Network type = private
   - Overwrite NPI = no
4. Set the parameters of second first trunk group as follows:
   - Protocol = PSS1
   - Network type = private
   - Overwrite NPI = PNP

6.5.4 Incorrect CLIP indication

Third-party systems may not use the numbering plan identifier of a CLIP number to create a correct CLIP. This is why it is possible that the CLIP number of a call from a third-party system is not displayed correctly.
The settings need to be adapted accordingly for such cases (see "Identification Elements", page 178).

6.6 Cordless systems in a private leased-line network

The DECT cordless system is tied to a single system. The radio area cannot be increased by networking several systems.

6.7 Abbreviated dialling and virtual network subscribers

In the past virtual network subscribers were set up with abbreviated-dialling numbers. While the same procedure can still be used, setting up a PISN subscriber offers a number of significant advantages:

- An incoming call from a virtual network subscriber is signalled internally and an internal number is displayed as the CLIP.
- A separate route can be defined for the outgoing routing.

Part 4 Installation

1 Overview of Chapters

Mounting the PBX

Chapter 2 explains the site conditions that need to be taken into account for the PBX installation. This is followed by a description of the two possibilities for PBX installation: Wall mounting or mounting in a 19” system cabinet.

Earthing and protecting the PBX

Chapter 3 describes the requirements for earthing the PBX. Other topics include connecting the cable screening and using surge voltage protectors.

PBX power supply

The PBX is powered as standard with 230 VAC. Chapter 4 tells you how to ensure an uninterruptible power supply, the modules required for a 12 VDC or 48 VDC power supply, and how to increase the available supply power for the terminals connected.

Equipping the PBX

After the installation, the system's operability is set up by fitting system and expansion cards. Chapter 5 tells you how many card slots are available on each basic systems and how the basic systems are fitted with network cards, subscriber cards and special cards.

Connecting the PBX

The PBX can be connected to the mains either directly or indirectly, i.e. via a main distribution board. Chapter 6 lists all the points to be observed when connecting the PBX and the requirements that a self-built 16-wire connecting cable has to comply with.

Cabling interfaces

Chapter 7 contains a description of the ports on the mainboard and their addressing as well as the wiring and connection of the various interface types. Digital and analogue network interfaces, digital and analogue user-network interfaces and special interfaces for special uses are described in detail.

Installing terminals

Chapter 8 contains information on how to install corded and cordless terminals of the Office family and their supplementary equipment.

Checking the installation

Chapter 9 has a check-list for verifying the installation in order to prevent malfunctions.

2 Mounting the PBX

The PBX can be both wall-mounted or installed vertically inside a 19" system cabinet. A special rack version is available for horizontal mounting in a 19" system cabinet (see "Rack version Ascotel IntelliGate", page 712).

2.1 PBX location

When positioning the PBX take note of the location requirements in Tab. 4.1.

Tab. 4.1: PBX location requirements

	Location requirement
Location in general	• Easily accessible and with adequate lighting
Heat radiation	• Do not position the PBX in direct sunlight or near heating sources
EMC	• Do not position the PBX in strong electromagnetic fields of radiation (e. g. near x-ray equipment, welding equipment, etc.)
Convection	• Observe the clearance requirements with the PBX (see Fig. 4.1 or Fig. 4.2) • Provide ventilation slots of approx. 600 cm^2 each at the top and bottom of built-in cabinets
Environment	• Room temperature 5...35 °C • Relative humidity 30.0.80%, non-condensing

2.2 Wall mounting

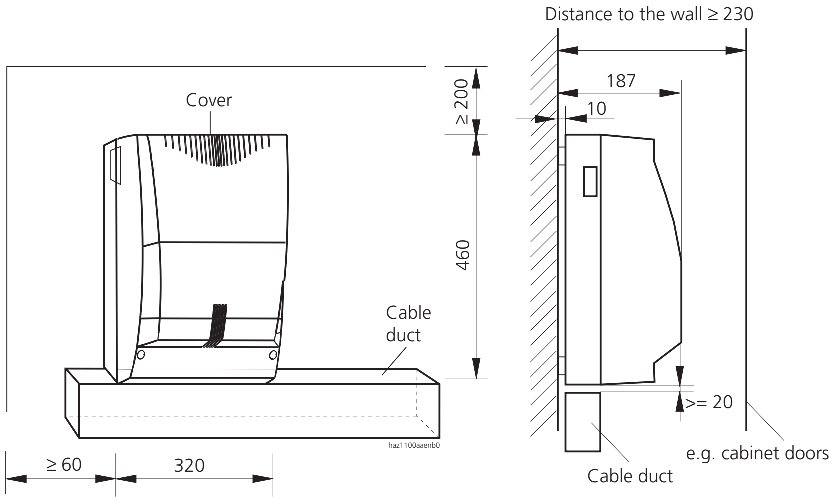


Fig. 4.1: Dimensional drawing for basic system 2025 / 2045

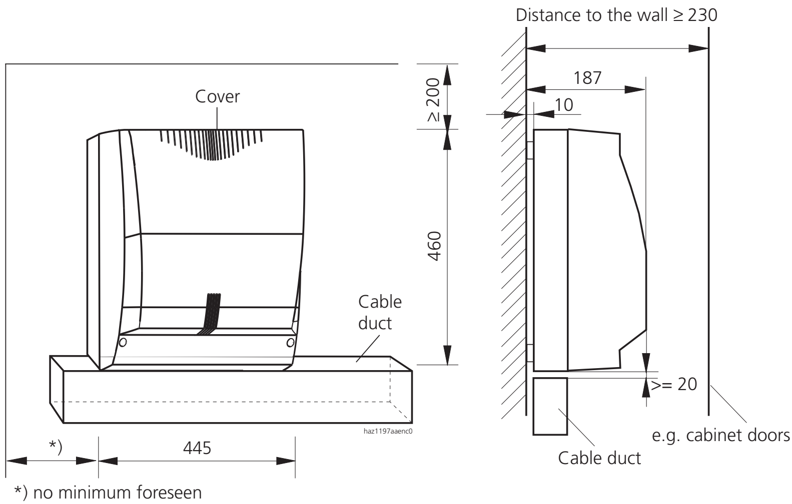
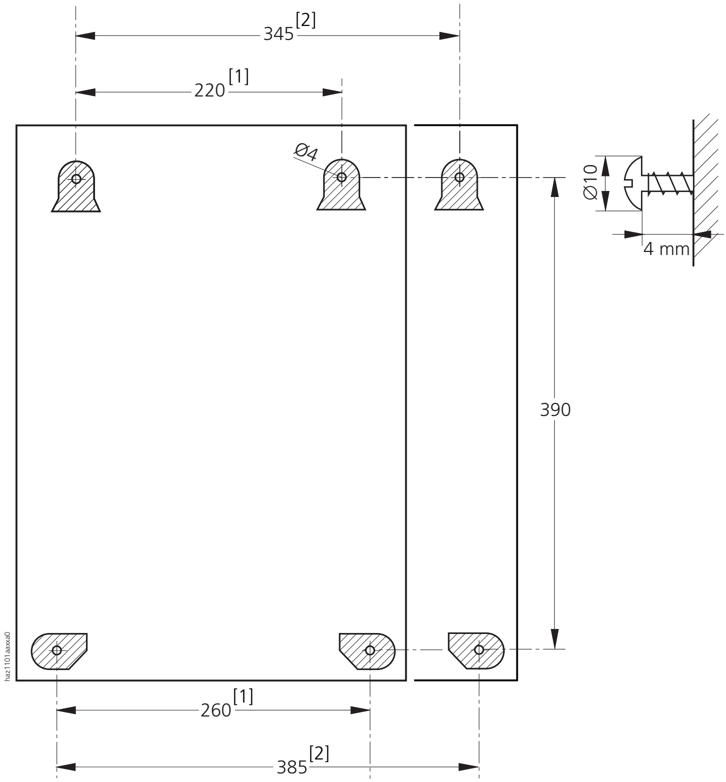


Fig. 4.2: Dimensional drawing for basic system 2065



[1] Dimensions of the basic system 2025 / 2045
[2] Dimensions of the basic system 2065

Fig. 4.3: Dimensional drawing for wall-mounting of the PBX

To mount the basic system on a wall, proceed as follows:

1. Mark out the position of the 4 screws as shown in the dimensional drawing for the wall mounting of the PBX (see Fig. 4.3).
2. Fit the screw dowels. Take note of the weight and technical data (see "Basic systems", page 974).
3. Fit the upper suspending screws.
4. Suspend the PBX housing onto the suspending screws, with the mainboard preinstalled and the cover open.
5. Use the screws to secure the lower half of the housing to the wall.
6. Fit the cover.

2.3 Cabinet mounting (Vertically in 19" system Cabinet)

The mounting set is used to install the PBX vertically inside a 19" system cabinet.

When using a system cabinet, take note of the following:

- The system takes up the space of 12 height modules inside the system cabinet. The depth is variable.

The following installation variants are possible:

- Behind patch panel, accessible from the front
- Behind dummy plates, accessible from the front
- Free and accessible from the rear
- Others where appropriate

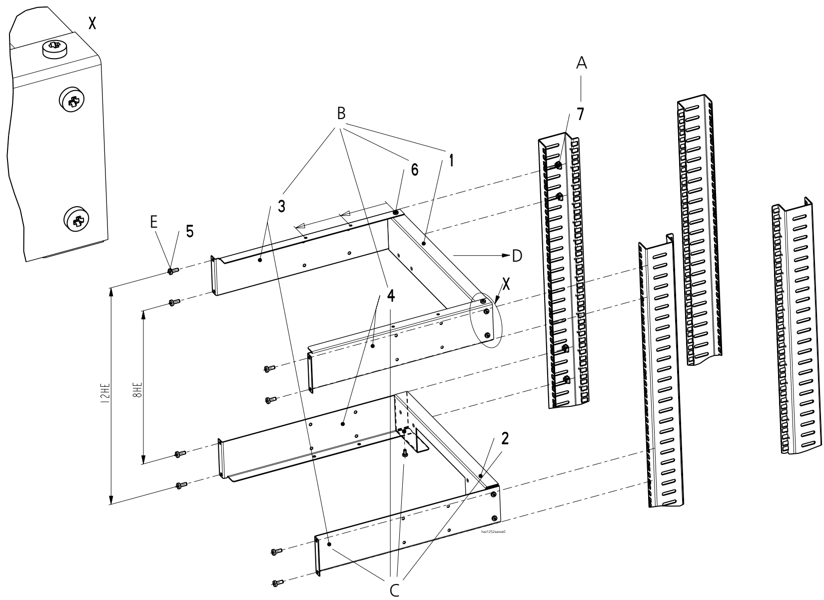


Fig. 4.4: Assembly installation set

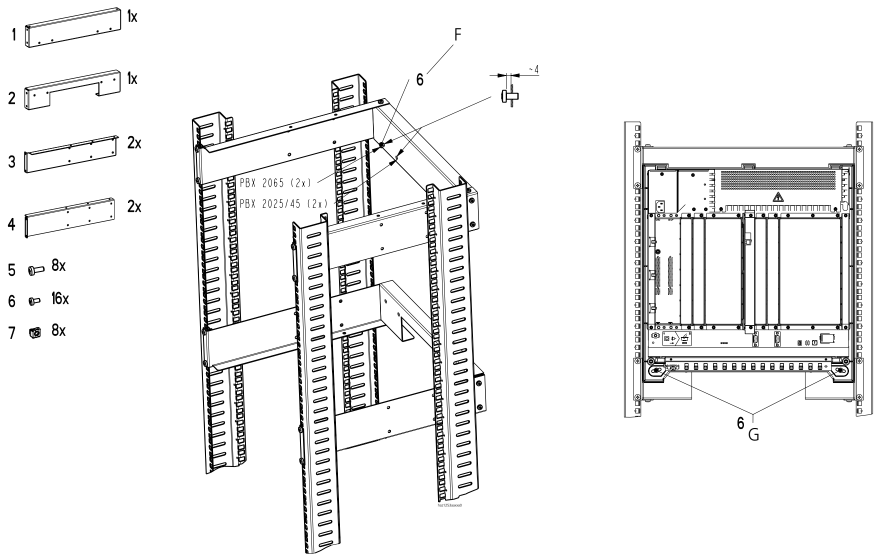


Fig. 4.5: Accessories for installation set and installation of the PBX in a 19" system cabinet

Installing the PBX into a 19" system cabinet

1. Step A: 8 Fit 8 cage nuts [7] (see Fig. 4.4). The space between the rails should be 8 height modules on the inside (or 12 height modules on the outside).
2. Step B: Premount upper plug-in unit with parts 1, 3, 4, 6 (see Fig. 4.4).
3. Step C: Premount lower plug-in unit with parts 2, 3, 4, 6 (see Fig. 4.4).
4. Step D: Slide the mounted plug-in units into the system cabinet.
5. Step E: Use 8 screws [5] to screw the plug-in units to the system cabinet (see Fig. 4.4).
6. Step F: 2 Loosely screw screws [6] approx. 4 mm into the upper back rails [1] (see Fig. 4.5):
   For 2025 / 2045 system use an internal thread, for 2065 system an external thread.
7. Step D: Suspend the PBX into the loose screws and secure to the lower back rail [2] with 2 screws [6].

2.4 Horizontal mounting

Under certain circumstances a PBX can also be mounted horizontally (e.g. lying inside a 19" cabinet). The PBX location requirements in Tab. 4.1 are to be observed. Particular attention has to be paid to convection (provide fan if required).

Warning:
For heat reasons the basic system 2065 can only be fitted horizontally if it is operated together with the external auxiliary terminal power supply (ATPS) (or when using the rack version, with an integrated fan).

2.5 Rack version Ascotel IntelliGate

With the greater IP integration of Ascotel IntelliGate systems all systems are also available in rack versions. The scope of performance and functionality do not differ from those of systems of the ”wall mounting” type. This means that all the information contained in the Ascotel IntelliGate system documentation is also valid for the rack versions. The extra letter "R" in the designations indicates a rack version.

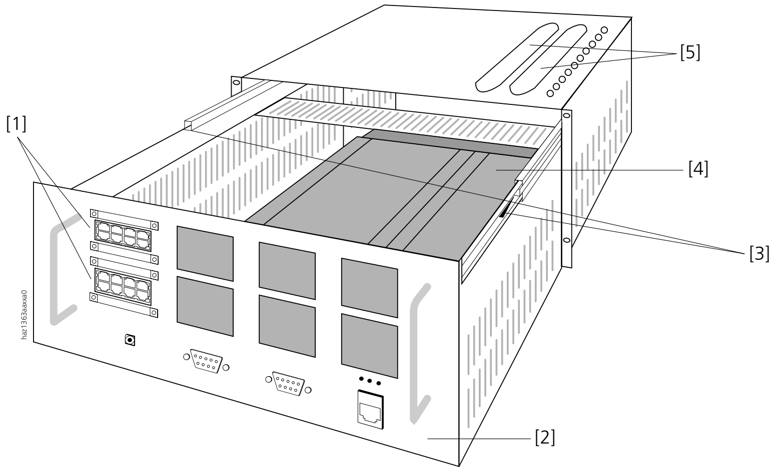
Tab. 4.2: Designations of rack versions

Basic systems designation	Systems designation
2025R / 2045R basic system	Ascotel IntelliGate 2025R / 2045R
2065R basic system	Ascotel IntelliGate 2065R

Unlike the mounting set used for installing a PBX vertically into a 19" cabinet, the rack versions are genuine 19" plug-in units for horizontal installation, with the following advantages:

- Only 5 HM (height modules) instead of 12 HM on the mounting set
- All user-network, network and special interfaces on the basic systems and expansion cards are accessible on the front panels and can be designated with labels.
- The rack plug-in unit with the integrated Ascotel IntelliGate system is designed so that the fully cabled system can be removed from the rack plug-in unit like a drawer, providing full access to the control elements such as the HEX rotary switch, Reset button, EIM card, etc. If required, the drawer can be completely removed from the rack plug-in unit by releasing the side locks (see Fig. 4.6).

- It is also possible to replace expansion cards with the system pulled out. Simply disconnect the connections to the expansion cards that need to be replaced. Make sure you disconnect the system from the power supply first (see "Installing and removing cards", page 904).
- The 2065R basic system is supplied with an integrated fan, which is powered by a plug-in power supply unit (supplied).

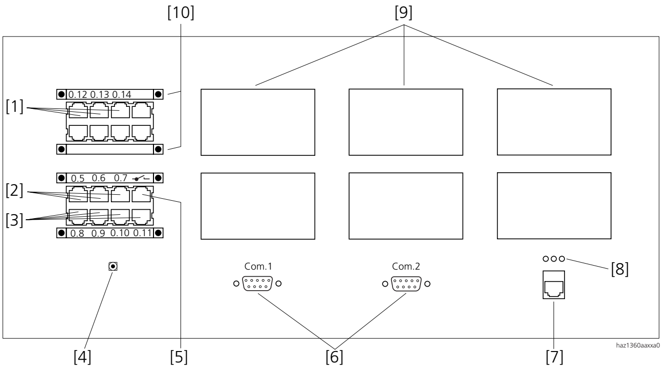


[1] Cabled interfaces of the mainboard with designation labels
[2] Basic front panel
[3] Locks on the rack plug-in unit
[4] Basic system 2025R / 2045R
[5] Apertures for cabling

Fig. 4.6: Ascotel IntelliGate 2025R / 2045R in its basic configuration

Basic Front Panels

The basic front panel is fitted differently depending on the system. The interfaces between the expansion cards and the basic front panel are connected using prefabricated 16-wire cables, which are available in lengths of 0.5 or 2 m



[1] S/T interfaces (2025R / 2045R only)
[2] a/b interfaces (2025R / 2045R only)
[3] AD2 interfaces (2025R / 2045R only)
[4] Audio interface
[5] Potential-free relay contact for a general bell (2025R / 2045R only)
[6] V.24 interfaces
[7] Ethernet interface
[8] Visual displays Ethernet interface
[9] Dummy covers
[10] Labels

Fig. 4.7: 2025R / 2045R / 2065R basic front panel

Basic Front Panel on the 2025R / 2045R Basic System

The basic front panel on the 2025R / 2045R basic system incorporates not only the pre-wired interfaces of the mainboard but also all the interfaces of any expansion cards. Depending on the system's configuration and expansion different connection modules may be fitted behind the empty openings provided (see "Connection Modules", page 716).

Interfaces on the 2025R / 2045R basic front panel:

- Visual displays:
  – Ethernet LINK, RX, TX
- Precabled interfaces:
  – 1 Audio interface
  – 2 V.24 interfaces
  – 1 Ethernet interface
  – 3 S/T interfaces
  – 4 AD2 interfaces
  – 3 a/b interfaces
  – 1 Potential-free relay contact for general bell

- 6 openings for cabling:
  - A total of 32 a/b, AD2, S-bus interfaces
  - A total of 4 T2 interfaces, Ethernet (IP interface cards)
  - Interfaces on the OI-2DOOR expansion card

Basic Front Panel on the 2065R Basic System

On the basic front panel of the 2065R basic system the standard mainboard interfaces (with the exception of the general bell relay) and the Ethernet interface of the processor card are cabled. 8 x 8 interfaces of expansion cards can be wired to the basic front panel of the 2065R basic system using the corresponding connection modules. Two expansion front panels are available for additional expansion stages.

Interfaces on the 2065R basic front panel:

- Visual displays:
  - Ethernet LINK, 100M, COLL
- Cabled interfaces:
  - 1 Audio interface
  - 2 V.24 interfaces
  - 1 Ethernet interface
- 8 openings for cabling:
  - 1 Potential-free relay contact for general bell
  - Other interfaces of expansion cards

Expansion Front Panels

The following expansion front panels are available for the cabling of other network, user-network and special interfaces of the 2065R system:

- Expansion front panel for 32 interfaces (1 HM).

- Expansion front panel for 64 interfaces (3 HM). The front panel can be folded up vertically.

Additional 19" Plug-in Unit

A separate drawer-type 19" plug-in unit is available for additional hardware and expansions (2 HM). The plug-in unit can be used for example for:

- NT converter
- LAN switch
- other

Connection Modules

Different connection modules can be fitted depending on a system's configuration and expansion. They are installed behind the openings provided in the front panels. Unused openings in the front panels can be closed using dummy covers. The connection modules and the dummy covers can be supplied singly. There are three different types of connection modules:

ab/AD2 Connection Module

The ab/AD2 connection module contains 8 RJ45 sockets for 8 a/b and/or 8 AD2 interfaces. The wiring to the expansion cards consists of one prefabricated 16-wire connecting cable or 4 quad two-conductor pillar terminal blocks. The second connection variant allows a/b and AD2 interfaces to be wired to the same connection module. The assignment of the 16-wire connecting cable, and also of the 16-pin cable on the connection module, is described in the Chapters "User-network interfaces a/b", page 783 and "AD2 user-network interfaces", page 766.

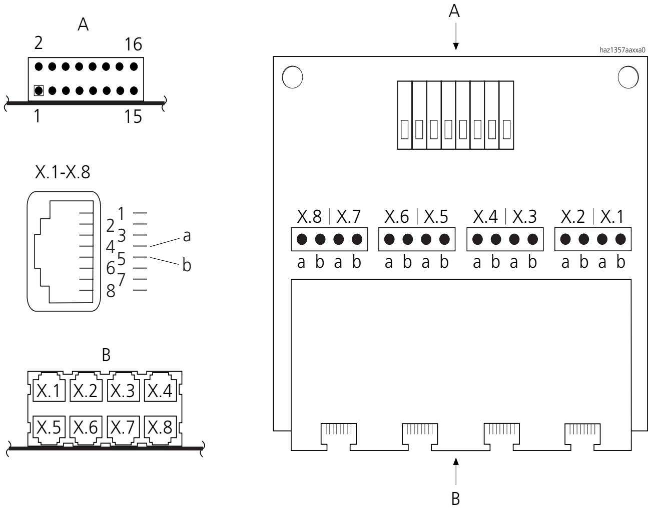


Fig. 4.8: **ab/AD2 Connection Module**

S/T Connection Module

The S/T connection module contains 8 RJ45 sockets for 8 S and/or 8 T interfaces. The wiring to the expansion cards consists of two prefabricated 16-wire connecting cables or 4 quad two-conductor pillar terminal blocks. Each connection module features two 16-pin sockets for S and T interfaces to accommodate the different wiring of the S and T interfaces. This means that the combinations 8 x S, 4 x T/4 x S or 8 x T are all possible with the 16-wire connecting cables. As the S/T interfaces on the mainboard and expansion cards are individually switchable, individual interfaces can also be wired to the connection module via pillar terminal blocks. The assignment of the 16-wire connecting cable, and also of the 16-pin S and T sockets on the connection module, is described in the Chapters "S user-network interfaces", page 776 and "Basic access T", page 752.

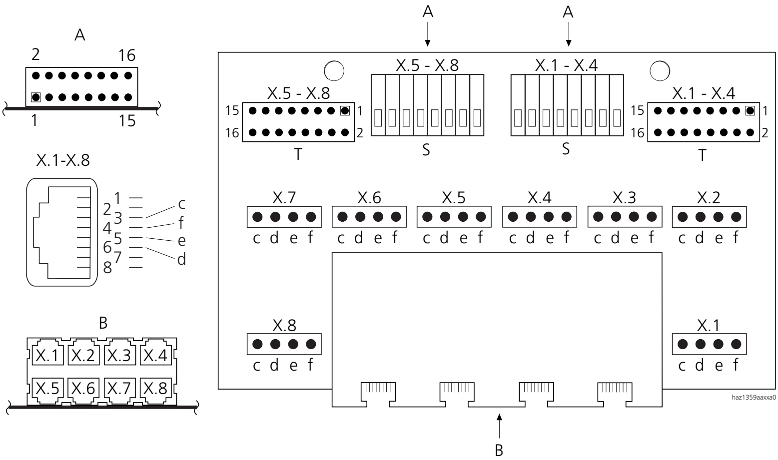


Fig. 4.9: S/T Connection Module

LAN/PRA Connection Module

The front of the LAN/PRA connection module features 8 RJ45 screened sockets, with in each case the superimposed sockets connected in parallel. 4 RJ45 screened sockets for 4 PRA and/or 4 Ethernet interfaces are available for the wiring to the expansion cards. In each case the interfaces are wired to the expansion cards using commercially available screened cables.

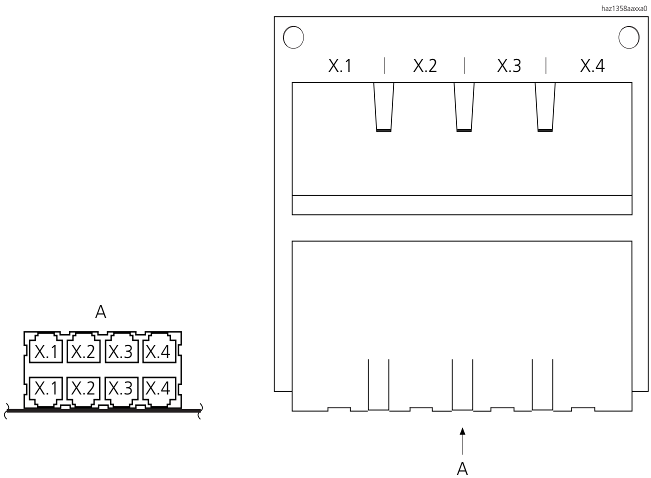


Fig. 4.10: LAN/PRA Connection Module

Note:
Only one of the two superimposed sockets connected in parallel may ever be used at any given time.

3 Earthing and protecting the PBX

The protective earth and equipotential bonding are important integral parts of the PBX's safety concept: Standard EN 60 950 relevant to safety matters stipulates protective earthing.

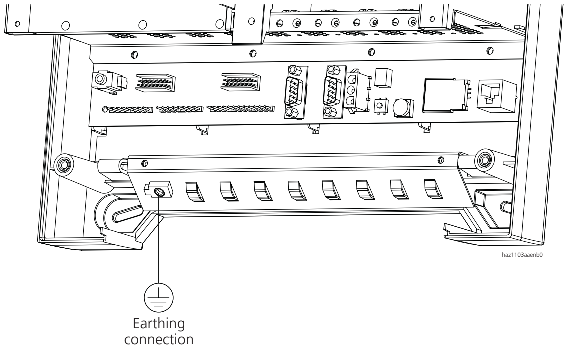
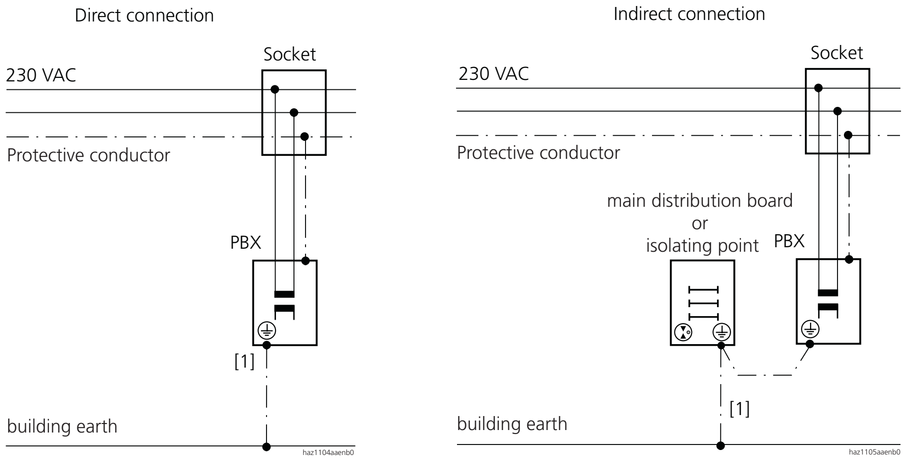


Fig. 4.11: Earthing connection to the PBX (basic system 2025 / 2045)

The position and appearance of the earthing connection on basic system 2065 are in accordance.

3.1 Earthing

Make sure you install the earthing carefully in accordance with Fig. 4.12.



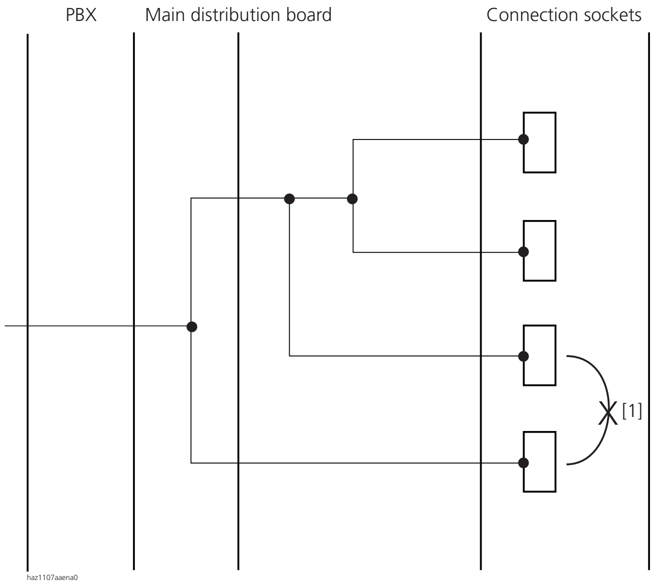
[1] Earthing (copper wire 2.5 mm^2, yellow / green)

Fig. 4.12: Earthing for direct connection (left) and indirect connection (right) of the building cable installation

Note:
Earthing in a connection to a universal building cable installation corresponds to earthing in a connection via main distribution board.

3.2 Connecting the cable screening

Connect the cable screens to one another at the splitting point only. Observe the tree structure principle to prevent earth loops.



[1] No earth loops

Fig. 4.13: Tree structure principle

1. Exposing the screening: Strip off the plastic outer sheath over a length of 30-40 mm in the area of the connecting terminals for cable ties.
2. Wrap the tracer tightly about 5 times around the extremity of the screen (Fig. 4.14).
3. Contact the screen with the housing: Use a cable tie to secure the cable section over a large area to the intended contact surface with the screening exposed (Fig. 4.15).

Tip:
Secure several small-diameter cables together in a bundle.

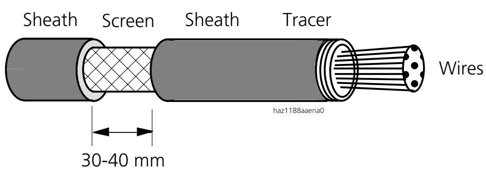
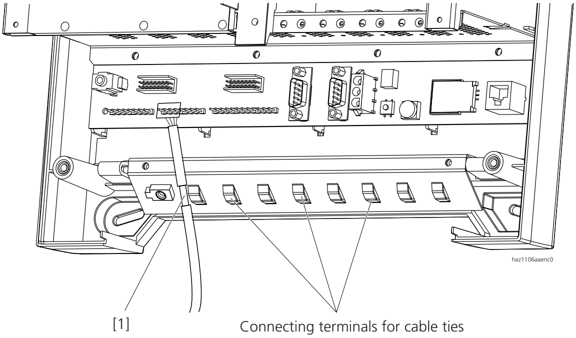


Fig. 4.14: Cable screening



[1] Exposed cable screening in the area of the connecting surface

Fig. 4.15: Connecting terminals for cable ties on the PBX (basic system 2025 / 2045)

The position and appearance of the connecting terminals for cable ties on the 2065 basic system are in accordance.

3.3 Installing a surge voltage protector

Protect each line installation leading from the building at the PBX location by using one surge voltage protector per core at the isolating point (main distribution frame or entry point into the building.

Surge voltage protector characteristics:

- Response voltage, static: 245 VDC
- Igniting voltage at 1 kV/µs: < 800 V
- Discharge current (impulse 8/20 µs): 10 kA

4 PBX power supply

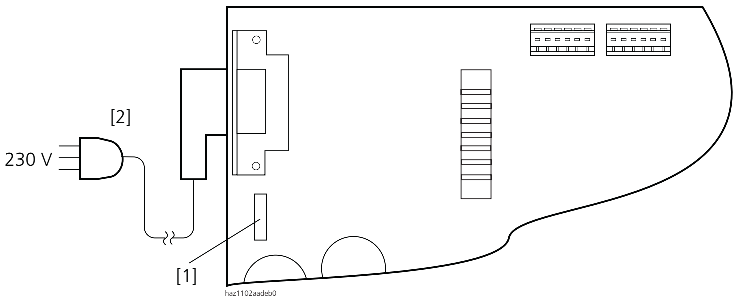
The standard PBX power supply is 230 VAC.

For an uninterruptible power supply (UPS) or for powering the PBX from a 12 VDC source, the UPS-12V module must be installed.
If the PBX is to be powered from a 48 VDC source, a DC-48V module has to be installed.

The external Auxiliary Terminal Power Supply (ATPS) is available to increase the supply power available for terminals.

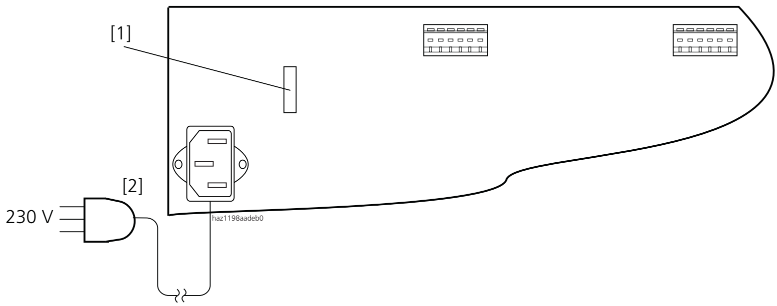
4.1 230 VAC power supply

The PBX is equipped with a socket-outlet and is connected to a 230 VAC socket-outlet using a mains cord.



[1] Power circuit breaker
[2] Mains cord (length 2m)

Fig. 4.16: Power supply unit socket and power supply area main board 2025 / 2045



[1] Power circuit breaker
[2] Mains cord (length 2m)

Fig. 4.17: Power supply unit socket and power supply area main board 2065

4.1.1 Uninterruptible power supply (UPS)

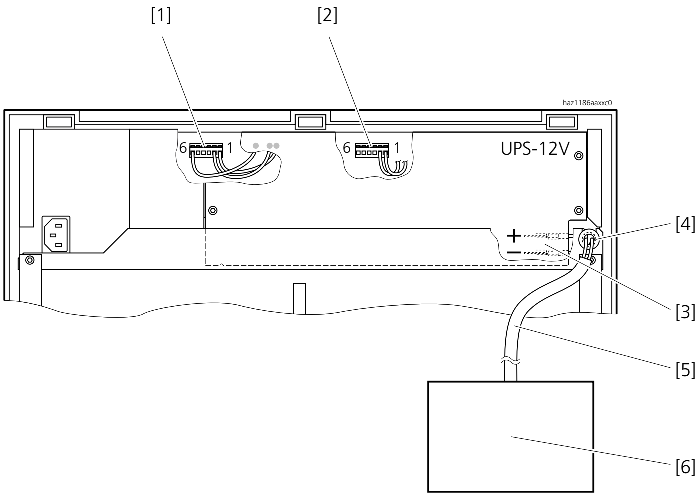
The use of an uninterruptible power supply (UPS) guarantees that the PBX can be operated even in the event of a failure of the 230 V power supply.

Uninterruptible power supply can be provided by the UPS-12V module or an external UPS.

4.1.1.1 UPS with UPS-12V module

The charging circuit of the UPS-12V module is designed for lead-acid batteries with a floating charge voltage of approx. 13.6 V. The charge current can be limited to 1 or 2 A (jumper on the module). The battery switching circuit is protected on the module with 20 A (Wickmann FK1 20 A). If the polarity is the wrong way around, the fuse breaks the circuit and must then be replaced.

If a battery isolating point is provided, it must be equipped to handle a current of 20 A (at 12 VDC).



[1] Primary connector
[2] Secondary connector
[3] Cable clamp (Fast-ON 6.3 mm)
[4] Rubber grommet
[5] Cable to 12V back-up battery (cable cross-section 4 mm^2)
[6] 12V back-up battery (external)

Fig. 4.18: Connecting the UPS-12V module

Fitting and connecting the UPS-12V module

Hazard:

Once the PBX is connected to the 230 V mains or the UPS-12V module is connected to the battery, there are hazardous voltages inside the PBX housing. Make sure the cover in the power supply area is securely closed before connecting the battery.

1. Disconnect the PBX from the power supply.
2. Remove the cover of the power supply area.
3. Mount the angle bracket with the provided screws.
4. With the jumper on the UPS-12V module, select the required charge current limiting (1 A or 2 A).
5. Fit the battery cable without battery connected to the UPS-12V module. Make sure the polarity is correct!

6. Connect the UPS-12V module with the PBX. To do that connect the cable to the primary connector and secondary connector (see Fig. 4.18).
7. Use the provided screws to fasten the UPS-12V module to the angle bracket.
8. Secure the cover to the power supply area.
9. Connect the PBX to the 230 VAC power supply.
10. Connect the battery. Make sure the polarity is correct!
11. Check the UPS functions by interrupting the 230 VAC power supply for a few minutes. The PBX must continue to run without interruption. The event message "PBX Power Supply: Battery" should appear after a few seconds (configurable).

Note:
When disconnecting a system with a UPS-12V module from the power supply, the battery must always be first disconnect. When connecting, always connect the mains voltage first and then the battery.

The battery capacity is rated according to the power requirements and the required bridging time of the PBX.

Tab. 4.3: Power requirements of the PBX and battery rating (12 V)

Basic system	Maximum power requirements for PBX (+ ATPS)		Battery capacity (Ah)[1] for stored energy time		
	OC	W	1 h	4 h	12 h
Ascotel IntelliGate 2025 / 2045	110	70	10	30	90
Ascotel IntelliGate 2025 / 2045 + ATPS[2]	180	140	20	60	180
Ascotel IntelliGate 2065	210	135	20	60	180
Ascotel IntelliGate 2065 + ATPS[2]	460	420	60	180	540

1) Maximum configuration and max. traffic volume. Typical conditions require approx. 60% of the specified battery capacity

2) Only one external USP is possible when using the ATPS

Note:
It is also possible in principle to supply the PBX with the UPS-12V module via a centralized 12 VDC system.

4.1.1.2 External UPS

The uninterruptible power supply is selected in accordance with the power requirements of the PBX (see Tab. 4.3).

If a 12V device is used, the required battery capacity (Ah) can be also taken from Tab. 4.3. In the case of terminals with a 24 V or 48 V battery, the battery capacity values indicated in are to be divided by 2 and 4 respectively.

Uninterrupted operation of the PBX is ensured if the UPS takes control of the power supply within the times indicated in Tab. 4.4:

Tab. 4.4: UPS: Maximum admissible making capacity for the UPS

Mains voltage 230 VAC	Max. making capacity of the UPS
Interruption	< 30 ms
Drop to < 80%	< 30 ms

See also:
For more technical details on the PBX see "Technical Data", page 974.

4.2 48 VDC power supply

To operate the system with 48 VDC, the DC-48V module has to be fitted. This module is connected in the same way as the UPS-12V module (see Fig. 4.18). The DC input is secured on the module with a 5 A special fuse (Wickmann FUN125 5A). If the polarity is the wrong way around, the fuse breaks the circuit and must then be replaced. Unlike the UPS-12V module, here a power supply cable with a cross section of 2.5 mm^2 is sufficient.

Fitting and connecting the DC-48V module

Hazard:
Once the 230 V mains is connected or the DC-48V module is connected to the DC power supply, there are hazardous voltages inside the PBX housing. Make sure the cover in the power supply area is securely closed before connecting the power supplies.

1. Disconnect the PBX from the power supply.
2. Remove the cover of the power supply area.
3. Mount the angle bracket with the provided screws.

4. Fit the power supply cable without connected battery to the DC-48V module. Make sure the polarity is correct!
5. Connect the DC-48V module to the PBX. To do that connect the cable to the primary connector and secondary connector (see Fig. 4.18).
6. Use the screws provided to fasten the DC-48V module to the angle brackets.
7. Secure the cover to the power supply area.
8. Connect the DC power supply. Make sure the polarity is correct!

If an isolating point is provided for the 48 VDC, it must be equipped to handle a current of 5 A (with 48 VDC).
The PBX with the installed DC-48V module can also be operated on the 230 VAC power supply.

4.3 Supply power available for terminals

The 40V power supply required forthe connected terminals is rated for the power requirements of a typical system configuration. The power available is 28W for the basic system 2025 / 2045 and 70W for the 2065 system. To check the power requirements of an existing system refer to the Tab. 3.22 for details of the average power requirements of terminals. The check can also be carried out automatically using the Planning Manager, which can be started from AIMS (see "Project Manager", page 148).

4.3.1 Overload shutdown

A red LED (OVLD) on the connector strip above the 7-segment display is used to signal that the rated power (100%) of the terminal power supply has been exceeded (see Fig. 4.20). If the rated power is exceeded for more than approx. 4 seconds, the event message "Terminal Power Supply Overload" is generated.

If the rated power is exceeded by 25% for more than approx. 4 seconds, the event message "Terminal Power Shutdown" is generated. The power supply is then shut down step by step, starting with the expansion slots with the highest numbers.

Once the power demand drops back below 100% as a result of the gradual shutdowns, the disconnected ports are reconnected again after approx. 10 seconds. (Once power is restored, it can take up to 90 seconds for terminals that were in an active connection at the time of the shutdown to be up and running again.) If the limit of 125% is again exceeded, the overload shutdown will trigger once again.

If an overload occurs, either reduce the required supply power (e.g. by powering DECT radio units locally) or use the external Auxiliary Terminal Power Supply (ATPS).

4.3.2 External Auxiliary Terminal Power Supply (ATPS)

An external 40V Auxiliary Terminal Power Supply (ATPS) has to be used if the internal power supply is no longer sufficient based on the power requirements calculated for powering the terminals or due to event messages signalling a supply overload. The ATPS provides a supply power of 250W for the terminals. The maximum terminal supply requirements for a full configuration of the 2065 system is approx. 180 W, or approx. 50 W for the 2045 system.

Note:
It is not possible to use ATPS and a DC supply module (UPS-12V or DC-48V) at the same time. An external UPS or an inverter is required to be able to ensure uninterruptible operation when using an ATPS system.

Mouting the ATPS system

The ATPS power supply device is mounted in the immediate vicinity of the PBX. Wall mounting has to be vertical (connections at the top or at the bottom). The power supply unit is connected to the mainboard by its connecting cable (see Fig. 4.20). Connection to the 230 V network is via the standard PBX mains cord. When a PBX rack version is used, the ATPS can be mounted in the rack plug-in unit behind the PBX or in a separate plug-in unit altogether.

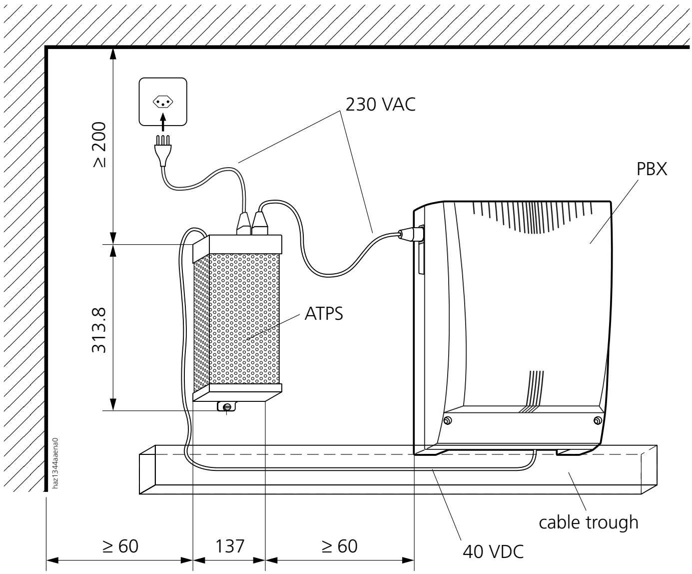


Fig. 4.19: Wall mounting of the ATPS system

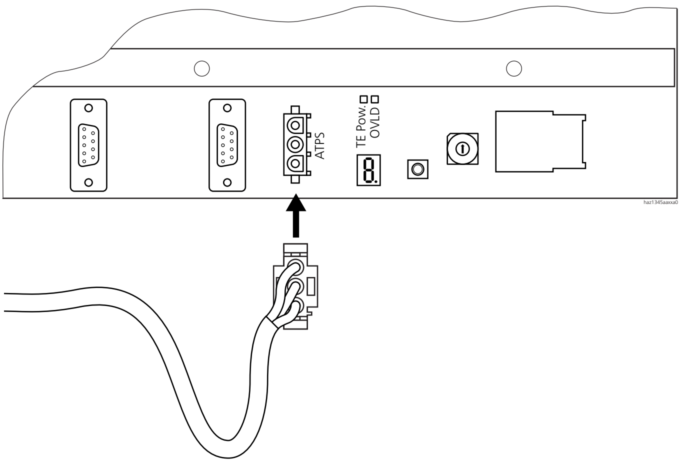


Fig. 4.20: Connector strip 2065 mainboard (MBL-2)

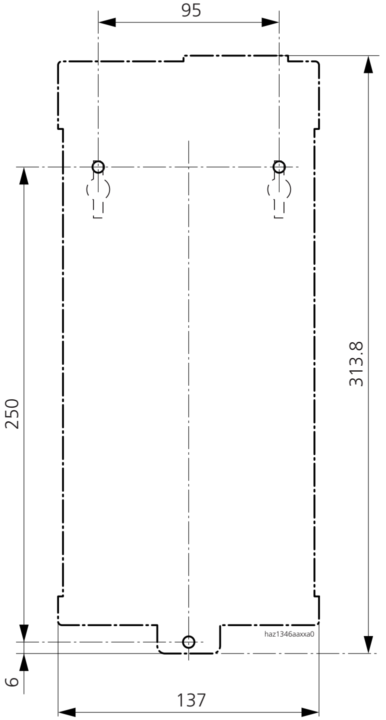


Fig. 4.21: Dimensional drawing for wall-mounting the ATPS

5 Equipping the PBX

5.1 System family

The Ascotel IntelliGate system family consists of three systems:

- Ascotel IntelliGate 2025
- Ascotel IntelliGate 2045
- Ascotel IntelliGate 2065

The systems differ in the number of expansion slots and are based on different basic systems:

- 2025 / 2045 basic system
- 2065 basic system

Tab. 4.5: Basic systems and number of expansion slots per system

System	Basic system	Expansion slots
Ascotel IntelliGate 2025	2025 / 2045	5
Ascotel IntelliGate 2045	2025 / 2045	5
Ascotel IntelliGate 2065	2065	14[1)]

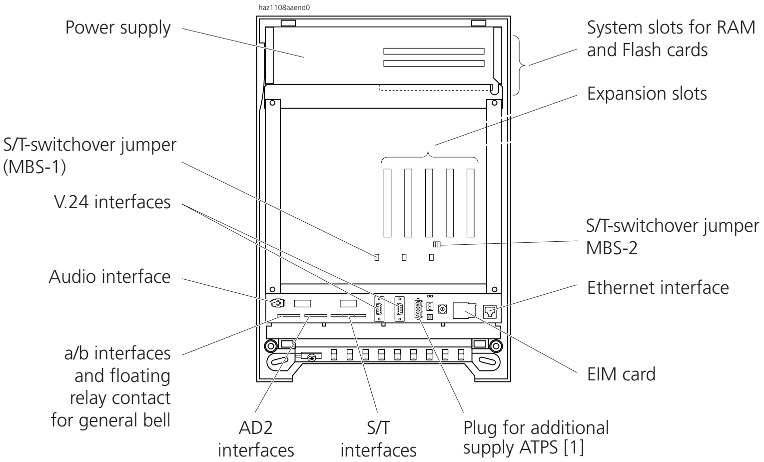
[1)] An additional slot (No. 8) is reserved for the processor card

See also:
"Component Mounting Rules and Examples", page 588"

5.1.1 Basic system 2025 / 2045

The 2025 / 2045 basic system provides the basis for the systems Ascotel IntelliGate 2025 and Ascotel IntelliGate 2045, and consists of the following components:

- 2025 / 2045 housing
- 2025 / 2045 mainboard
- Memory cards (SDRAM, Flash, EIM)



[1] As of mainboard hardware version MBS-2

Fig. 4.22: Mainboard 2025 / 2045

Tab. 4.6: Resources on the 2025 / 2045 mainboard

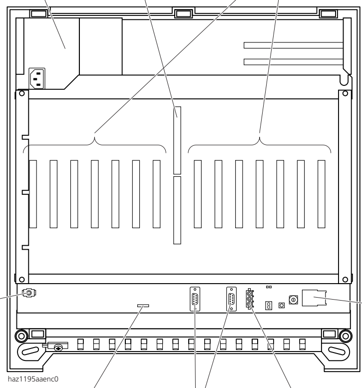
System	a/b	S/T (switch-able)	AD2	V.24	Ethernet interfaces	Audio Interfaces	Courtesy	General Bell	Internal music
2025 / 2045	3	3	4	2	1 (10 Base T)	1	approx. 1 minute	1	approx. 1 minute

For assignments of interfaces to cables and clamps, see Tab. 4.14.

5.1.2 2065 basic system

The 2065 basic system provides the basis for the Ascotel IntelliGate 2065 system. It consists of the following components:

- 2065 housing
- 2065 mainboard
- Processor card MPC-8260 incl. SDRAM memory card
- Memory cards (Flash, EIM)



[1] As of mainboard hardware version MBL-2

Fig. 4.23: Mainboard 2065

Tab. 4.7: Resources on the 2065 mainboard

System	V.24	Ethernet interface	Audio Interfaces	Courtesy	General Bell	Internal music
2065	2	—1)	1	approx. 2 x 1 minute	1	approx. 1 minute

1) On the Ascotel IntelliGate 2065 the processor card contains has 1 Ethernet interface (10 / 100Base T) MPC-8260

The 2065 mainboard of the 2065 basic system does not have any trunk connections or user-network interfaces.

5.2 Cards

Expansion cards and system cards are used in the system.

System cards

System cards ensure the system's operability. They are part of the basic system and are factory fitted.

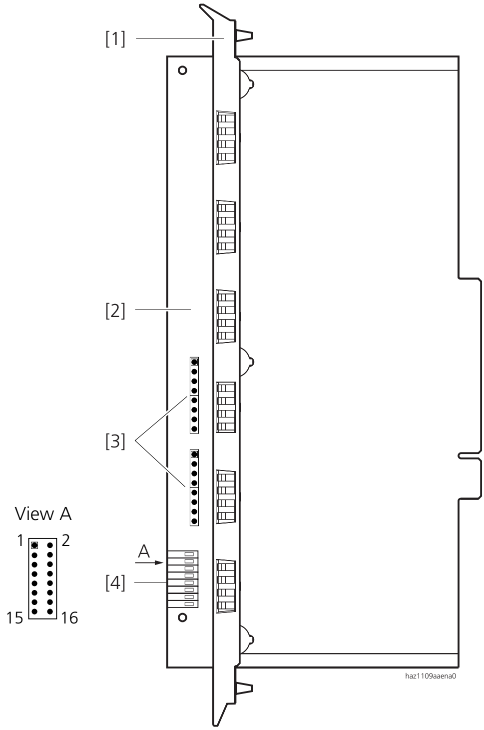
Note:
A PBX is supplied with the system cards already fitted. For information on system cards see "System cards", page 909.

Expansion cards

Expansion cards support special functions are provide interfaces (network interfaces, user-network interfaces and special interfaces). Cards are of the following type:

- Trunk cards
- Subscriber cards
- Special cards

The expansion cards are used to expand the system according to individual requirements. A number of rules have to be observed when fitting the cards (see "Component Mounting Rules and Examples", page 588).



[1] Shielding strip (front covering)
[2] Connector panel
[3] Direct connection; plug connectors for two-conductor pillar terminal blocks
[4] Indirect connection; connector for 16-wire connecting cable
▣ = Pin 1

Fig. 4.24: Expansion card

Maximum number of expansion cards

Tab. 4.8: Max. number of trunk cards of one particular type per PBX

Card type	Trunk connections	Max. number for Ascotel IntelliGate		
		2025	2045	2065
ISDN-04ST	S/T	1	2	8
ISDN-08ST	S/T	1 [1)]	1	8
ISDN-01PRA	PRA	1	2	8
ISDN-02PRA	PRA	—	1	8
TC-04AB	a/b	1	2	8

1) Due to the system limits not all the interfaces can be used (see Tab. 3.18)

Tab. 4.9: Max. number of subscriber cards of one particular type per PBX

Card type	User-network interfaces	Max. number for Ascotel IntelliGate		
		2025	2045	2065
SC-04AD2	AD2	2	4	14
SC-08AD2	AD2	1	2	14
SC-16AD2	AD2	1[1)]	1	14
SC-24AD2	AD2	1[1)]	1	13
SC-32AD2	AD2	1[1)]	1	10
SC-04AB	a/b	2	4	14
SC-08AB	a/b	1	2	10
SC-16AB	a/b	1[1)]	1	10
ISDN-04ST	S/T	1	2	8
ISDN-08ST	S/T	1[1)]	1	8

1) Due to the system limits not all the interfaces can be used (see Tab. 3.18)

Tab. 4.10: Max. number of special cards of one particular type per PBX

Card type	Special interfaces	Max. number for Ascotel IntelliGate		
		2025	2045	2065
OI-2DOOR	• 2 door intercom interfaces • 1 Interface for switching over switch group 1 • 3 freely connectable relays	1	1	1
DSP-01 / DSP-02	—	1	2	4
DSP-04	—	1	1	2
VM-02P / VM-04P[1)]	—	1	1	1
AIP-6400[1)]	Ethernet 10 / 100 Base T	1	2	4
AIP-6350[1)]	Ethernet 10 / 100 Base T	1	2	4

1) The relevant detailed documentation is available for each of these cards

Note:
The number of expansion cards per PBX is also determined by the number of expansion slots on the mainboard and the overall system limits.

5.2.1 Trunk cards

Trunk cards provide network interfaces.

Tab. 4.11: Trunk card interfaces

Trunk card	Network interfaces		
	a/b	S/T (switchable)	T2
ISDN-04ST	—	4	—
ISDN-08ST	—	8	—
ISDN-01PRA	—	—	1
ISDN-02PRA	—	—	2
TC-04AB	4	—	—

5.2.2 Subscriber cards

Subscriber cards provide user-network interfaces.

Tab. 4.12: Subscriber card interfaces

Subscriber card	User-network interfaces		
	a/b	AD2	S[1]/T (switchable)
ISDN-04ST	—	—	4
ISDN-08ST	—	—	8
SC-04AD2	—	4	—
SC-08AD2	—	8	—
SC-16AD2	—	16	—
SC-24AD2	—	24	—
SC-32AD2	—	32	—
SC-04AB	4	—	—
SC-08AB	8	—	—
SC-16AB	16	—	—

1) S interfaces can also be switched as S external for networking with another PINX.

5.2.3 Special cards

Special cards provide special interfaces and support special functions.

Tab. 4.13: Special cards

Special card	Function
OI-2DOOR	Contains 2 door intercom interfaces, 1 interface for switching over switch group 1, and 3 freely connectable relays
DSP-01	Supports Ascotel DECT cordless system; 12 DECT voice channels
DSP-02	Supports Ascotel DECT cordless system and full-duplex hands-free mode; 18 DECT voice channels, 6 hands-free channels
DSP-04	Supports Ascotel DECT cordless system and full-duplex hands-free mode; 36 DECT voice channels, 9 hands-free channels
VM-02P[1)]	For Voice Mail; 128 mailboxes, 2 ports[2)], 4 hours of recording capacity
VM-04P[1)]	For Voice Mail; 128 mailboxes, 4 ports[2)], 8 hours of recording capacity
AIP-6400[1)]	For VoIP / QSIG via IP; contains Ethernet interface (gateway)
AIP-6350[1)]	For linking up IP system terminals via IP; contains Ethernet interface (gateway)

1) The relevant detailed documentation is available for each of these cards

2) One a/b user-network interface is seized virtually per port

5.2.4 Fitting expansion cards

Hazard:
Expansion cards can be damaged by electrical voltage.
Always disconnect the power supply before fitting or removing any cards!
Take note of the warning plate on the PBX.

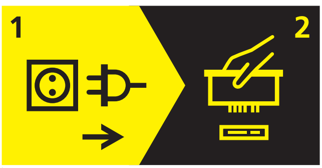


Fig. 4.25: Warning plate on the PBX

Warning:
Expansion cards can be damaged by electrostatic discharge (ESD)!
Always touch the earthed metal cage of PBX before touching the cards!

Expansion cards are electronic PCBs susceptible to ESD and are marked with the ESD pictogram (Fig. 4.26).

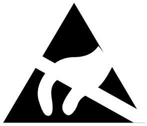


Fig. 4.26: ESD pictogram

Expansion cards can be fitted to any expansion slot.
On the Ascotel IntelliGate 2065, the middle slot (No. 8) is reserved for the MPC-8260 processor card (Fig. 4.23).

Fitting expansion cards

A Torx screwdriver, size Torx 10, is needed to secure the screening strip (front cover) to the metal cage.

1. Switch off the PBX: When using a UPS disconnect the battery first, then disconnect the PBX and any external Auxiliary Terminal Power Supply (ATPS) from the 230 V mains.
2. Remove the PBX cover.
3. Before handling an expansion card, touch the earthed metal cage of the PBX.
4. Take the expansion card out of the ESD protective sleeve.
5. Fit the expansion card to the correct slot. Make sure the card is lying in the lower guide slot of the metal cage and that the ends of the screening strip (front cover) are flush against the edge of the metal cage once the card has been inserted.

Note:
The maximum torque for screws used for fitting expansion cards and dummy covers is 0.9 Nm.

6. Screw the screening strip to the metal cage using the 2 screws supplied.

7. Fit dummy covers to any empty slots remaining.

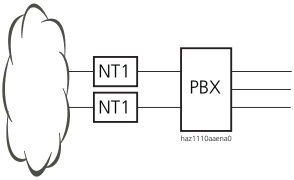
Note:
The dummy covers must be fitted to ensure EMC.

8. Fit the PBX cover.
9. Connect the PBX and any ATPS system to the 230 V mains, then connect the UPS battery where applicable.
10. Keep the ESD protective sleeve in a safe place.

6 Connecting the PBX

There are two possibilities for connection to the telephone network and the subscriber-side cabling:

- Direct connection via pillar terminal blocks (see Fig. 4.28)
- Indirect connection using prefabricated cables (see Fig. 4.31)



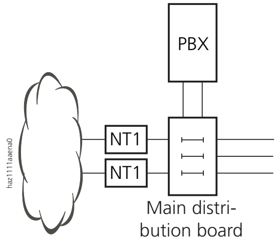


Fig. 4.27: Direct connection (left) and indirect connection (right)

Most expansion cards accommodate both connection variants. The table below provides an overview of existing possibilities:

Tab. 4.14: Connectors and terminals

Expansion card / mainboard	Number of 16-pin connectors for 16-wire cable	Number of quad two-conductor pillar terminal blocks
SC-04AD2	1 1)	2
SC-08AD2	1	4
SC-16AD2	2	8
SC-24AD2	3	--
SC-32AD2	4	--
SC-04AB	1 1)	2
SC-08AB	1	4
SC-16AB	2	8
ISDN-04ST	1	4
ISDN-08ST	2	8
TC-04AB	1 1)	2

Expansion card / mainboard	Number of 16-pin connectors for 16-wire cable	Number of quad two-conductor pillar terminal blocks
OI-2DOOR	1[2)]	10[3)]
2025 / 2045(MBS)	2[4)]	7
2065 (MBL)	—	1

1) 8 contacts blank

2) Only if a second door intercom is being used.

3) If all the functions of the first door intercom are being used.

4) 4 contacts blank

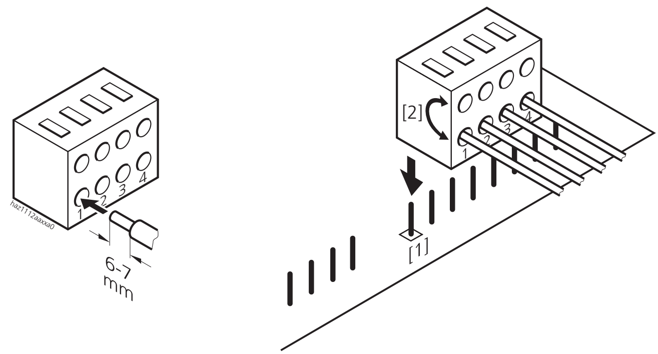
6.1 Direct connection

Direct connection is via quad two-conductor pillar terminal blocks. The pillar terminal blocks are supplied with the PBX; they can also be ordered in packs from the system supplier.

Suitable connecting wire:

- Solid copper conductor
- Conductor diameter 0.4 to 0.8 mm

Connecting the PBX



[1] ▣ = Pin 1
[2] The apertures above one another are electrically connected.

Fig. 4.28: Connecting and plugging in a quad two-conductor pillar terminal block

No tools are required to fit the connecting wires.

1. Strip the connecting wires over 6-7 mm.
2. Fit the connecting wires into the apertures on the pillar terminal blocks.
3. Fit the prefabricated pillar terminal block onto the plug connector on the mainboard 2025 / 2045 or expansion card.

Note:
The "Direct connection" variant with socket clamps is not suitable for shunting..

Disconnecting the connecting wires

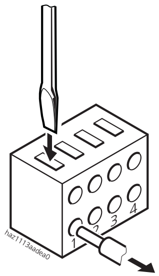


Fig. 4.29: Disconnecting the connecting wires

Use a size 1 screwdriver to loosen the connecting wires.

1. Insert the screwdriver into the rectangular aperture.
2. Press the screwdriver down until the wire clamp opens, and pull out the connecting wire.

Note:
Direct connection is also possible using stranded copper conductors (litz wires), conductor diameter 0.4 to 0.8 mm, and the corresponding commercial clamps (e.g. Messrs. Riacon; type 166; contact spacing 3.5 mm; 2-pin Order No. 311661 02, 4-pin Order No. 311661 04 or 8-pin Order No. 311661 08). Ensure sufficient strain relief!

6.2 Indirect connection

There are two possibilities for connecting the PBX indirectly to the telephone network and subscriber-side cabling:

- Connection via main distribution board
- Connection to a universal building cable installation (UBC)

6.2.1 Connection via main distribution board

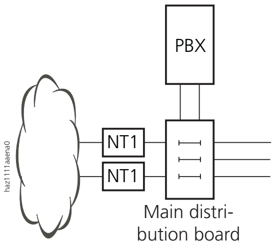


Fig. 4.30: Connecting via main distribution board (example)

16-pin connecting cable

The interface connectors on the 2025 / 2045 mainboard or the expansion cards in the PBX are connected to the main distribution board or the patch panels using a prefabricated 16-wire connecting cable.

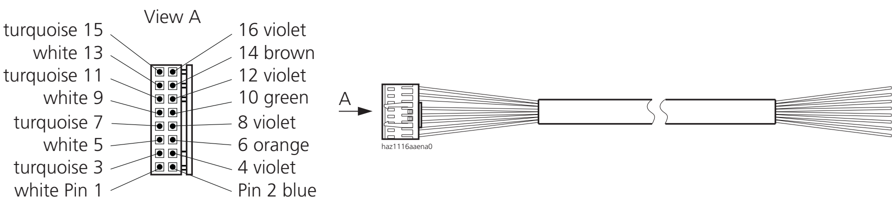


Fig. 4.31: 16-wire connecting cable with socket contact

The 16-wire connecting cable is supplied with the PBX or can be ordered from the system supplier in lengths of 3 m, 6 m and 20 m.

Making your own connecting cable

You can also make the 16-wire connecting cable yourself. To do so, use a 2 x 8-pin socket housing (e.g. AMP 0-926476-8), contacts (e.g. AMP 2-167301-4 or AMP 0-166500-2) and cable type with the following characteristics:

Tab. 4.15: Requirements for a 16-wire connecting cable

Core pairs x cores	4 x 4
Stranded	yes
Wire diameter, core	0.4…0.6 mm
Screening	Not necessary

6.2.2 Connection to a universal building cable installation (UBC)

For connecting the PBX indirectly to the universal building cabling, provide the following connection points in the main distribution frame:

Tab. 4.16: Connection points per interface

Interface	Connection points per interface	Connecting cable
S/T	4	16-wire connecting cable (see page 746)
a/b	2	16-wire connecting cable (see page 746)
AD2	2	16-wire connecting cable (see page 746)
T2	4	Patch cable, straight Cat. 5 (see page 760)
Ethernet interface	4	Patch cable, straight Cat. 5 (see page 803)
V.24	9	Modem cable (see page 795)
General Bell	2	16-wire connecting cable (see page 746)
Interface Switch group 1	2	Connecting cable (see page 813)
Freely connectable relay	6	Connecting cable (see page 815)
Door intercom interface 1	16	Connecting cable (see page 805)
Door intercom interface 2	16	16-wire connecting cable (see page 746)

Possibilities for connecting the PBX to a universal building cabling

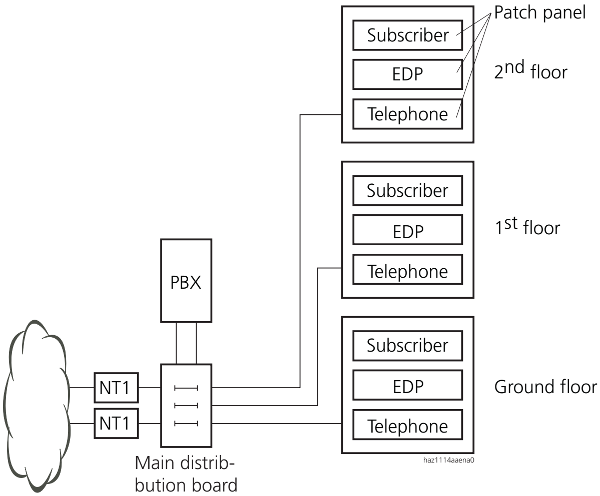


Fig. 4.32: Connecting to a UBC via main distribution board (example)

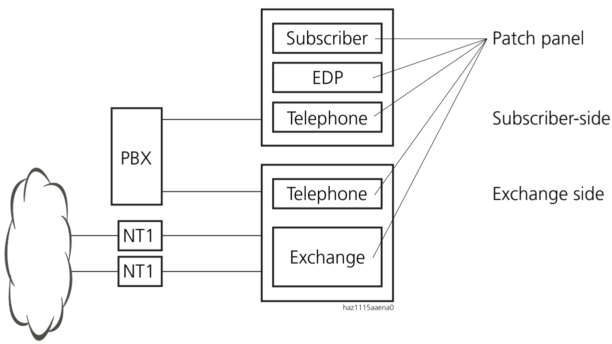


Fig. 4.33: Connecting to a UBC via wiring centre (example)

AD2 interface

Never install the Y bus by interrupting it at the storey distribution board. Only use commercial adapters to connect for instance two Office system terminals to the same terminal socket-outlet.

The max. length of an AD2 connection via secondary, tertiary and patch cables must not exceed 100 m in compliance with UGV guidelines. Both patch cables together can be a max. 10 m.

S-bus crossover

On the S-bus user-network interface the pairs c/d and e/f are wired in a different sequence on an RJ45 connector than on the UBC / ISDN network side.

Observe the following installation rules:

- Always terminate the bus extremity with 2 x 100 Ω (0.25 W, 5%)!
- Carry out the necessary crossover on the subscriber side of the PBX.
- If the subscriber cabling is integrated into the installation of a universal building cable installation (UBC), cross over the S bus as shown in Fig. 4.34.

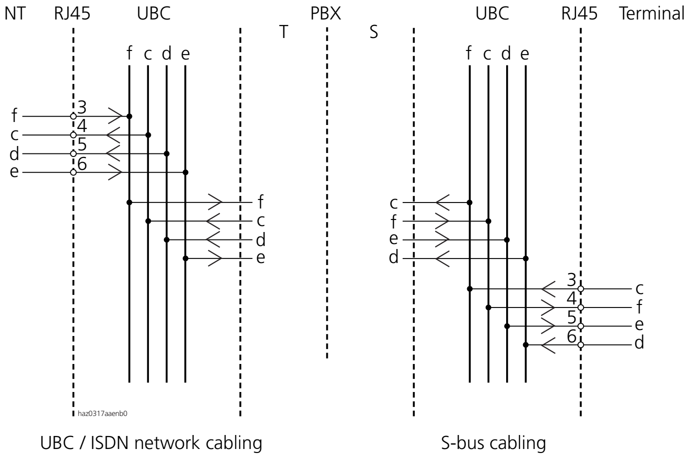


Fig. 4.34: S-bus crossover in the universal building cable installation

7 Cabling interfaces

7.1 Interfaces on the mainboard

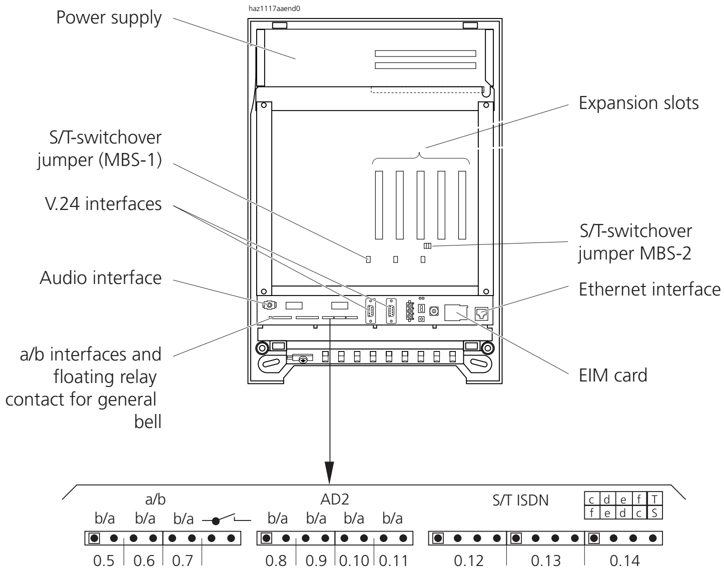


Fig. 4.35: Interfaces and port numbering on mainboard 2025 / 2045

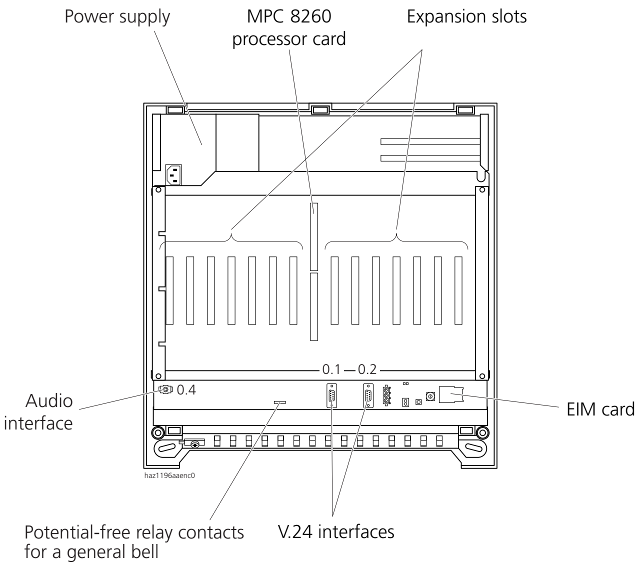


Fig. 4.36: Interfaces and port numbering on 2065 mainboard

Note:
Unlike the 2025 / 2045 system the 2065 system does not have any network interfaces or user-network interfaces on its mainboard. The Ethernet interface is located on the processor card.

7.2 Port addressing

A port address is always of the type x.y. x is the number of the expansion slot and y, the port number.

The slot numbering begins with 0 (= mainboard). Slot 8 is reserved for the processor card.

With S interface and AD2 interface addresses, the terminal selection digit (TSD) is displayed in addition to the slot and port numbers.

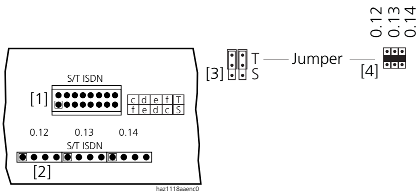
Tab. 4.17: Examples of interface addressing

Slot	Port address
Mainboard; interface 0.8	0.8
Card in expansion slot 1; interface x.4	1.4
Card in expansion slot 2; interface x.16	2.16
Terminal with TSD 6 on 2025 / 2045 mainboard on S interface x.12	0.12-6

7.3 Network Interfaces

7.3.1 Basic access T

7.3.1.1 Basic access T on the 2025 / 2045 mainboard



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection for T
[3] MBS-1: 1 jumper pair for each S/T interface; drawn position "T"
[4] MBS-2: 1 jumper per S/T interface, "T" if jumper fitted
For jumper arrangement see Fig. 4.35

▣ = Pin 1

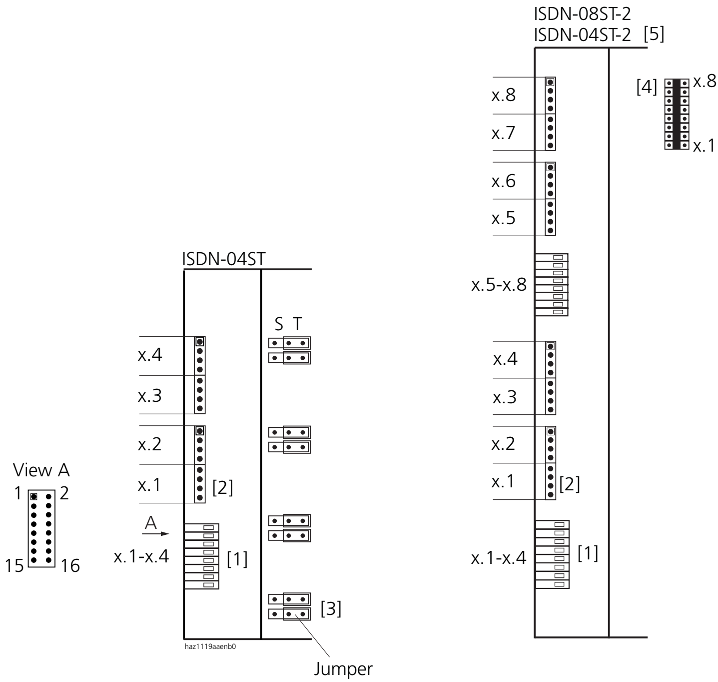
Fig. 4.37: Basic access T on the 2025 / 2045 mainboard

Tab. 4.18: Cabling for basic access T on the 2025 / 2045 mainboard

Network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
—	—	—	16	4	violet
	—	—	15		turquoise
	—	—	14		brown
	—	—	13		white
0.14	c	1	12	3	violet
	d	2	11		turquoise
	e	3	10		green
	f	4	9		white

Network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
0.13	c	1	8	2	violet
	d	2	7		turquoise
	e	3	6		orange
	f	4	5		white
0.12	c	1	4	1	violet
	d	2	3		turquoise
	e	3	2		blue
	f	4	1		white

7.3.1.2 Basic access T on the cards ISDN-0xST



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection for T
[3] ISDN-04ST: 1 jumper pair for each S/T interface; drawn position "T"
[4] ISDN-08ST-2 and ISDN 04ST-2: 1 jumper per S/T interface, "T" if jumper fitted
[5] On ISDN-04ST-2, only ports x.1 - x.4 are available
▣ = Pin 1

Fig. 4.38: Basic access T on the cards ISDN-04ST / ISDN-08ST

Tab. 4.19: Wiring of basic access T on the cards ISDN-04ST / ISDN-08ST

Interface on card		User-network interface		Direct connection	Indirect connection (16-wire cable)		
		No.	Connection	Pin	Pin	Stranded element	Core colour
ISDN-08ST		x.8	c	1	16	4	violet
			d	2	15		turquoise
			e	3	14		brown
			f	4	13		white
		x.7	c	5	12	3	violet
			d	6	11		turquoise
			e	7	10		green
			f	8	9		white
		x.6	c	1	8	2	violet
			d	2	7		turquoise
			e	3	6		orange
			f	4	5		white
		x.5	c	5	4	1	violet
			d	6	3		turquoise
			e	7	2		blue
			f	8	1		white
	ISDN-04ST	x.4	c	1	16	4	violet
			d	2	15		turquoise
			e	3	14		brown
			f	4	13		white
		x.3	c	5	12	3	violet
			d	6	11		turquoise
			e	7	10		green
			f	8	9		white
		x.2	c	1	8	2	violet
			d	2	7		turquoise
			e	3	6		orange
			f	4	5		white
		x.1	c	5	4	1	violet
			d	6	3		turquoise
			e	7	2		blue
			f	8	1		white

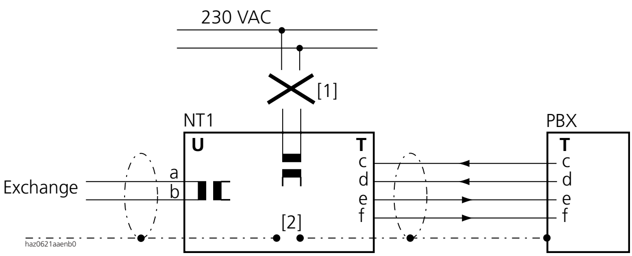
7.3.1.3 Cable Requirements

Tab. 4.20: Cable requirements for the basic access (NT1 to PBX)

Core pairs x cores	1 x 4 or 2 x 2
Stranded	yes
Wire diameter, core	0.4…0.6 mm
Screening	recommended
Characteristic impedance	< 125 Ω (100 kHz), < 115 Ω (1 MHz)
Wave attenuation	< 6 dB/km (100 kHz), < 26 dB/km (1 MHz)
Near / crosstalk attenuation	> 54 dB/100 m (1 kHz bis 1 MHz)

A prefabricated cable (NT-side with RJ45) is available as an option.

7.3.1.4 Basic access T, network-side



[1] Do not connect power supply NT1
[2] Do not fit the jumper

Fig. 4.39: Basic access on NT1

7.3.1.5 Basic access in the private leased-line network

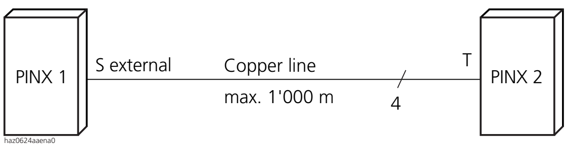


Fig. 4.40: Basic access S external, networked with copper line

Tab. 4.21: Cabling for basic access S external, networked with copper line

PINX 1 signal Basic access S external	Cable cores	PINX 2 signal Basic access T
c	←	c
d	←	d
e	→	e
f	→ haz1120aaxxb0	f

Bus configuration

S external is subject to the conditions that apply to user-network interface S (see "S user-network interfaces", page 776).

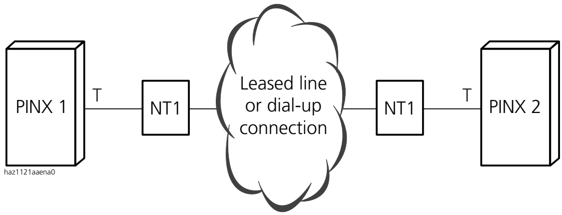


Fig. 4.41: Basic access T, networked with leased-line or dial-up connection

Tab. 4.22: Cabling for basic access T, networked with leased-line or dial-up connection

PINX 1 signal basicaccess T	Cable cores	NT1	Network	NT1	Cable cores	PINX 2 signal basicaccess T
c	→	c		c	←	c
d	→	d		d	←	d
e	←	e		e	→	e
f	← haz1122aaxxa0	f		f	→ haz1123aaxxa0	f

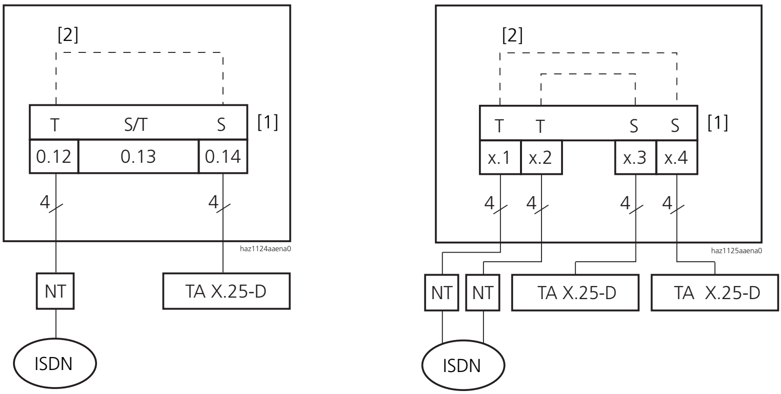
See also:
"Connections with basic accesses", page 665" and "Synchronization", page 669

7.3.1.6 X.25 connection in the D channel

The appropriate terminal adapters with S interface are required to be able to transmit X.25 data through the PBX to the public ISDN network via the D channel, for example from a PC via V.24.

X.25 D channel terminal adapters are connected to the S interfaces of the 2025 / 2045 mainboard or to the S interfaces of the ISDN-04ST / ISDN-08ST ISDN card.

A maximum of 3 (Ascotel IntelliGate 2025), 5 (Ascotel IntelliGate 2045) or 16 (Ascotel IntelliGate 2065) D channels are operated on each system.



[1] The interfaces must be configured accordingly (S/T)
[2] Connection in the PBX

Fig. 4.42: Configuration of the S/T interface for X.25 mode on the 2025 / 2045 mainboard (left) and on the ISDN-04ST ISDN card (right)

For information on how to operate X.25 on the D channel, see "Data services", page 353.

7.3.2 Primary Rate Access T2

7.3.2.1 Primary rate access T2 on the trunk cards ISDN-0xPRA

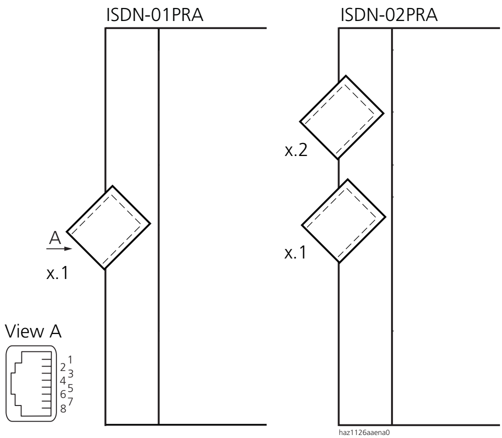
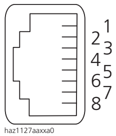
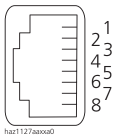


Fig. 4.43: Primary rate access T2 on the trunk cards ISDN-0xPRA

Tab. 4.23: Cabling primary rate access T2 on the trunk cards ISDN-0xPRA

Interface on card	Trunk connection No.	RJ45, 8-pin, screened	Pin	Connection
ISDN-02PRA	x.2		1	RxA2
			2	RxB2
			3	—
			4	TxA2
			5	TxB2
			6	—
			7	—
			8	—
ISDN-01PRA	x.1		1	RxA1
			2	RxB1
			3	—
			4	TxA1
			5	TxB1
			6	—
			7	—
			8	—

7.3.2.2 Cable requirements

The PRA connection to NT1 (Network Termination) is implemented using commercially available screened cables with 8-pin RJ45 connectors at both ends, e.g. S-FTP 4P, PVC, Cat. 5e.

Tab. 4.24: Cable requirements for primary rate access (NT1 to PBX; ISDN-0xPRA trunk card)

Core pairs x cores Stranded Wire diameter, core Screening	2 x 2 (for short distances also 1 x 4) yes 0.4...0.6 mm yes (Cat. 5)
Characteristic impedance Wave attenuation Near / crosstalk attenuation	< 125 Ω (100 kHz), < 115 Ω (1 MHz) < 6 dB/km (100 kHz), < 26 dB/km (1 MHz) > 54 dB/100 m (1 kHz bis 1 MHz)

7.3.2.3 Primary rate access T2, network side

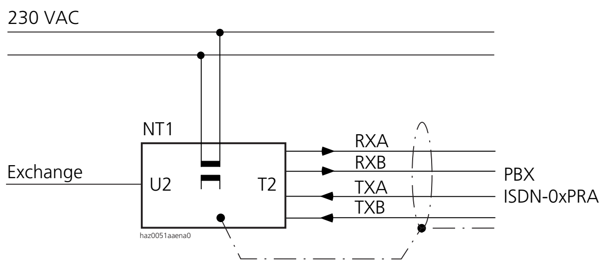


Fig. 4.44: Primary rate access T2 on NT1

Tab. 4.25: Cabling for primary rate access T2 on NT1

T2 signal NT1	Cable cores Straight patch cables	ISDN-0xPRA	
		T2 signal PBX	RJ45 Pin
RxA	→	RxA	1
RxB	→	RxB	2
		—	3
TxA	←	TxA	4
TxB	← haz1128aaxxa0	TxB	5
		—	6
		—	7
		—	8

7.3.2.4 Primary rate access in the private leased-line network

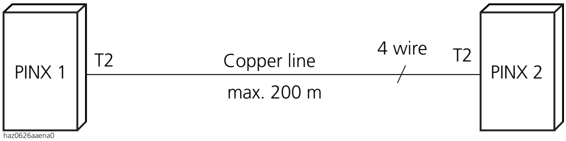
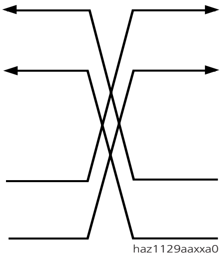


Fig. 4.45: Primary rate access, networked with copper line

Tab. 4.26: Cabling for primary rate access T2, networked with copper line

ISDN-0xPRA		Cable cores Crossed patch cables	ISDN-0xPRA	
RJ45 Pin	T2 signal PINX 1		T2 signal PINX 2	RJ45 Pin
1	RxA		RxA	1
2	RxB		RxB	2
3	—		—	3
4	TxA		TxA	4
5	TxB		TxB	5
6	—		—	6
7	—		—	7
8	—		—	8

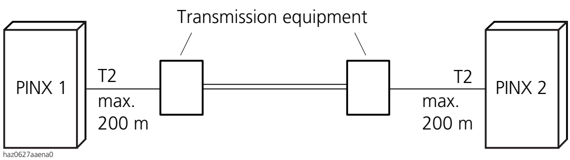


Fig. 4.46: Primary rate access, networked with transmission equipment TE

Tab. 4.27: Cabling for primary rate access T2, networked with transmission equipment

ISDN-0xPRA		Cable cores Straight patch cables	Signal TE		Signal TE	Cable cores Straight patch cables	ISDN-0xPRA	
RJ45 Pin	T2 signal PINX 1						T2 signal PINX 2	RJ45 Pin
1	RxA	←	RxA		RxA	→	RxA	1
2	RxB	←	RxB		RxB	→	RxB	2
3	—						—	3
4	TxA	→	TxA		TxA	←	TxA	4
5	TxB	→ haz1130aaxxa0	TxB		TxB	← haz1131aaxxa0	TxB	5
6	—						—	6
7	—						—	7
8	—						—	8

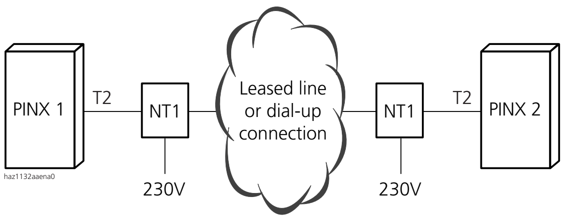


Fig. 4.47: Primary rate access T2, networked with leased-line or dial-up connection

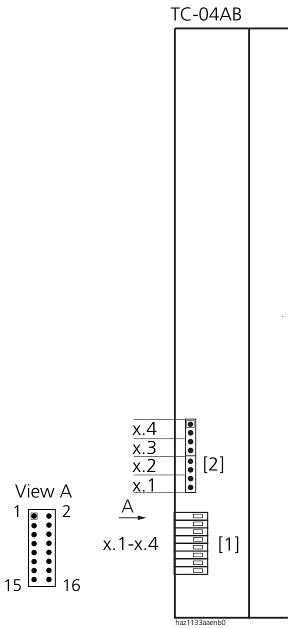
Tab. 4.28: Cabling for primary rate access T2, networked with leased-line or dial-up connection

ISDN-0xPRA		Cable cores Straight patch cables	T2 signal NT1	Network	T2 signal NT1	Cable cores Straight patch cables	ISDN-0xPRA	
RJ45 Pin	T2 signal PINX 1						T2 signal PINX 2	RJ45 Pin
1	RxA	←	RxA		RxA	→	RxA	1
2	RxB	←	RxB		RxB	→	RxB	2
3	—						—	3
4	TxA	→	TxA		TxA	←	TxA	4
5	TxB	→ haz1130aaxxa0	TxB		TxB	← haz1131aaxxa0	TxB	5
6	—						—	6
7	—						—	7
8	—						—	8

See also:
"Connections with primary rate accesses", page 664 and "Synchronization", page 669

7.3.3 Network interfaces a/b

7.3.3.1 Network interfaces a/b on the card TC-04AB



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection
▣ = Pin 1

Fig. 4.48: Analogue network interfaces on the card TC-04AB

Tab. 4.29: Cabling for analogue network interface on trunk card TC-04AB

Network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
—	—	—	16	4	violet
			15		turquoise
			14		brown
			13		white
—	—	—	12	3	violet
			11		turquoise
			10		green
			9		white

Network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
x.4	b	1	8	2	violet
	a	2	7		turquoise
x.3	b	3	6		orange
	a	4	5		white
x.2	b	5	4	1	violet
	a	6	3		turquoise
x.1	b	7	2		blue
	a	8	1		white

7.3.3.2 Network interface a/b network-side

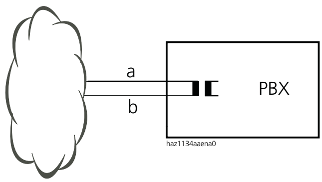
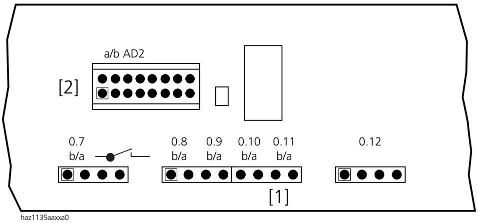


Fig. 4.49: Analogue network interface a/b

7.4 User-Network Interfaces

7.4.1 AD2 user-network interfaces

7.4.1.1 User-network interfaces AD2 on the 2025 / 2045 mainboard



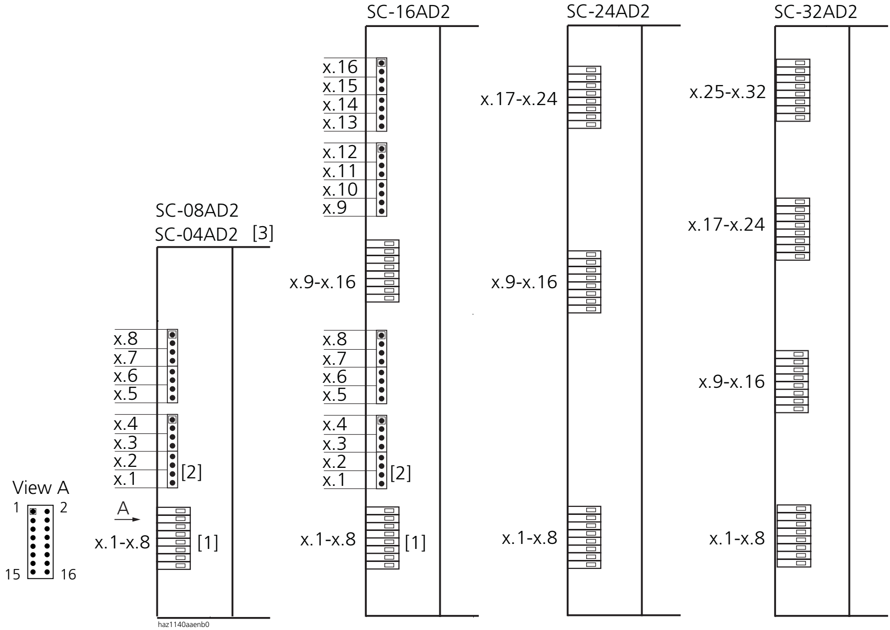
[1] Direct connection AD2
[2] Indirect connection; connector for prefabricated 16-wire connecting cable
▣ = Pin 1

Fig. 4.50: User-network interfaces AD2 on the 2025 / 2045 mainboard

Tab. 4.30: Cabling for user-network interfaces AD2 on the 2025 / 2045 mainboard

User-network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
0.11	b	7	16	4	violet
	a	8	15		turquoise
0.10	b	5	14		brown
	a	6	13		white
0.9	b	3	12	3	violet
	a	4	11		turquoise
0.8	b	1	10		green
	a	2	9		white
—	—	—	8	2	violet
			7		turquoise
			6		orange
			5		white
—	—	—	4	1	violet
			3		turquoise
			2		blue
			1		white

7.4.1.2 User-network interfaces AD2 on the cards SC-xxAD2



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection
[3] On SC-04AD2, only ports x.1 - x.4 are available

▣ = Pin 1

Fig. 4.51: User-network interfaces AD2 on the cards SC-xxAD2

Tab. 4.31: Wiring of user-network interfaces AD2 on the cards SC-04AD2 / SC-08AD2 / SC-16AD2

Interface on card	User-network interface		Direct connection	Indirect connection (16-wire cable)		
	No.	Connection	Pin	Pin	Stranded element	Core colour
SC-16AD2	x.16	b	1	16	4	violet
		a	2	15		turquoise
	x.15	b	3	14		brown
		a	4	13		white
	x.14	b	5	12	3	violet
		a	6	11		turquoise
	x.13	b	7	10		green
		a	8	9		white
	x.12	b	1	8	2	violet
		a	2	7		turquoise
	x.11	b	3	6		orange
		a	4	5		white
	x.10	b	5	4	1	violet
		a	6	3		turquoise
	x.9	b	7	2		blue
		a	8	1		white
SC-08AD2	x.8	b	1	16	4	violet
		a	2	15		turquoise
	x.7	b	3	14		brown
		a	4	13		white
	x.6	b	5	12	3	violet
		a	6	11		turquoise
	x.5	b	7	10		green
		a	8	9		white
SC-04AD2	x.4	b	1	8	2	violet
		a	2	7		turquoise
	x.3	b	3	6		orange
		a	4	5		white
	x.2	b	5	4	1	violet
		a	6	3		turquoise
	x.1	b	7	2		blue
		a	8	1		white

With subscriber cards SC-24AD2 and SC-32AD2 the connection can only be made using prefabricated 16-wire connecting cables.

Tab. 4.32: Wiring of user-network interfaces AD2 on the cards SC-24AD2 and SC-32AD2

Interface on card	User-network interface		Direct connection	Indirect connection (16-wire cable)		
	No.	Connection	Pin	Pin	Stranded element	Core colour
SC-32AD2	x.25 – x.32	b32	—	16	4	violet
		a32		15		turquoise
		b31		14		brown
		a31		13		white
		b30	—	12	3	violet
		a30		11		turquoise
		b29		10		green
		a29		9		white
		b28	—	8	2	violet
		a28		7		turquoise
		b27		6		orange
		a27		5		white
		b26	—	4	1	violet
		a26		3		turquoise
		b25		2		blue
		a25		1		white
SC-24AD2 / 32AD2	x.17 – x.24	b24	—	16	4	violet
		a24		15		turquoise
		b23		14		brown
		a23		13		white
		b22	—	12	3	violet
		a22		11		turquoise
		b21		10		green
		a21		9		white
		b20	—	8	2	violet
		a20		7		turquoise
		b19		6		orange
		a19		5		white
		b18	—	4	1	violet
		a18		3		turquoise
		b17		2		blue
		a17		1		white

Interface on card	User-network interface No.	Connection	Direct connection Pin	Indirect connection (16-wire cable) Pin	Stranded element	Core colour
SC-24AD2 / 32AD2	x.9 – x.16	b16	—	16	4	violet
		a16		15		turquoise
		b15		14		brown
		a15		13		white
		b14	—	12	3	violet
		a14		11		turquoise
		b13		10		green
		a13		9		white
		b12	—	8	2	violet
		a12		7		turquoise
		b11		6		orange
		a11		5		white
		b10	—	4	1	violet
		a10		3		turquoise
		b9		2		blue
		a9		1		white
	x.1 – x.8	b8	—	16	4	violet
		a8		15		turquoise
		b7		14		brown
		a7		13		white
		b6	—	12	3	violet
		a6		11		turquoise
		b5		10		green
		a5		9		white
		b4	—	8	2	violet
		a4		7		turquoise
		b3		6		orange
		a3		5		white
		b2	—	4	1	violet
		a2		3		turquoise
		b1		2		blue
		a1		1		white

7.4.1.3 AD2 bus configuration

Depending on the line length 1 or 2 terminals[1] can be connected on each AD2 interface. The following requirements apply with regard to the bus length to ensure that the maximum permissible signal delay is not exceeded:

Tab. 4.33: AD2 bus length and number of terminals

Number of terminals	Total length AD2 bus	Distance between the 1st and 2nd connection point (excl. connection cord)
1	A: max 1200 m	-
2	B: max. 700 m or. 1200 m[1]	C: max. 10 m

[1] For 1200 m both terminals must belong to the new family of system terminals (Office 10 / 25 / 35 / 45)

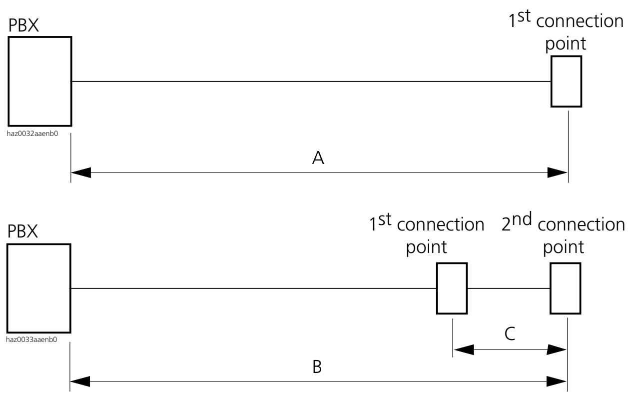


Fig. 4.52: AD2 bus

Note:
The total length of the cables from the PBX to the Office terminal must not be less than 10 m.

[1] In this connection the Pocket Adapter (PA) and the DECT radio unit are also considered as terminals.

Restrictions

The maximum length of an AD2 bus is further restricted by:

- the maximum power requirements of the connected terminals and their supplementary equipment
- the line resistance (depending on the line length and wire diameter)

Tab. 4.34: Maximal power requirements of the terminals on the AD2 bus

System terminal	Supplementary equipment	Max. power input [mW]
Office 45pro[1)]	3 expansion keypads	< 10
Office 45	—	1050
Office 45	1 expansion keypad	1260
Office 45	2 expansion keypads	1470
Office 45	3 expansion keypads	1680
Office 45	Alpha keyboard	1080
Office 45	alpha keyboard + 1 expansion keypad	1290
Office 45	alpha keyboard +2 expansion keypads	1500
Office 35	—	570
Office 35	1 expansion keypad	720
Office 35	2 expansion keypads	870
Office 35	3 expansion keypads	1020
Office 35	Alpha keyboard	600
Office 35	Alpha keyboard + 1 Expansion keypad	750
Office 35	alpha keyboard +2 expansion keypads	900
Office 25	—	900
Office 40	—	830
Office 40	Additional keypad	1120
Office 30	—	830
Office 30	Additional keypad	1040
Office 20	—	370
Office 10	—	900
Pocket Adapter (PA)	—	400
DECT radio unitSB-4	without power supply unit	1900
DECT radio unitSB-4	with power supply unit	600
DECT radio unitSB-8	without power supply unit	1300
DECT radio unitSB-8	with power supply unit	< 10

1) Power requirements from the PBX, when the terminal is provided power via a plug-in power supply unit (required for illuminating the display)

The table below shows the power available in relation to the line length and the line cross-section. The table can then be used to determine the number and type of terminals that can be connected to the AD2 bus under the given conditions.

Tab. 4.35: Power available for terminals on the AD2 bus

Loop resistance R	20	40	60	80	100	120	140	160	180	200	220	240	260	280	300
Power available $P_{max.}$ (mW)	2620	2492	2364	2236	2108	1980	1852	1724	1596	1459	1343	1245	1160	1086	1021
Length (m) with:															
0.4 mm wire ø	71	142	213	284	356	427	498	569	640	711	782	853	924	996	1000
0.5 mm wire ø	111	222	333	444	556	667	778	889	1000	1111	1200	1200	1200	1200	1200
0.6 mm wire ø	160	320	480	640	800	960	1120	1200	1200	1200	1200	1200	1200	1200	1200

Permissible range (max. 700 m) for 2 terminals on the AD2-Bus if one or both terminals are of the type Office 20, Office 30 or Office 40. For the system terminals Office 10, Office 25, Office 35 and Office 45 this restriction does not apply.

With the SB-8 radio unit only half the output is to be used, provided it is connected to two AD2 interfaces.

Automatic detection of critical power supply situations

When an Office terminal (or a second such terminal) is connected to the AD2 bus, the maximum power input is automatically determined; all the terminals (incl. EKP, AKB) connected to the interface are taken into account. The maximum power available is also determined based on the calculated line length (assumption: wire ø = 0.5 mm). If the calculated power available is below the maximum possible power input of the connected terminals, the message "Power supply critical xy m" is generated on the terminal connected last (accuracy approx. 150 m).

Rating examples

Example 1

Office 45 with 1 expansion keypad and alpha keyboard
Maximal power requirements as per Tab. 4.34: 1290 mW

Tab. 4.35 indicates:

- Maximum line length for a wire diameter of 0.4 mm: 782 m
- Maximum line length for a wire diameter of 0,6 mm: 1200 m

Example 2

2 Office 30 without expansion keypad
Power requirements as per Tab. 4.34: 2 x 830 mW = 1660 mW.

Tab. 4.35 indicates:

- Maximum line length for a wire diameter of 0.4 mm: 569 m
- Maximum line length for a wire diameter of 0,6 mm: 1200 m
  As the two terminals belong to the old system family, the greyed area of Tab. 4.35 is applied here, i. e. the permissible line length is only 640 m instead of 1200 m.

Example 3

Evaluation of an existing line installation with two partial lines:

Tab. 4.36: Existing line consisting of 2 partial lines

Partial line	Diameter [mm]	Resistance R [Ω]	Line length from Tab. 4.35 [m]
1	0.4	60	213
2	0.6	140	1120
1 + 2			1333

Conclusion: The existing line cannot be used as an AD2 bus as the permissible line length in accordance with Tab. 4.35 is exceeded.

Installation rules

- If an Ascotel DECT radio unit is used, do not connect any other system terminal to the same AD2 bus.
- Do not use any terminating resistors at the bus extremity.
- Avoid using different cable cross-sections on the same bus
- Use the supplied cables for connecting the terminals

Cable Requirements

Tab. 4.37: Requirements for an AD2 bus cable

Core pairs x cores	1 x 4 or 2 x 1
Stranded	yes[1)]
Wire diameter, core	0.4…0.6 mm
Screening	recommended
Characteristic impedance	< 130 Ω (1 MHz)

1) Note: max. 25 m can be crossed unstranded.
(CH: Applies also to cable type G51)

Connection sockets

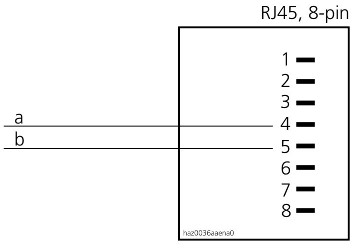


Fig. 4.53: RJ45 connection, single socket

7.4.1.4 Terminals

The following system terminals can be operated on the AD2 bus:

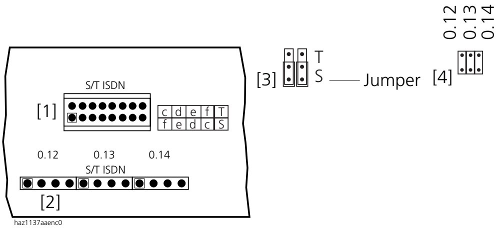
- System terminals of the Office family
- Office Pocket Adapter
- Ascotel DECT radio unit

The terminals on an AD2 bus are addressed via a single-digit terminal selection digit (TSD).

Example:
The address of a terminal with TSD 2 on AD2 interface 0.9 on the 2025 / 2045 mainboard is 0.9-2.

7.4.2 S user-network interfaces

7.4.2.1 User-network interfaces S on the 2025 / 2045 mainboard



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection for S
[3] MBS-1: 1 jumper pair for each S/T interface; drawn position "S"
[4] MBS-2: 1 jumper per S/T interface; "S" if jumper not fitted
For jumper arrangement see Fig. 4.35

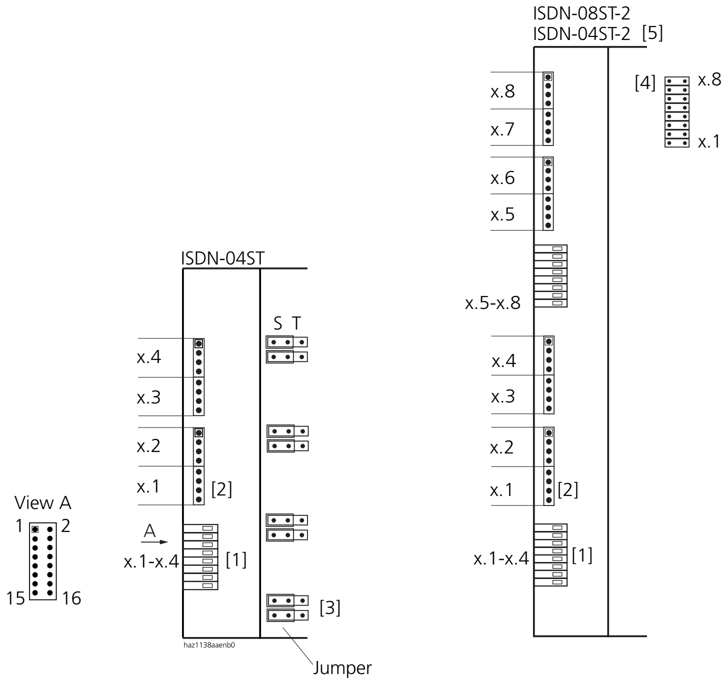
▣ = Pin 1

Fig. 4.54: User-network interfaces S on the 2025 / 2045 mainboard

Tab. 4.38: Cabling for user-network interfaces S on the 2025 / 2045 mainboard

User-network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
—	—	—	16	4	violet
			15		turquoise
			14		brown
			13		white
0.14	f	1	12	3	violet
	e	2	11		turquoise
	d	3	10		green
	c	4	9		white
0.13	f	1	8	2	violet
	e	2	7		turquoise
	d	3	6		orange
	c	4	5		white
0.12	f	1	4	1	violet
	e	2	3		turquoise
	d	3	2		blue
	c	4	1		white

7.4.2.2 User-network interfaces S on the cards ISDN-0xST



[1]	Indirect connection; connector for prefabricated 16-wire connecting cable
[2]	Direct connection for S
[3]	ISDN-04ST: 1 jumper pair for each S/T interface; drawn position "S"
[4]	ISDN-08ST-2 and ISDN 04ST-2: 1 jumper per S/T interface; "S" if jumper not fitted
[5]	On ISDN-04ST-2, only ports x.1 - x.4 are available
▣ =	Pin 1

Fig. 4.55: User-network interfaces S on the cards ISDN-0xST

Tab. 4.39: Wiring of the user-network interfaces S on the cards ISDN-0xST

Interface on card		User-network interface		Direct connection	Indirect connection (16-wire cable)		
		No.	Connection	Pin	Pin	Stranded element	Core colour
ISDN-08ST		x.8	c	1	16	4	violet
			d	2	15		turquoise
			e	3	14		brown
			f	4	13		white
		x.7	c	5	12	3	violet
			d	6	11		turquoise
			e	7	10		green
			f	8	9		white
		x.6	c	1	8	2	violet
			d	2	7		turquoise
			e	3	6		orange
			f	4	5		white
		x.5	c	5	4	1	violet
			d	6	3		turquoise
			e	7	2		blue
			f	8	1		white
	ISDN-04ST	x.4	c	1	16	4	violet
			d	2	15		turquoise
			e	3	14		brown
			f	4	13		white
		x.3	c	5	12	3	violet
			d	6	11		turquoise
			e	7	10		green
			f	8	9		white
		x.2	c	1	8	2	violet
			d	2	7		turquoise
			e	3	6		orange
			f	4	5		white
		x.1	c	5	4	1	violet
			d	6	3		turquoise
			e	7	2		blue
			f	8	1		white

7.4.2.3 S bus configuration

The S bus is a four-wire, serial ISDN bus based on the DSS1 protocol (ETSI standard). It starts in each case at an S interface of the PBX. Four bus configurations are possible, depending on the line length and the number of terminals:

Tab. 4.40: S-bus configurations depending on line length and the number of terminals

S bus	Short	Short, V-shaped	Long	Point-to-point
Length (max.) PBX ↔ Terminal Terminal 1 ↔ Terminal 4	 150 m —	 2 x 150 m —	 500 m 20 m	 1’000 m —
Number of terminals (max.)	8	8	4	1

Note:
The maximum number of terminals per S bus depends on the power requirements of the terminals (see "Restrictions", page 772).

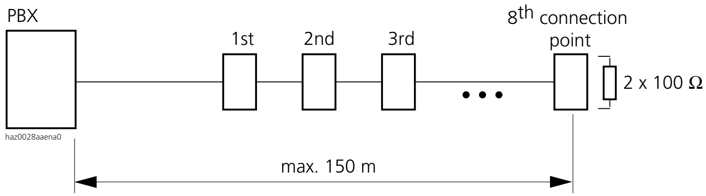


Fig. 4.56: S bus, short

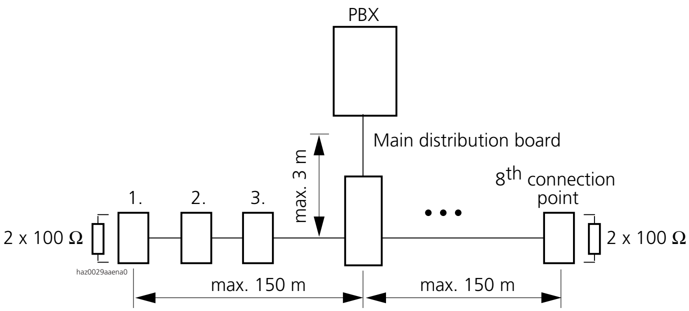


Fig. 4.57: S bus, short, V-shaped

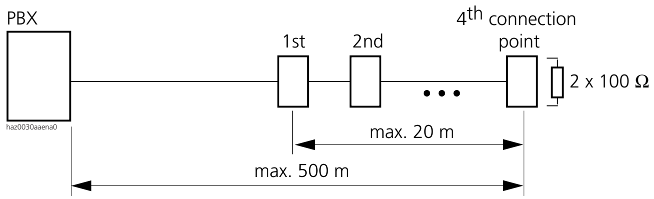


Fig. 4.58: S bus, long

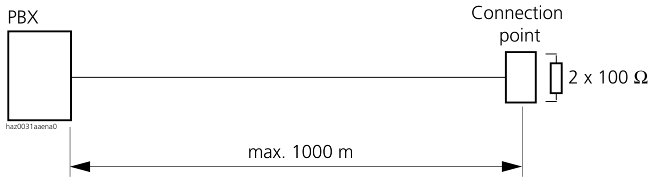


Fig. 4.59: S bus, point-to-point

Greater distances (up to 8 km) can be achieved using the S bus extension PT 10 (see "S Bus Extension PT 10", page 138).

Restrictions

The maximum number of terminals per S bus is further restricted by the power requirements of the terminals and their supplementary equipment:

Tab. 4.41: Power balance on the S bus

	Power available [W]
S bus short	5[1)]
S bus, long	3.5[1)]

1) These values are based on a wire diameter of 0.5 mm.

The number of terminals is the sum of the power requirements of the individual terminals and the power available on the S bus.

Installation rules

- Always terminate the bus extremity with 2 x 100 Ω (0.25 W, 5%)!
- If the subscriber cabling is integrated into the installation of a universal building cable installation (UBC), cross-over the S bus see "Connection to a universal building cable installation (UBC)", page 747.

Cable Requirements

Tab. 4.42: Requirements for an S bus cable

Core pairs x cores	1 x 4 or 2 x 2
Stranded	yes
Wire diameter, core	0.4…0.6 mm
Screening	recommended
Ohmic resistance	< 98 Ω/km (conductor), < 196 Ω/km (loop)
Characteristic impedance	< 125 Ω (100 kHz), < 115 Ω (1 MHz)
Wave attenuation	< 6 dB/km (100 kHz), < 26 dB/km (1 MHz)
Near / crosstalk attenuation	> 54 dB/100 m (1 kHz to 1 MHz)

Connection sockets

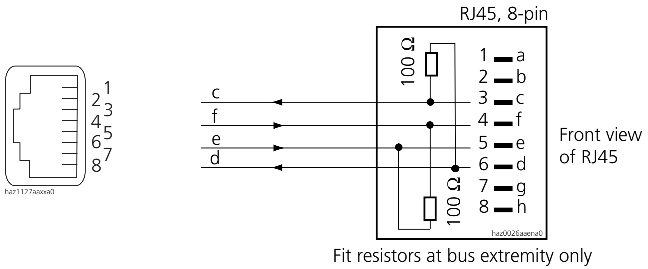


Fig. 4.60: RJ45 connection, single socket

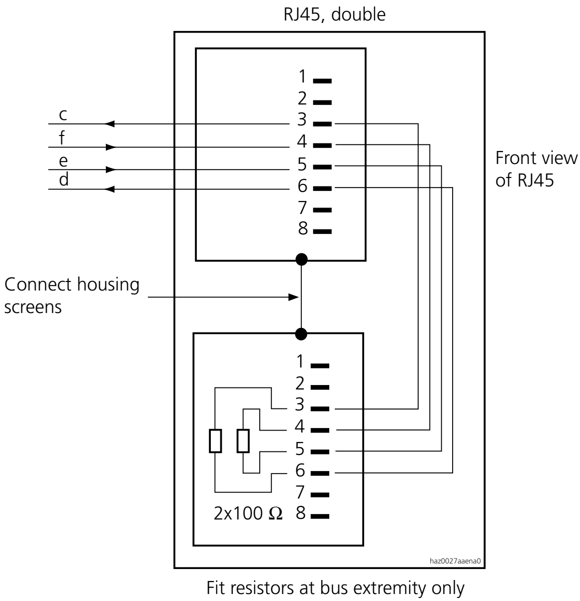


Fig. 4.61: RJ45 connection, double socket

7.4.2.4 Terminals

ISDN terminals

The ETSI protocol must be set in the interface configuration.

Up to 8 terminals of different types can be connected to one S bus.

- Standard ISDN terminals
- ISDN Terminal Adapter
- PC cards
- Group 4 fax machines, etc.

Two call connections are possible simultaneously for each S bus.

The terminals are addressed via a single-digit terminal selection digit (TSD). On an ISDN terminal the TSD must be entered under the "MSN" menu item.

Example:
The address of a terminal with TSD 6 on S interface 0.12 on the 2025 / 2045 mainboard is: Trunk connection 0.12-6.

In some cases ISDN terminals do not log on to the PBX with a TSD. The system automatically enters the first ISDN terminal to log on without an TSD as TSD 1 in the numbering plan. Other ISDN terminals without a TSD cannot be operated on the same S bus since TSD 1 has already been allocated. The other terminals on the bus must log on with TSD 2...8 on the PBX.

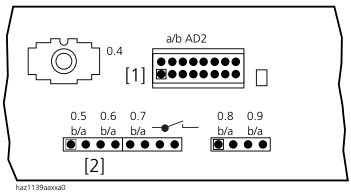
PC Operator Console Office 1550

The V2 protocol must be set in the interface configuration.

2 PC Operator Consoles Office 1550 can be connected to an S bus in V2 mode.

7.4.3 User-network interfaces a/b

7.4.3.1 User-network interfaces a/b on the 2025 / 2045 mainboard



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection for user-network interface a/b

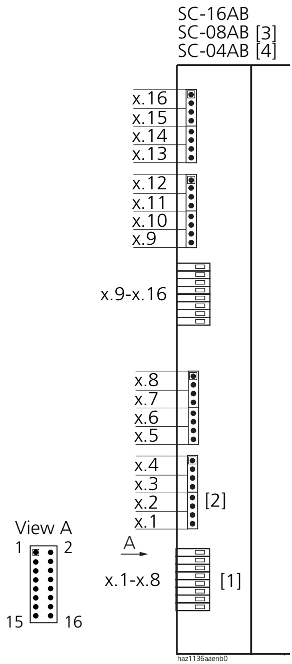
▣ = Pin 1

Fig. 4.62: Analogue user-network interfaces a/b on the 2025 / 2045 mainboard

Tab. 4.43: Cabling for analogue user-network interfaces a/b on the 2025 / 2045mainboard

User-network interface		Direct connection	Indirect connection (16-wire cable)		
No.	Connection	Pin	Pin	Stranded element	Core colour
—	—	—	16	4	violet
			15		turquoise
—	—		14		brown
			13		white
—	—	—	12	3	violet
			11		turquoise
—	—		10		green
			9		white
—	General bell	7	8	2	violet
		8	7		turquoise
0.7	b	5	6		orange
	a	6	5		white
0.6	b	3	4	1	violet
	a	4	3		turquoise
0.5	b	1	2		blue
	a	2	1		white

7.4.3.2 User-network interfaces a/b on the cards SC-xxAB



[1] Indirect connection; connector for prefabricated 16-wire connecting cable
[2] Direct connection
[3] SC-08AB only contains port x.1 - x.8
[4] SC-04AB only contains port x.1 - x.4

▣ = Pin 1

Fig. 4.63: Analogue user-network interfaces a/b on the cards SC-xxAB

Tab. 4.44: Cabling for analogue user-network interfaces a/b on subscriber card SC-xxAB

<table>
<tr><th colspan="3" rowspan="2">Interface on card</th><th colspan="2">User-network interface</th><th>Direct connection</th><th colspan="3">Indirect connection (16-wire cable)</th></tr>
<tr><th>No.</th><th>Connection</th><th>Pin</th><th>Pin</th><th>Stranded element</th><th>Core colour</th></tr>
<tr><td rowspan="32">SC-16AB</td><td colspan="2" rowspan="16"></td><td rowspan="2">x.16</td><td>b</td><td>1</td><td>16</td><td rowspan="8">4</td><td>violet</td></tr>
<tr><td>a</td><td>2</td><td>15</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.15</td><td>b</td><td>3</td><td>14</td><td>brown</td></tr>
<tr><td>a</td><td>4</td><td>13</td><td>white</td></tr>
<tr><td rowspan="2">x.14</td><td>b</td><td>5</td><td>12</td><td rowspan="8">3</td><td>violet</td></tr>
<tr><td>a</td><td>6</td><td>11</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.13</td><td>b</td><td>7</td><td>10</td><td>green</td></tr>
<tr><td>a</td><td>8</td><td>9</td><td>white</td></tr>
<tr><td rowspan="2">x.12</td><td>b</td><td>1</td><td>8</td><td rowspan="8">2</td><td>violet</td></tr>
<tr><td>a</td><td>2</td><td>7</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.11</td><td>b</td><td>3</td><td>6</td><td>orange</td></tr>
<tr><td>a</td><td>4</td><td>5</td><td>white</td></tr>
<tr><td rowspan="2">x.10</td><td>b</td><td>5</td><td>4</td><td rowspan="8">1</td><td>violet</td></tr>
<tr><td>a</td><td>6</td><td>3</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.9</td><td>b</td><td>7</td><td>2</td><td>blue</td></tr>
<tr><td>a</td><td>8</td><td>1</td><td>white</td></tr>
<tr><td rowspan="16">SC-08AB</td><td rowspan="8"></td><td rowspan="2">x.8</td><td>b</td><td>1</td><td>16</td><td rowspan="8">4</td><td>violet</td></tr>
<tr><td>a</td><td>2</td><td>15</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.7</td><td>b</td><td>3</td><td>14</td><td>brown</td></tr>
<tr><td>a</td><td>4</td><td>13</td><td>white</td></tr>
<tr><td rowspan="2">x.6</td><td>b</td><td>5</td><td>12</td><td rowspan="8">3</td><td>violet</td></tr>
<tr><td>a</td><td>6</td><td>11</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.5</td><td>b</td><td>7</td><td>10</td><td>green</td></tr>
<tr><td>a</td><td>8</td><td>9</td><td>white</td></tr>
<tr><td rowspan="8">SC-04AB</td><td rowspan="2">x.4</td><td>b</td><td>1</td><td>8</td><td rowspan="8">2</td><td>violet</td></tr>
<tr><td>a</td><td>2</td><td>7</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.3</td><td>b</td><td>3</td><td>6</td><td>orange</td></tr>
<tr><td>a</td><td>4</td><td>5</td><td>white</td></tr>
<tr><td rowspan="2">x.2</td><td>b</td><td>5</td><td>4</td><td rowspan="8">1</td><td>violet</td></tr>
<tr><td>a</td><td>6</td><td>3</td><td>turquoise</td></tr>
<tr><td rowspan="2">x.1</td><td>b</td><td>7</td><td>2</td><td>blue</td></tr>
<tr><td>a</td><td>8</td><td>1</td><td>white</td></tr>
</table>

Cable Requirements

Tab. 4.45: Requirements for a/b cables

Core pairs x cores	1 x 2
Stranded	only for lengths > 200 m
Wire diameter, core	0.4 … 0.8 mm
Resistance a/b	max. 2 x 250 Ω
Line length with ø 0.6	max. 4 km
Screening	not required

Connection sockets

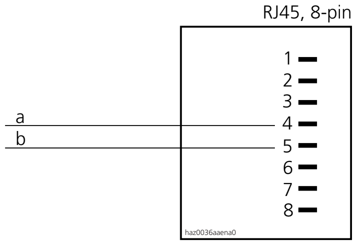


Fig. 4.64: RJ45 connection, single socket

7.4.3.3 Terminals

The following analogue terminals can be connected to the PBX:

- DTMF telephones
- Pulse dialling phones (earth key not supported)
- Cordless phones
- Group 3 fax machines
- Answering machines
- Modem

7.5 Special Interfaces

7.5.1 V.24 Interface

V.24 standard

V.24 is a serial interface used for connecting a printer or PC. In accordance with the V.24 recommendation, a cable length of 15 m limits the transmission speed to 38 400 bit/s. The system's maximum transmission speed is 115200 bit/s. The cable length must not exceed 2 m. Transmission errors can occur if these limits are exceeded. A current loop converter (TTY) has to be used if the cable length is insufficient. Cable lengths of up to approx. 1 km are possible with this interface.

Two types of equipment can be connected to the V.24 interface:

- DCE: Data Communication Equipment
- DTE: Data Terminal Equipment

The pin assignment differs according to the type of equipment used.

Signals are specified as follows:

Tab. 4.46: V.24 signals

Signal	Pin		Signal direction		CCITT standard V.24	Name / function
	D-Sub-9	D-Sub-25	DTE	DCE		
TXD	3	2	out	in	103	Transmitted Data
RXD	2	3	in	out	104	Received Data
RTS	7	4	out	in	105	Request To Send
CTS	8	5	in	out	106	Clear To Send
DTR	4	20	out	in	108	Data Terminal Ready
DSR	6	6	in	out	107	Data Set Ready
DCD	1	8	in	out	109	Data Carrier Detect
SGND	5	7	—	—	102	Signal Ground

A straight connecting cable (modem cable; straight) must be used when connecting DTE-type equipment with DCE-type equipment (e.g. PC → modem, modem → PC).

A null modem cable must be used when connecting equipment of similar type DTE – DTE (e.g. PC→PC). The null modem cable is crossed.

The DCD signal should only be used if a DCE is connected with a DTE.
DCD is activated by the DCE once the connection to the DTE is established.

Flow control

Depending on the set mode, the data flow is controlled differently by the equipment involved. The most common modes are Xon / Xoff or RTS /CTS.

Xon / Xoff mode

This mode is also known as a software handshake.
The data flow is controlled by the data receiving equipment. The data receiving equipment sends an SW signal over the data line as soon as its input memory is full, and an SW signal as soon as it is ready to receive again.

Xon / Xoff requires only a three-wire connection for signals RxD, TxD and SGND.

Drawback: The data transmitting equipment cannot tell whether or not a receiving equipment is connected.

RTS / CTS mode

This mode is also known as a hardware handshake.
The two equipment involved use the RTS and CTS signals to indicate that they are ready to send and to receive, each signal being transmitted over a separate signalling line.

Advantage: Data exchange cannot take place if there is no equipment ready to receive. For example no more data will be sent to the printer if the printer is switched off or out of paper.

Note:
To exclude any transmission errors, always operate the printer on the PBX in RTS / CTS mode.

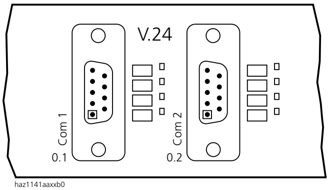
Initialization values

Tab. 4.47: V.24 initialization values

Interface	Transmission rate (bit/s)	Format	Parity	Stop bit	Flow control
COM1	9600	8 bit	none	1	HW
COM2	115 200	8 bit	none	1	HW

7.5.1.1 V.24 interface on the mainboard

Connection



= Pin 1

Fig. 4.65: V.24 interfaces on the 2025 / 2045 and 2065 mainboards

Settings

The port address of V.24 interfaces on the mainboard is 0.1 and 0.2.

See also:
For details of the settings see AIMS, Configuration Manager, Call Logging and Event Messages

Signalling the Ready Status

The PBX uses the DTR signal (Data Terminal Ready) to indicate that the V.24 interface status is ready.

7.5.1.2 V.24 interface on the Pocket Adapter (PA)

The PA provides a V.24 interface for connecting an AD2 interface with a PC. The PA's V.24 cable is connected to a serial interface on the PC[1].
Two terminals can be connected to an AD2 interface. An Office terminal can be connected via the PA. The PBX differentiates the two terminals on the basis of the position of DIP switch S2 on the PA (see Fig. 4.67) and the setting of the terminal selection digit (TSD) on the Office terminal. Both terminals must have a different TSD.

[1] PA Version ≥ V2.4

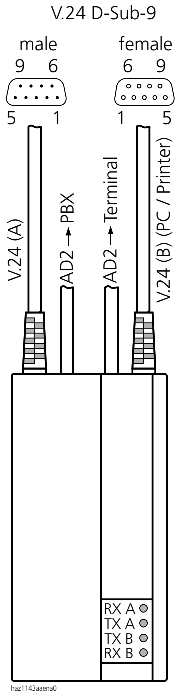


Fig. 4.66: LED displays and connections on the Pocket Adapter

Settings

The same communication parameters must be set on all the connected equipment. On the PA this is done using DIP switches S1... S8 (see Fig. 4.67).

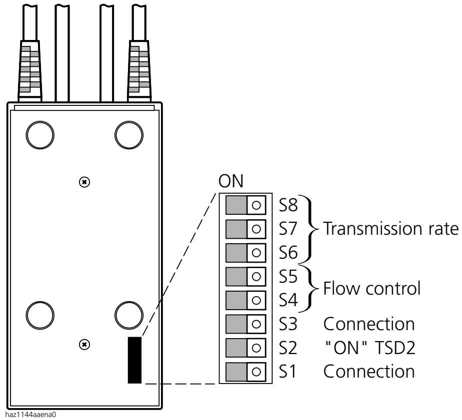


Fig. 4.67: DIP switches

Connection

DIP switches S1 and S3 are used for setting the two pieces of equipment that are to communicate with each other via the PA.

Tab. 4.48: Setting the connection

Switch S1	Switch S3	Connections
ON	ON	Reserve
ON	OFF	V.24 (B) ↔ AD2 terminal
OFF	ON	V.24 (A) ↔ V.24 (B)
OFF	OFF	V.24 (A) ↔ AD2 terminal

Example:

V.24 (B) ´ AD2 terminal
Connects the Office system terminal with PC or Mac

Transmission rate

Tab. 4.49: Setting the transmission rate

Switch S8	Switch S7	Switch S6	Transmission rate
ON	ON	ON	Test mode
ON	ON	OFF	19200
ON	OFF	ON	9600
ON	OFF	OFF	4800
OFF	ON	ON	2400
OFF	ON	OFF	1200
OFF	OFF	ON	600
OFF	OFF	OFF	reserved

Flow control

Tab. 4.50: Setting the flow control

Switch S5	Switch S4	Flow control
ON	ON	none
ON	OFF	Xon / Xoff
OFF	ON	Hardware with RTS / CTS[1)]
OFF	OFF	Xon / Xoff and RTS / CTS

[1)] Factory setting. Essential for online operation with AIMS

Terminal selection digit (TSD)

Usually the Pocket Adapter is addressed as Terminal 2.

Tab. 4.51: Setting the address

Switch S2	Address
ON	2. Terminal
OFF	1. Terminal

Notes:
– The PA cannot be set with AIMS.
– With AIMS a PA cannot be configured on to an interface.
– Connected PAs are indicated in AIMS after a download (but cannot be modified).

Significance of the LEDs

The 4 LEDs indicate the equipment status and the current direction of the data transmission.

Tab. 4.52: LED display

	LED on	LED blinking	LED flashes once	LED flashes twice
RX A	DTR B = on	Data from PBX to V.24 (A)	Xoff to V.24 (A)	—
TX A	DSR A = on	Data from the equipment on V.24 (A) to PBX or PC	Xoff from the PBX	Startup
TX B	DSR B = on	Data from PC to PBX	Xoff from the PBX	Startup
RX B	RTS B = on	Data from PBX to PC	Xoff from PC	—

In the test mode all the LEDs blink very quickly and in unison.

Signalling the Ready Status

The DTR signal (Data Terminal Ready) must be used to indicate to the PA that the data terminal equipment connected is ready. If this signal is not available no data will be sent.

7.5.1.3 Connector types

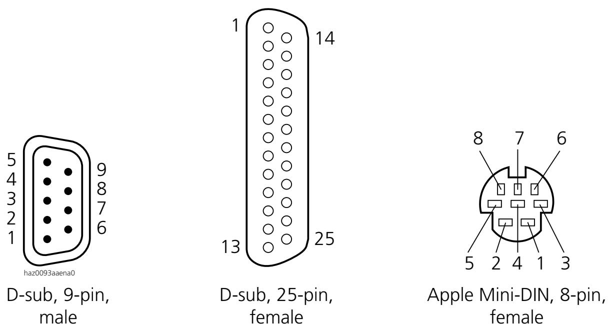


Fig. 4.68: V.24 connector types, front view of connector and socket

Tab. 4.53 indicates the type of plug-in connections used on the equipment (female = socket and male = connector) and the DTE or DCE equipment type.

Tab. 4.53: Type of connectors on the interfaces to the Ascotel IntelliGate

Device	Type	Connection type (connector on the equipment)
Mainboard	DTE	D-sub, 9-pin (male)
Pocket Adapter (PA)	DCE	D-sub, 9-pin (female)
Serial printer	DTE	D-sub, 25-pin (female)
PC	DTE	D-sub, 9-pin or D-sub, 25-pin (male)
Mac	DTE	Mini-DIN, 8-pin (female)
PC Operator Console Terminal Adapter	DCE	D-sub, 9-pin (female)

7.5.1.4 Cable types

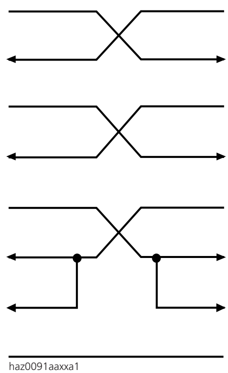
Tab. 4.54 to Tab. 4.59 provide an overview of the V.24 connections that occur in Ascotel IntelliGate systems and the connector and cable types used.

Tab. 4.54: V.24 cable types on the PBX

	Mainboard DTE	PA Office DCE	Serial printer DTE	PC / Mac DTE	TA PC Operator Console DCE
Mainboard DTE	—	—	PR	X/[1]	—
PA Office DCE	—	—	=	=/= Mac	—
Serial printer DTE	PR	=	—	—	—
PC / Mac DTE	X/[1]	=/= Mac	—	—	= [2]
TA PC Operator Console DCE	—	—	—	= [2]	—

X crossed cable (null modem)
= straight cable (modem cable)
= Mac straight cable for Mac computers
[1] special cable 1
[2] combination with Mac not possible
PR printer cable (serial)

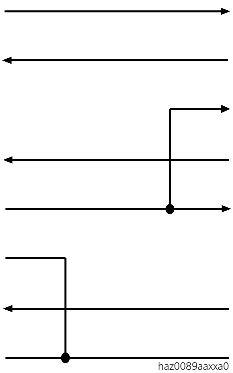
Tab. 4.55: Crossed cable (null modem)

Signal	DTE		Cable cores	DTE		Signal
	D-Sub-9 female	D-Sub-25 female		D-Sub-25 female	D-Sub-9 female	
TXD	3	2		2	3	TXD
RXD	2	3		3	2	RXD
RTS	7	4		4	7	RTS
CTS	8	5		5	8	CTS
DTR	4	20		20	4	DTR
DSR	6	6		6	6	DSR
DCD	1	8		8	1	DCD
SGND	5	7		7	5	SGND
Application with Ascotel IntelliGate	Mainboard Mainboard	— —		— PC	PC —	

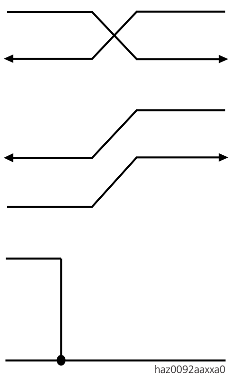
Tab. 4.56: Straight cables (modem cables)

Signal	DTE D-Sub-9 female	DTE D-Sub-25 female	Cable cores	DCE D-Sub-25 male	DCE D-Sub-9 male	Signal
TXD	3	2	→	2	3	TXD
RXD	2	3	←	3	2	RXD
RTS	7	4	→	4, 11	7	RTS
CTS	8	5	←	5	8	CTS
DTR	4	20	→	20	4	DTR
DSR	6	6	←	6	6	DSR
DCD	1	8	←	8	1	DCD
SGND	5	7	— haz0088aaxxa0	7	5	SGND
Application with Ascotel IntelliGate	—	PC		—	PA	
	PC	—		—	PA	
	Printer	—		—	PA	
	—	PC		—	TA PC Operator Console	
	PC	—		—	TA PC Operator Console	

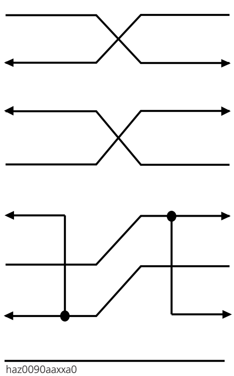
Tab. 4.57: Straight cables (modem cables) for Mac applications

Apple standard Signal	DTE Mini-DIN, 8-pin male	Cable cores	DCE D-Sub-25 male	DCE D-Sub-9 male	Signal
TXD-	3		2	3	TXD
RXD-	5		3	2	RXD
			4, 11	7	RTS
Handshake on	2		5	8	CTS
Handshake off	1		20	4	DTR
RXD+	8		6	6	DSR
General input	7		8	1	DCD
GND	4		7	5	SGND
Application with Ascotel IntelliGate	Mac Mac		Mainboard —	— PA	

Tab. 4.58: Special cable 1: Mac – Mainboard

Apple standard Signal	DTE Mini-DIN, 8-pin male	Cable cores	DTE D-Sub-9 female	Signal
TXD-	3		3	TXD
RXD-	5		2	RXD
			7	RTS
Handshake on	2		8	CTS
Handshake off	1		4	DTR
RXD+	8		6	DSR
General input	7		1	DCD
GND	4		5	SGND
Application with Ascotel IntelliGate	Mac		Mainboard	

Tab. 4.59: Printer cables: Printer – Mainboard

Signal	DTE D-Sub25 (fe)male	Cable cores	DTE D-Sub9 female	Signal
TXD	2		3	TXD
RXD	3		2	RXD
DCD	8		1	DCD
RTS	4		7	RTS
CTS	5		8	CTS
DTR	20		4	DTR
DSR	6		6	DSR
SGND	7		5	SGND
Application with Ascotel IntelliGate	Printer		Mainboard	

7.5.2 Ethernet Interface

A 10 Base T Ethernet interface is integrated on the 2025 / 2045 mainboard. A 10 / 100 Base T Ethernet interface is integrated on the MPC-8260 processor card on the 2065 system. These Ethernet interfaces have system functions exclusively.

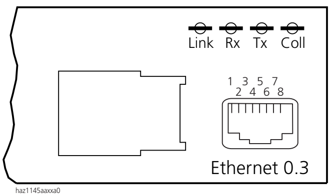


Fig. 4.69: Ethernet interface on the 2025 / 2045 mainboard

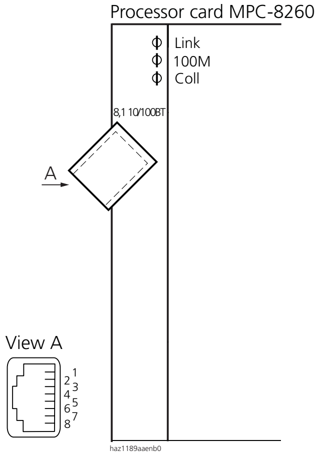
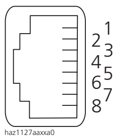


Fig. 4.70: Ethernet interface on the MPC-8260 processor card

Connection

Tab. 4.60: Connection of the Ethernet interface

RJ45 socket, 8-pin, screened	Pin	Designation
	1	Tx+
	2	Tx–
	3	Rx+
	4	—
	5	—
	6	Rx–
	7	—
	8	—

Settings

The port address for the Ethernet interface is 0.3. The IP address is determined using the Configuration Manager.

The initialization values are as follows:

- IP Address: 192.168.104.13
- IP Netmask: 255.255.255.0
- IP Gateway: 192.168.104.1

The settings for the Ethernet interface are stored in the EIM licence chip and therefore are not lost during initialization. The IP address can be viewed using the AIMS Configuration Manager.

Cable types

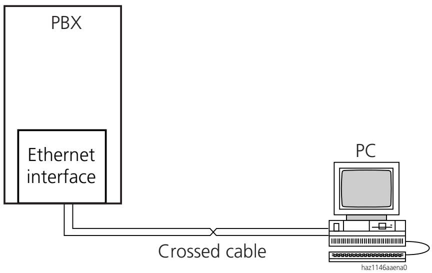
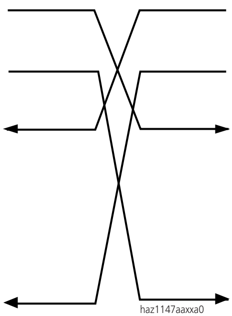


Fig. 4.71: Cat. 5 cable, crossed, for direct connection Ethernet interface – PC

Tab. 4.61: Wiring for direct connection Ethernet interface – PC

PBX		Cable cores	PC	
RJ45 Pin	Signal	Crossed cables	Signal	RJ45 Pin
1	Tx+		Tx+	1
2	Tx–		Tx–	2
3	Rx+		Rx+	3
4	—		—	4
5	—		—	5
6	Rx–		Rx–	6
7	—		—	7
8	—		—	8

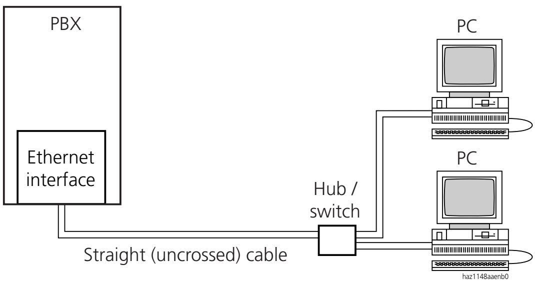


Fig. 4.72: Cat. 5 cable, straight, for Ethernet interface – hub (– PC) connection

Tab. 4.62: Wiring for Ethernet interface – hub connection

PBX		Cable cores Straight cables	Hub	
RJ45 Pin	Signal		Signal	RJ45 Pin
1	Tx+	→	Rx+	1
2	Tx–	→	Rx–	2
3	Rx+	←	Tx+	3
4	—		—	4
5	—		—	5
6	Rx–	← haz1149aaxxa0	Tx–	6
7	—		—	7
8	—		—	8

Cable requirements

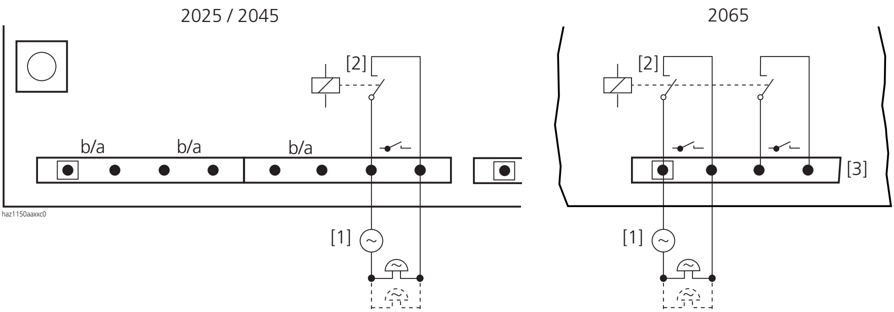
Use commercial Cat. 5 cable, or choose a cable type with the following characteristics:

Tab. 4.63: Requirements for an Ethernet cable

Core pairs x cores	2 x 2 (for short distances also 1 x 4)
Stranded	yes
Wire diameter, core	0.4...0.6 mm
Screening	yes (Cat. 5)

7.5.3 General bell

Connection via potential-free relay contact



[1] External ringing voltage source
[2] Maximum contact loading 60 VDC, 0.5 A / 50 VAC, 0.5 A
[3] On mainboard2065 the general bell can be connected either on Pin 1/2 or Pin 3/4

▣ = Pin 1

Fig. 4.73: Connecting the general bell to the central relay with floating contacts on the 2025 / 2045 mainboard (left) and 2065 mainboard (right)

The relay on the mainboard can only be used if there is an external ringing voltage source. The maximum number of general bells connected in parallel depends on the power of the ringing voltage source.

It is possible to use commercial auxiliary bells designed for connection in parallel to analogue terminals as a general bell.

Connection via Analogue User-network interface a/b

One analogue user-network interface per system can be configured in such a way that it is also used for connecting a general bell. This eliminates the need for an external ringing voltage source. However the impedance of the connected general bell (or total impedance in the case of several devices connected in parallel) must not fall below 1 kΩ (see also "General Bell on an Analogue user-network interface a/b", page 522).

Tip:
Both connection types can also be combined if several general bells are used.

7.5.4 Audio interface

The Audio interface can be used to

- play music or an announcement on to connections with a caller on hold ("Music on Hold" function). Any playback equipment (tape recorder,CD player,etc.) with a line output can be used as the music source
- play (copy) music or announcements into the memories of the "Courtesy" function.

The customer is responsible for all copyright matters relating to any music playback.

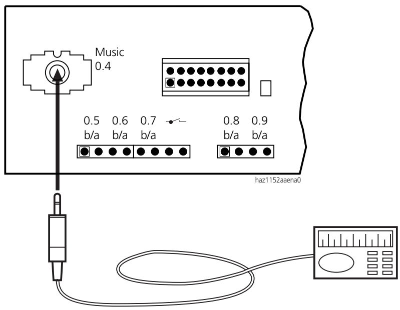


Fig. 4.74: Audio interface (in accordance 2025 / 2045 mainboard, 2065 mainboard)

Tab. 4.64: Technical data

Input impedance	approx. 15 kΩ
Input level	0.1…5 V (configurable in 8 levels via AIMS)
Output resistance, music source	< 1 kΩ
Installation cable	NF cable screened (required for low levels)
Mainboard socket	3.5 mm stereo jack

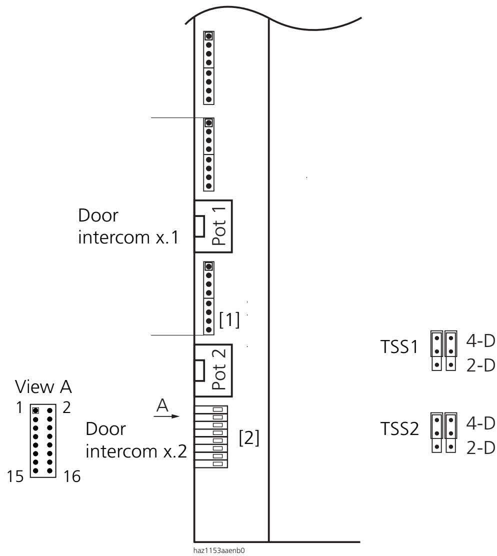
7.5.5 Door intercom system

The two electrically identical signal circuits for connecting a door intercom or announcement system are located on the OI-2DOOR special card. The interfaces for the two signal circuits are metallically isolated from each other.

Connect the first door intercom interface with pillar terminal blocks. Connect the second door intercom interface using a prefabricated 16-wire connecting cable (see "16-pin connecting cable", page 746).

The connection variants are as follows:

- Connection without amplifier
- Connection two-wire
- Connection four-wire
- Connecting a loudspeaker system



[1] Direct connection
[2] Indirect connection; connector for prefabricated 16-wire connecting cable
▣ = Pin 1

Fig. 4.75: Interfaces, changeover switches and potentiometers for the door intercom

Tab. 4.65: Signals and electrical values for the door intercom interface

Function	Signal	Value
Power supply input	AC, AC'	8...14 VAC
Ringing signal input	SON, SON'	5...30 VAC / DC
Connection for two-wire door intercom	Ta, SGND	600 Ω
Microphone signal input	MIC	130 mV (for full output)
Microphone remote supply output	MIC+	approx. 8 VDC
Loudspeaker connection output	LS, SGND	max. 3 W on 4 Ω
Floating contact, "Switch on / off power supply to door intercom"	TS, TS'	max. 24 VDC, 30 VAC, 1 A
Floating contact, "door opener"	TO, TO'	max. 24 VDC, 30 VAC, 1 A

Tab. 4.66: Wiring for door intercom interface 1 (door intercom x.1)

Signal Interface x.1	Pin
—	1
—	2
SON'1	3
SON1	4
AC'1	1
AC1	2
TO'1	3
TO1	4
TS'1	1
TS1	2
Tb1 (= SGND1)	3
Ta1	4
SGND1	1
LS1	2
MIC1	3
MIC+1	4

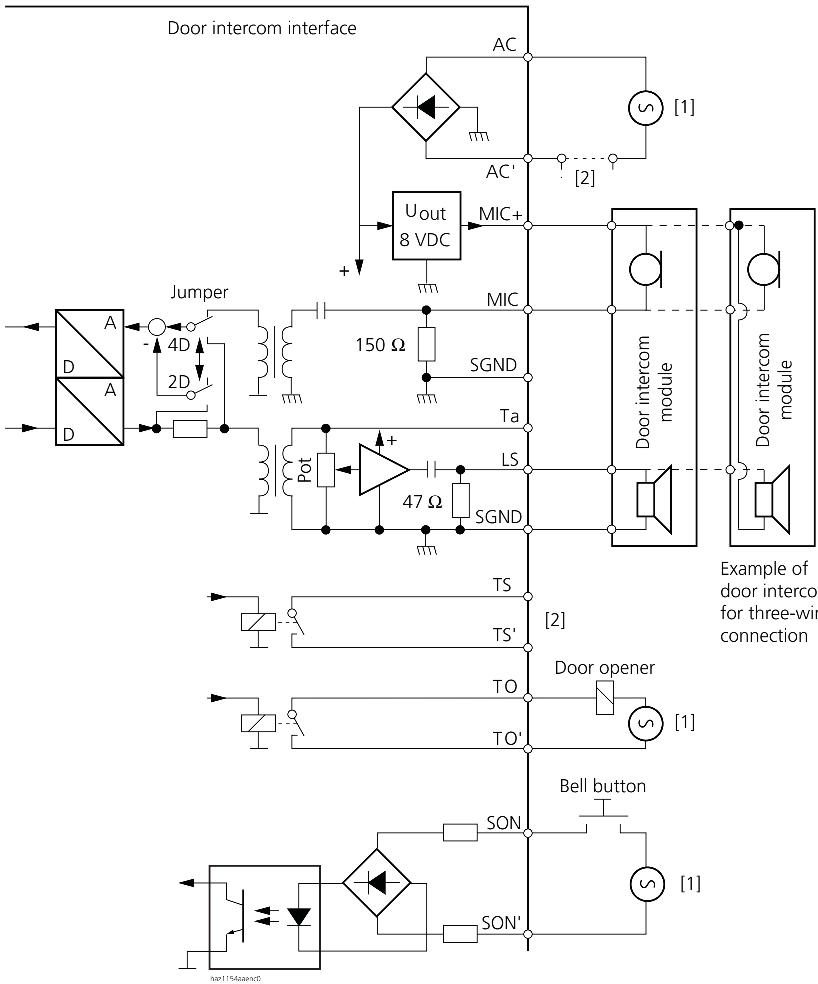
Tab. 4.67: Wiring for door intercom interface 2 (door intercom x.2)

Signal Interface x.2	16-wire cable		
	Pin	Stranded element	Core colour
MIC+2	1	1	white
MIC2	2		blue
LS2	3		turquoise
SGND2	4		violet
Ta2	5	2	white
Tb2 (= SGND2)	6		orange
TS2	7		turquoise
TS'2	8		violet
TO2	9	3	white
TO'2	10		green
AC2	11		turquoise
AC'2	12		violet
SON2	13	4	white
SON'2	14		brown
—	15		turquoise
—	16		violet

Tab. 4.68: Requirements for door intercom connecting cables

Core pairs x cores	4 x 4
Stranded	recommended
Wire diameter, core	0.4…0.6 mm
Screening	recommended for microphones

7.5.5.1 Connecting a TFE without amplifier



[1] Bell-ringing transformer as per EN 60742 (all the AC sources shown may be one and the same transformer).
[2] The power supply for the interface and / or the power supply for a door intercom system or a system panel can be connected via the contact TS, TS'.

Fig. 4.76: Schematic circuit diagram: Connecting a TFE without a separate amplifier

Connection of a simple loudspeaker microphone module or of a loudspeaker (4...8 Ω) and an electret microphone with integrated preamplifier:

- Fit jumper TFE1 and TFE2 for door intercom interface 1 and 2 in position 4D.
- To power the interface connect an external bell-ringing transformer (8...12 VAC) with the AC and AC' connections.
- Make sure the bell-ringing transformer's no-load voltage is < 16 VAC and that a minimum current of 0.6 A (for 12 V) is available in the case of an announcement system.
- Connect the microphone to MIC and MIC+, using a screened cable if necessary (observe the correct polarity).
- Connect the loudspeakers to LS and signal ground (SGND). If the full output power is to be used, use loudspeakers with a min. load-carrying capability of 3 W.
- Set the loudspeaker volume on potentiometer Pot1 and Pot2.
- Connect the door intercom (wire-wire connection) to MIC, MIC+ and LS.

7.5.5.2 Connection two-wire

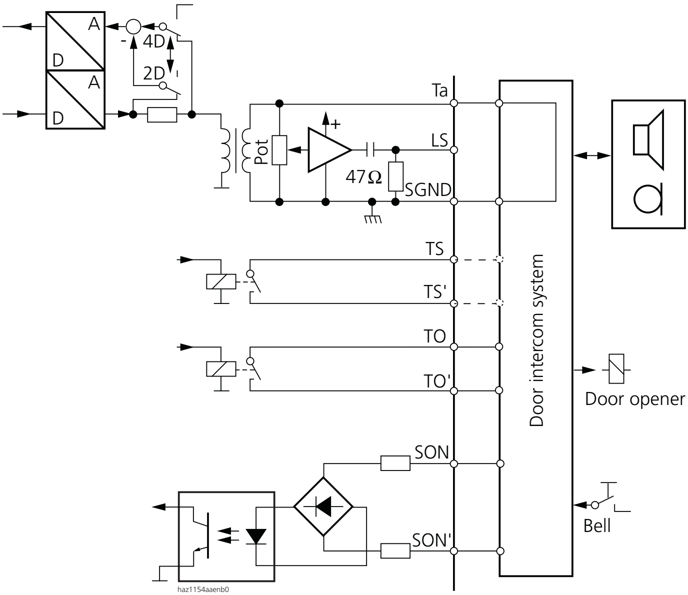


Fig. 4.77: Schematic circuit diagram for two-wire connection

- Fit jumpers TFE1 and TFE2 for door intercom interface 1 and 2 in position 2D.
- Do **not** connect AC and AC’ (the interface signal circuit does not need a power supply).
- Connect the voice path (a/b, 600 Ω, DC-free) to Ta and signal ground (SGND).

The interface volume control (Pot1 and Pot2) is not effective and the volume is set on the door intercom system.

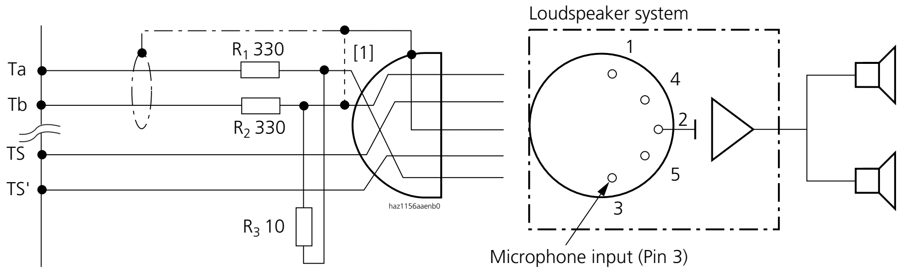
7.5.5.3 Connection four-wire

- Fit jumper TFE1 and TFE2 for door intercom interface 1 and 2 in position 4D.
- Connect the microphone (with a screened cable if required) to MIC and signal ground SGND.

- If the door intercom interface is DC-free and has a high input impedance (≥ 600 Ω), connect the output path (in the loudspeaker direction) to Ta and signal ground SGND.
- If the door intercom interface is not DC-free and if it has a low input impedance (< 600 Ω), connect the output path (in the loudspeaker direction) to LS and signal ground SGND. In this case power the interface at the AC and AC’ connections via an external bell-ringing transformer with 8...12 VAC and set the volume control to minimum.

7.5.5.4 Connecting a loudspeaker system

The loudspeaker system is connected via a line input (e.g. audio input for advertising texts) or a microphone input. The input should have a priority circuit and a separate volume control. The priority circuit is used to fade out the background music when an announcement is being made.



[1] Jumper (1-2) for asymmetric input
R_1, R_2, R_3 voltage divider when using a microphone input

Fig. 4.78: Schematic circuit diagram of the connection variant for a loudspeaker system

- When using a microphone input on the loudspeaker system:
  - Since microphone inputs are usually designed for low signal levels only, use a voltage divider to reduce the signal from the door intercom interface (R_1, R_2, R_3).
  - Use a screened cable for the microphone wires for the connection between the PBX and the loudspeaker system.
  - Connect the cable screening to the loudspeaker system only, **not** to the PBX or else ground loop humming will occur.
- Connect the loudspeaker system input to Ta and Tb.
- Connect the priority circuit to TS and TS’.

Note:
Make sure there are no loudspeakers in the vicinity of the terminals used to make announcements (to prevent acoustic feedback).

7.5.6 Switch group interface

The switch group 1 routing element can be controlled via control inputs ME1 and ME2 on the OI-2DOOR special card. Control is effected using external switches (door contacts, time switches, etc.) which switch the appropriate inputs to the PBX ground (GNDC). The signal no-load voltage is approx. - 40 , the short-circuit current approx. 4 mA.

The permissible switch and loop resistances are as follows:

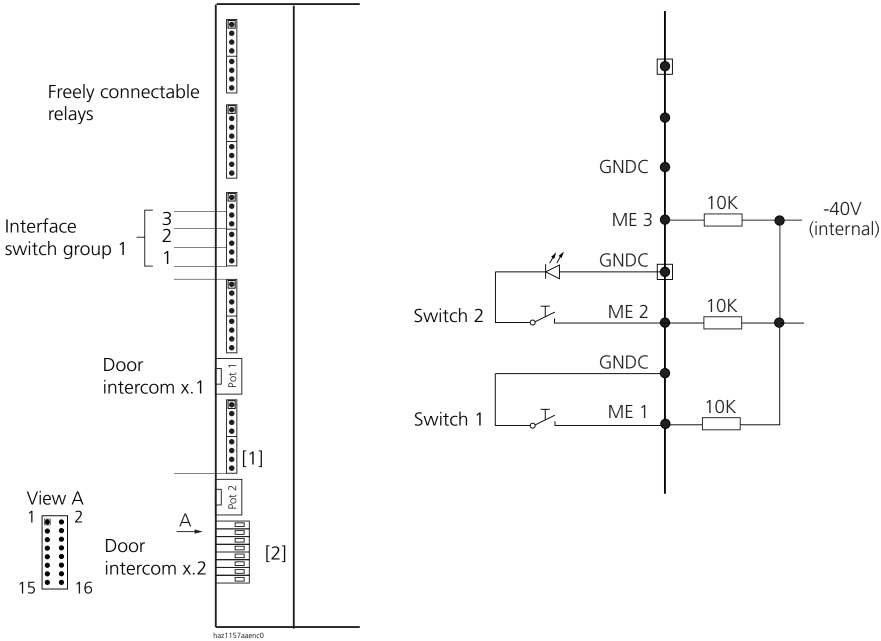
- Active state: < 4.5 kΩ
- Passive state: > 11 kΩ

There are no special requirements for the cables.

Control of the switch group using the control inputs takes priority over control using */# procedures.

Tab. 4.69: Control of switch group 1 via the control inputs

Switch positions for switch group 1	ME1	ME2
Position 1	Off	Off
Position 2	On	Off
Position 3	Random	On



[1] Direct connection
[2] Indirect connection; connector for prefabricated 16-wire connecting cable
▣ = Pin 1

Fig. 4.79: Control inputs on the OI-2DOOR special card

Tab. 4.70: Line assignment for control inputs

Interface	Connection	Pin
Control input 2	GNDC	1
	ME2	2
Control input 1	GNDC	3
	ME1	4

If required, signalling LEDs **without** series resistor can be connected in series with the external switches.

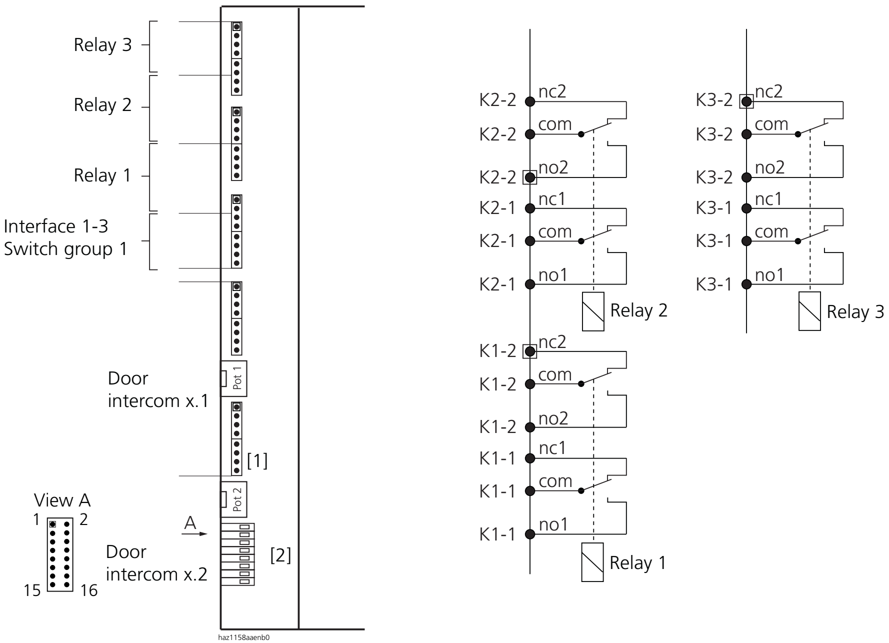
7.5.7 Freely connectable relay contacts

The freely connectable relay contacts can be used to control heating, alarm or outside lighting systems (possibly via external relay for 230 VAC).

There are no special requirements for the cables.

Tab. 4.71: Operating data

Number of changeover switches	2
Insulation between the changeover switches	0.5 kV
Contacts per changeover switch	com: common nc: normally closed no: normally open
max. contact loading	24 VDC, 30 VAC, 1A



[1] Direct connection
[2] Indirect connection; connector for prefabricated 16-wire connecting cable

▣ = Pin 1

Fig. 4.80: Freely connectable relays on the OI-2DOOR special card

Tab. 4.72: Allocation of relay contacts to connectors

Relays	Contact	Pin
3	K3nc2	1
	K3ncom2	2
	K3no2	3
	K3nc1	4
	K3com1	1
	K3no1	2
2	K2nc2	3
	K2com2	4
	K2no2	1
	K2nc1	2
	K2com1	3
	K2no1	4
1	K1nc2	1
	K1com2	2
	K1no2	3
	K1nc1	4
	K1com1	1
	K1no1	2
	(GNDC)	3
	(ME3)	4
	(GNDC)	1
	(ME2)	2
	(GNDC)	3
	(ME1)	4

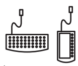
8 Installing terminals

8.1 System terminals

Accesses

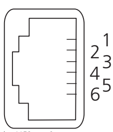
The connections on the underside of the system terminal are identified by the following symbols.

Tab. 4.73: Connection symbols on the system terminals

Symbol	**Connect at this symbol**
 haz1255aaxxa0	Handset (coiled cable)
 haz1256aaxxa0	Connection socket (connecting cable)
 haz1257aaxxa0	Supplementary equipment EKP, AKB and ZTF (connecting cable)
 haz1258aaxxa0	Power supply (power supply unit)

AD2 interface

Tab. 4.74: User-network interface AD2 on the terminal

RJ11 socket	Pin	Function
 haz1159aaxxa0	1	—
	2	—
	3	b
	4	a
	5	—
	6	—

Terminal selection

Two Office system terminals can be connected to an AD2 interface. The PBX can only differentiate the two terminals by the position of the address switch on the terminal. The following settings are possible:

- TSD1
- TSD2

Type of set

The set type (feature phone or key phone)is determined by during the configuration of the installation, when the lines are also assigned to the line keys.

8.1.1 Office 10

The system terminal is a desktop model. A wall-mounted bracket is available as an option.

Installation of the desktop model

1. Feed the connecting cable through the strain relief on the handset rest.
2. Position the handset rest as required and put the handset in place.

Installation of the wall-mounted bracket (Option)

1. Feed the connecting cable through the strain relief on the wall-mounted bracket.
2. Screw the wall-mounted bracket onto the wall using the screws supplied and hook the handset into position.

Connection

1. Set the AD2 bus address (Fig. 4.81).
2. Plug the connector into the socket-outlet.
3. To register the terminal: Press the Foxkey twice.
   The acknowledgement tone is heard.
4. If the system is configured, test the operation of the terminal.
5. Label the telephone.

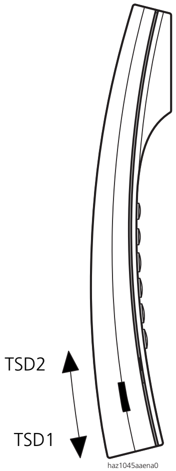


Fig. 4.81: Set the AD2 bus address

Note:
Make sure the TSD (address switch) is pushed in as far as the stop or the switchover will not function correctly.

If the LED flashes slowly, the wrong terminal type is configured.

8.1.2 Office 25, Office 35, Office 45

Installation

1. Set the AD2 bus address under the terminal's designation label.
   – TSD1 = Address switch not pressed (disengaged)
   – TSD2 = Address switch pressed (engaged)
2. Plug the coiled cable into the underside of the Office.
3. Feed the coiled cable through the strain relief.
4. Plug the coiled cable into the handset.
5. Plug the connecting cable to the terminal and feed it through the strain relief.
6. Plug the connecting cable into the socket.
7. If the system is configured, test the operation of the terminal.
8. Label the telephone.

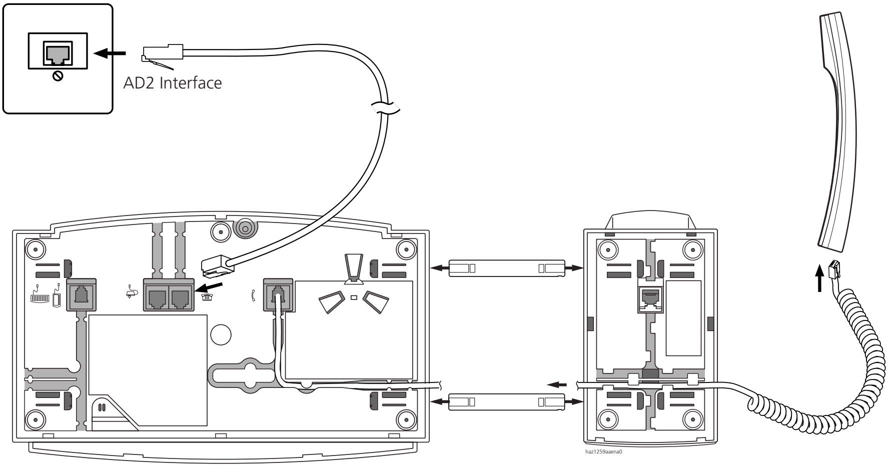


Fig. 4.82: Fit the connecting cable as shown in the example with the Office 45

Note:
The total length of the cables from the PBX to the Office terminal must not be less than 10 m.

Expansion keypad (EKP) for Office 35 and Office 45

The EKP adds 10 configurable keys to an Office 35 / Office 45. Up to three EKP can be connected to each terminal. If an alpha keyboard is connected, only two other EKPs can be connected.

The EKP contains a connection socket for another EKP or the alpha keyboard.

Installation and commissioning

1. Unplug the terminal cable from the socket.
2. Plug the connecting cable of the expansion keypad into the socket for supplementary equipment on the underside of the terminal or EKP.
3. Feed the connecting cable through the strain relief.
4. If required, secure the EKP to the terminal or EKP (see Fig. 4.83).
5. Plug the terminal connecting cable into the socket.

Secure the expansion keypad EKP to the terminal or expansion keypad

1. Plug the two connecting strips on the right or left into the connecting sockets of the EKP until you hear them click into place.
2. Fit the projecting extremities of the connecting strips into the connecting sockets of the terminal or EKP.
3. Carefully push the two housings together.

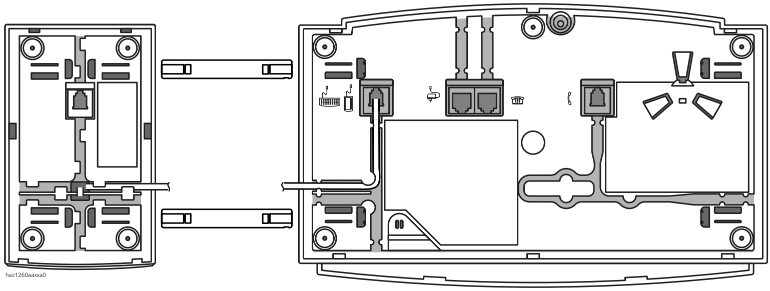


Fig. 4.83: Installation of the EKP as shown with the example of the Office 45

If the EKP is connected to an Office 35, the LEDs will be lit red. If the EKP is connected to an Office 45, the LEDs will be lit red and yellow.

Alpha keyboard AKB for Office 35 and Office 45

The AKB simplifies the use of alphanumeric dialling. It simplifies text input and the configuration of the Office 35 / Office 45. The AKB can also be used to operate the Office terminal and run the main telephony functions using function keys or shortcut keys.

1 AKB can be connected to each terminal.
The AKB can be connected directly to the Office terminal or to an expansion keypad EKP.

Installation and commissioning

1. Unplug the terminal cable from the socket.
2. Plug the connecting cable of the AKB into the socket for supplementary equipment on the underside of the terminal or of an EKP.

3. Feed the connecting cable through the strain relief on the terminal and on the EKP.
4. Plug the terminal connecting cable into the socket.

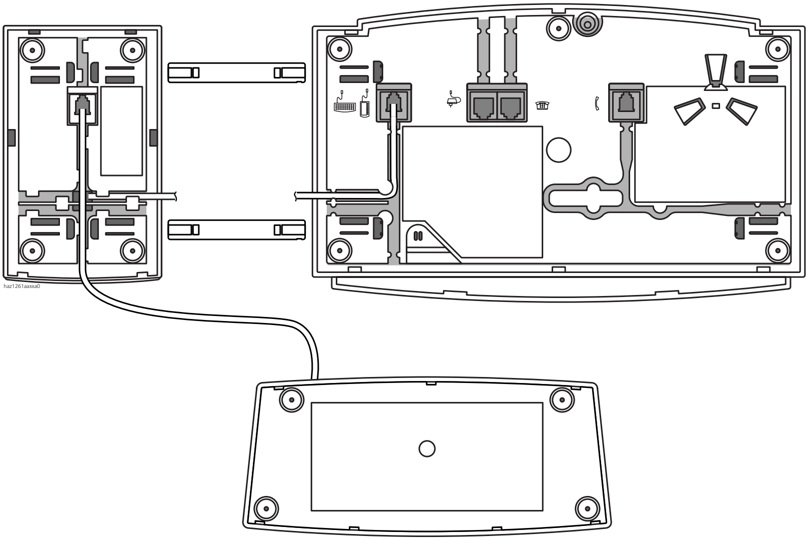


Fig. 4.84: Installation of the AKB as shown in the example of the Office 45 with EKP

8.1.3 Office 20, Office 30, Office 40

Installation

1. Set the AD2 bus address under the terminal's designation label.
   – TSD1 = Address switch not pressed (disengaged)
   – TSD2 = Address switch pressed (engaged)
2. Plug the coiled cable into the handset.
3. Plug the connecting cable to the terminal and feed it through the strain relief.
4. Plug the cable into the socket.
5. If the system is configured, test the operation of the terminal.
6. Label the telephone.

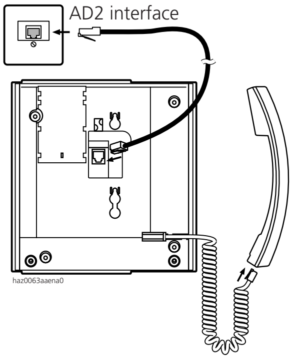


Fig. 4.85: Installation as shown with the example of the Office 40

Additional keypad ZTF for Office 30 and Office 40

The ZTF adds 20 configurable keys to an Office 30 or Office 40.

Installation

1. Unplug the terminal cable from the socket.
2. Connect the ZTF's connecting cable with connector X4 through the opening provided for that purpose (Fig. 4.86).
3. Guide the opening on the ZTF over the column of the Office and allow the ZTF to snap into place.
4. Fit the supplied column to the ZTF.
5. Screw the ZTF on to the Office.
6. Plug the terminal connecting cable into the socket.

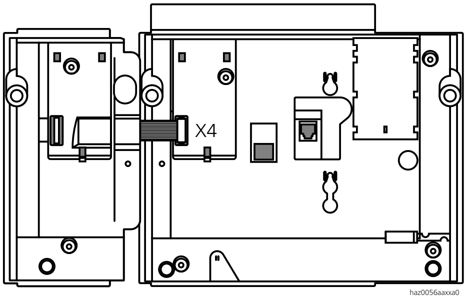


Fig. 4.86: Installation on the ZTF as shown in the example of the Office 40

If the ZTF is connected to an Office 30 , the LEDs will be lit red. If the ZTF is connected to an Office 40, the LEDs will be lit red and green.

8.1.4 Audio

Audio is used in connection with Office terminals whenever:

- A monitoring facility is required via a second handset (Microtel MT)
- A monitoring facility is required via an external loudspeaker
- A call is to be recorded using recording equipment

Note:
National laws governing the recording of phone calls must be observed.

Recording with Audio must always be started / stopped manually (even when using voice-actuated recording equipment). Otherwise there is a risk that office conversations will be recorded even when the handset is on-hook (idle).

Power supply

An external plug-in power supply unit (9 VDC) is supplied as the power supply for Audio.

Connection and views of the equipment

Audio is connected to the terminal between Microtel (MT) and the handset socket.

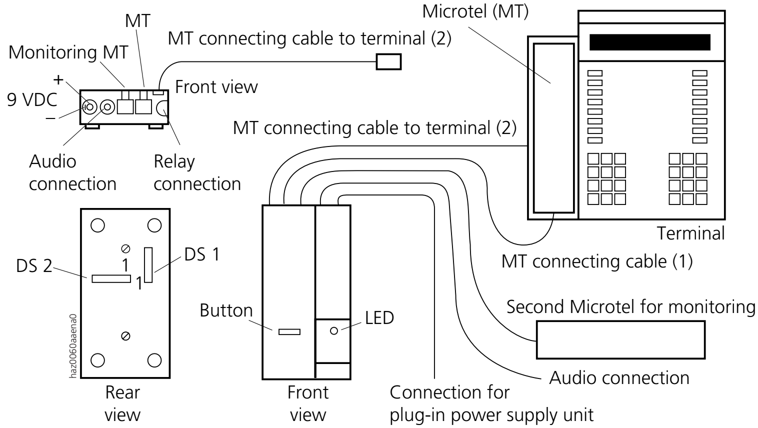


Fig. 4.87: Connection and views of the equipment

1. Unplug the MT connecting cable (1) from the terminal and plug it into the MT output socket of Audio.
2. Plug the MT connecting cable (2) of Audio into the handset socket of the terminal.

8.1.5 PC Operator Console Office 1550

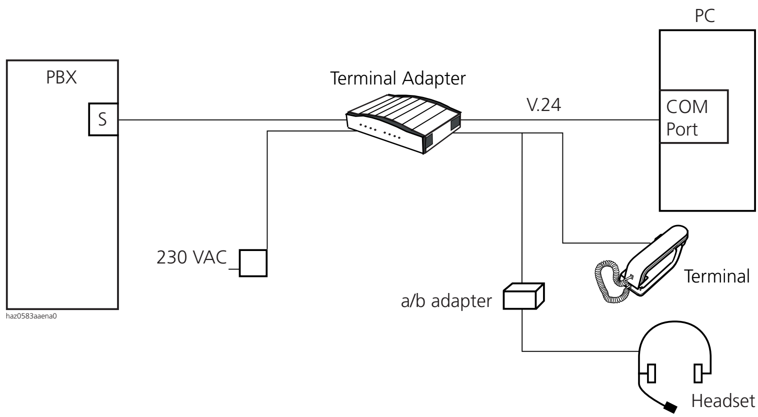


Fig. 4.88: Connection concept for the PC Operator Console

Terminal Adapter

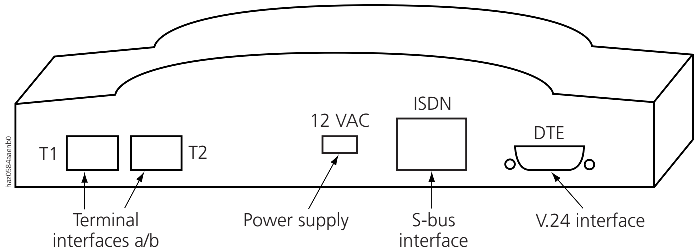
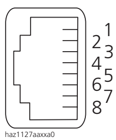


Fig. 4.89: Connections on the Terminal Adapter

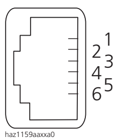
Interfaces

- S interface (see Tab. 4.75)
- V.24 interface (see "V.24 Interface", page 788)
- a/b interface on the telephone (see Tab. 4.76)
- a/b interface on the Terminal Adapter (see Tab. 4.77)

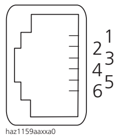
Tab. 4.75: S interface on the Terminal Adapter

RJ45 socket, 8-pin	Pin	Designation	Colour
	1	—	—
	2	—	—
	3	c	white
	4	f	blue
	5	e	violet
	6	d	turquoise
	7	—	—
	8	—	—

Tab. 4.76: a/b interface on the telephone

RJ12 socket	Pin	Designation	Colour
	1	—	—
	2	b	green
	3	—	—
	4	—	—
	5	a	red
	6	—	—

Tab. 4.77: a/b interface on the Terminal Adapter

RJ11 socket	Pin	Designation	Colour
	1	–	–
	2	–	–
	3	a	red
	4	b	green
	5	–	–
	6	–	–

Installation

Note:
The S interface to which a PC Operator Console is connected must be configured with mode "V2".

1. Switch off the PC.
2. Plug the S-bus cable into the S-bus socket on the Terminal Adapter.
   Plug the other end of the cable into the telephone socket.
3. Connecting the model cable:
   – Plug the 9-pin connector into the V.24 socket on the Terminal Adapter.
   – Plug the 9-pin or 25-pin connector into the PC's COM port.
4. Plug the cable of the telephone set or a/b adapter into terminal socket T1 on the Terminal Adapter. If both are being used at the same time, use the supplied intermediate cable.
5. Connect the headset to the a/b adapter.
6. Connect the power cord.
7. Set the Terminal Adapter.
8. Install the PC Operator Console application.
9. Set up remote data transmission for PPP communication (see also the Application Notes on the PC Operator Console CD).

For further information please refer to the Installation Instructions on the PC Operator CD.

8.2 Ascotel DECTCordless system

8.2.1 Location

The locations determined for the handsets, charging bays and radio units during the planning phase need to be checked against the following criteria:

- Influence on radio operation
- Ambient conditions

Influences on radio operation

Radio operation is affected by the following influences:

- Outside interference (EMC)
- Obstacles in the surrounding area affect the radio characteristic

To achieve optimum conditions for radio operation, observe the following points:

- Optimum radio operation depends on the radio unit → handset line of sight.
- Walls act as an obstacle to the propagation of radio waves. Losses depend on the wall thickness, construction material and reinforcement used.
- Do not place radio units and handsets in the immediate vicinity of TV sets, radios, CD players or power installations (for reasons of EMC, e.g. distribution boxes, rising power lines).
- Do not place radio units and handsets near X-ray installations (EMC).
- Do not place radio units and handsets near metal partitions.
- Observe the minimum distance requirements between adjacent radio units (see Fig. 4.91).
- Minimum distance between handsets for fault-free operation: 0.2 m. (The charging bays of the Office 135 can be linked using connecting strips. However, operating several phones on interconnected charging bays can lead to malfunctions.)
- Minimum distance between charging bays with handset on-hook for fault-free operation: 0.2 m.

Ambient conditions

Tab. 4.78: Ambient conditions

Room class	C
Operating temperature	0...40°C
Relative humidity	30...80%

- When installing: Ensure convection (space for ventilation).
- Avoid excessive dust.
- Avoid exposure to chemicals.
- Avoid direct sunlight.

Note:
If these requirements cannot be met (e.g. outdoor installation), use the appropriate protective housing.

See also:
"Planning DECT systems", page 611".

8.2.2 Installing the radio units

Do **not** remove the cover of the radio unit. (Warranty protection will lapse if the cover is removed)

Fit the mounting bracket (see Fig. 4.90 dimensional drawing for wall mounting). Observe the minimum distances (see Fig. 4.91).

Position the AD2 socket(s) near the radio unit.
Each radio unit requires one AD2 bus (two optional on the SB-8): Do not connect any other terminals.

Up to a line length of 660 m (wire diameter 0.5 mm) the SB-4 radio unit can be powered from the PBX via the AD2 bus. If the line is longer, an outside power supply is required locally. Special 230 V plug-in power supply units (9...15 VDC, 400 mA) can be supplied for this purpose.

The SB-8 radio unit can be powered from the PBX up to the maximum line length of 1200 m specified for operation (wire diameter 0.5 mm).

Warning: SB-4 and SB-8 require plug-in power supply units with different contact assignments, see Tab. 4.79 (the plug-in power supply unit for SB-8 is the same as the power supply unit for the Office 135 charging bay).

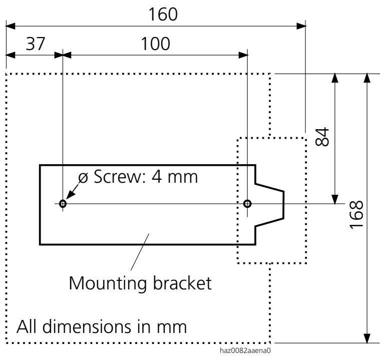
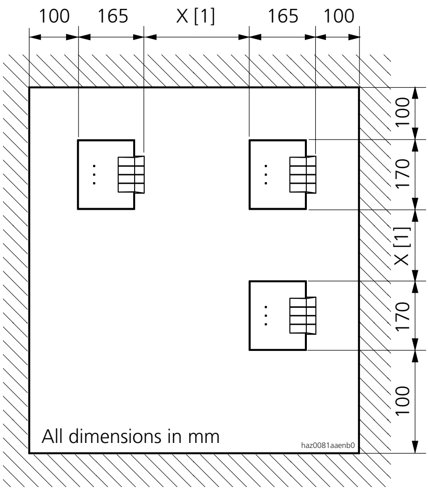


Fig. 4.90: Dimensional drawing for wall-mounting the mounting bracket



[1] X = 200: Minimum distance if the radio units are connected to the same PBX (synchronous)
X = 2000: Minimum distance if the radio units are not connected to the same PBX (not synchronous)

Fig. 4.91: Installation distances

Connecting the radio unit

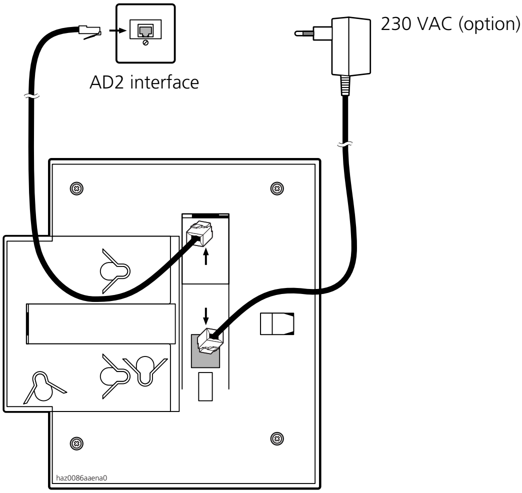
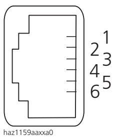


Fig. 4.92: Underside of the radio units with connection points

Tab. 4.79: Connections on the Ascotel DECT radio units

RJ12 sockets	Pin	Socket 1: AD2 interface		Socket 2: Power supply	
		SB-4	SB-8	SB-4	SB-8
	1	Local power supply +	Local power supply –	Local power supply +	Local power supply –
	2	Local power supply –	b2	Local power supply –	—
	3	b	b1	—	—
	4	a	a1	—	—
	5	—	a2	—	—
	6	—	Local power supply +	—	Local power supply +

Tip:
If an SB-8 radio unit is operated with two AD2 interfaces, the rack-version connection module S/T can be used for the wiring. Socket 1 of the radio unit is then automatically correctly wired with the neighbouring

ports of the AD2 interface card. If a plug-in power supply unit is required, it is connected to socket 2.

Tab. 4.80: Operation LED (displays)

LED flashing (two LEDs on the SB-8)	Information
green	Operating state
red / green	Startup procedure running
orange	Transmission of DECT sequences
red	Fault
not flashing and not lit	Radio unit defective or not in operation

For further display variants, see "State display for cordless system", page 932

Charging bay

Place the charging bay on a pad to prevent its plastic feet from damaging polished or varnished furniture surfaces. The manufacturer assumes no liability for damage caused to furniture or sets.

A connecting cable with 230 VAC plug-in power supply unit is included in the equipment supplied.

8.3 teleCOURIER 900 paging system

See the teleCOURIER 900 documentation. The system function and its features are described in "teleCOURIER 900 Paging System", page 128.

9 Checking the installation

Carefully check the following points to prevent any malfunctions.

Loop resistance of the analogue exchange line

1. Determine and record the loop resistance or attenuation.
2. Configure the attenuation of the analogue network interfaces (network card TC-04AB) accordingly:
   Attenuation≤ 2 dB or loop resistance R ≤ 280 Ω: "Short" or "Short D"
   Attenuation> 2 dB or loop resistance R > 280 Ω: "Long" or "Long D"

S bus

- Have the terminating resistors been fitted to the last socket-outlet on the S bus?
- Are the values for the resistors used correct? (100 Ω)
- Are all cable screens conductively connected to the PBX housing or the main distribution board / patch panel?

Protective earth and equipotential bonding

- Does the installation comply with the earthing concept? (See "Earthing", page 720)

Checking for earth loops:

1. Disconnect the PBX from the 230 V mains and the USP where applicable.
2. Disconnect the earth connections (yellow / green) on the PBX and the main distribution board / patch panel.
3. Connect the ohmmeter between the earthing terminal on the PBX or the main distribution frame / patch panel and the building potential (see Fig. 4.93).
   The resistance should be greater than 1 MΩ.
4. Remove the ohmmeter.
5. Reconnect the earth connections on the PBX.
6. Reconnect the PBX from the 230 V mains and the USP where applicable.

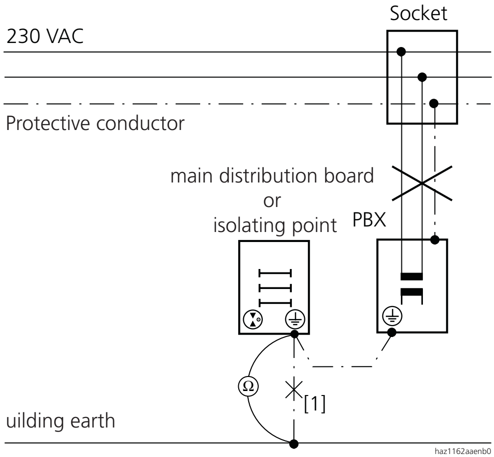


Fig. 4.93: Measuring the resistance between PBX and building potential

Part 5 Configuration

1 Overview of Chapters

AIMS configuration tool

The AIMS Configuration Manager helps to create and manage the system's customized features. Chapter 2 shows you how to enter directly on the screen the settings for the PBX and the concept for internal and external access.

Enabling local access

Chapter 3 features the four authorization levels for local access. Authorization regulates the settings that a subscriber is entitled to make or modify. Other topics include the syntax and updating of passwords, and the recording of access or failed access attempts to the system's configuration in the PBX's access log.

Enabling remote access

Remote access can be enabled or barred using AIMS or */# procedures. The */# procedures can be stored under a function key. Information on enabling a one-off or permanent remote access can be found in Chapter 4.

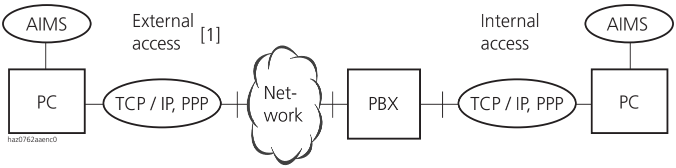
Configuration Steps

The actual configuration work for the PBX / PINX begins once the system has been initialized using the hardware or AIMS. Chapter 5 describes the sequence in which the numbering plans, subscribers, call distribution and routing elements should be defined. It also describes the configuration processes for cordless systems. The procedure for reading the initialization values is described at the end of the Chapter.

2 AIMS configuration tool

2.1 Access concept

With AIMS an Ascotel IntelliGate PBX can be configured locally (internal access) or remotely (external access) using a standardized communication protocol.



[1] Access via Network Termination (NT) or down-circuit from a PBX.

Fig. 5.1: General connection concept for PBX configuration

The PBX can be configured not only via the internal and external access but also via a LAN access.

The connection possibilities for the various accesses are described in "Connection options", page 837.

Configuration using AIMS

The Configuration Manager featured in AIMS provides a menu-prompted software tool to help you set up and manage your customer data conveniently and easily. It is designed to enable:

- System Configuration in offline mode independently of location and time
- system Configuration via internal or external access in online mode
- flexible and quick adaptation of the configuration and customer data
- data management for a large number of PBXs

The AIMS configuration tool is used for configuring all the data. All the configuration and customer data is set up or modified via AIMS.

AIMS does not depend on the sales channel or language. A sales channel is defined for each new PBX created with AIMS. With the command "Tools – Set/ Change Sales Channel" in the AIMS Shell the PBX carries out an initialization with the corresponding initialization values.

Note:
AIMS 6 is backwards compatible, i.e. AIMS 6 can also be used to configure ISDN 5 systems[1].

Besides the Configuration Manager, AIMS also features a number of other Managers for related tasks (see "Ascotel Information Management System (AIMS)", page 144).

Communication protocol

The TCP / IP communication protocol is used for communications between PBX and PC. The PPP protocol is used for serial connections (via V.24 for example). To be able to set up such a connection, you need to ensure that the appropriate PPP software is installed on the PC (see "Settings on the PC", page 841).

2.2 Connection options

Overview of connection options

Tab. 5.1: Connection options for configuration with AIMS

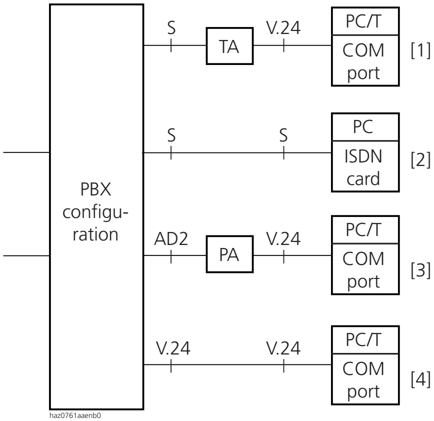
Access	Connection on the PC	Connection on the PBX	Effective data transmission rate
Local access (internal) • via V.24 interface • via Pocket Adapter	 V.24 interface V.24 interface	 V.24 interface AD2 interface	 approx. 130 kbit/s approx. 10 kbit/s
Dial-up access • internal access • external access (via the public network)	 ISDN TA or ISDN PC card ISDN TA or ISDN PC card	 S interface Network interface	 64 kbit/s 64 kbit/s
LAN (Ethernet)	Ethernet interface	Ethernet interface (on the basic system)	10 Mbit/s

1) As of PBX software version 5.2

2.2.1 Connection options for internal access

The internal access to the PBX configuration is possible via S or AD2 user-network interfaces or via one of the two V.24 interfaces of the basic system. The V.24 interface settings are made via the AIMS Configuration Manager.

Note:
The two V.24 interfaces of the basic system have different initialization value settings (see "V.24 Interface", page 788).



[1] PC on Terminal Adapter
[2] PC with ISDN card on S user-network interface
[3] PC on Pocket Adapter
[4] PC directly on the basis system

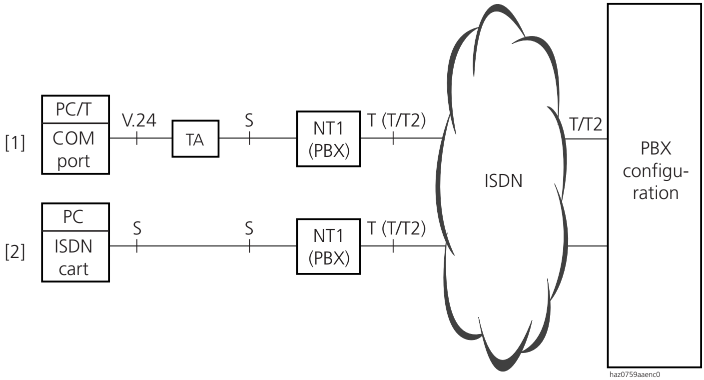
Fig. 5.2: Connection options for internal access for PBX configuration

Cables for internal access

- For [1], supplied Terminal Adapter cable
- For [3] Pocket Adapter cables (see "V.24 interface on the Pocket Adapter (PA)", page 790)
- For [4] V.24 cable (see "V.24 Interface", page 788)

2.2.2 Connection options for external access

An Ascotel IntelliGate can be configured remotely via the public network. This requires a PC with access to the public network either via an ISDN card or a Terminal Adapter, either directly via an NT1 or via a PBX. If the PC has an ISDN card, the connection can be made directly via the S interface.



[1] PC with Terminal Adapter on the S interface of an NT1 or PBX
[2] PC with ISDN card on the S interface of an NT1 or PBX

Fig. 5.3: Connection options for external access for PBX configuration

Cables for external access

- For [1], supplied Terminal Adapter cable

2.2.3 Connection option for LAN access

Access via LAN is connected via the Ethernet interface on the mainboard. The Ethernet interface settings are made using the AIMS Configuration Manager. The current settings can also be viewed with System Assistant. For the initialization values see "Ethernet Interface", page 799.

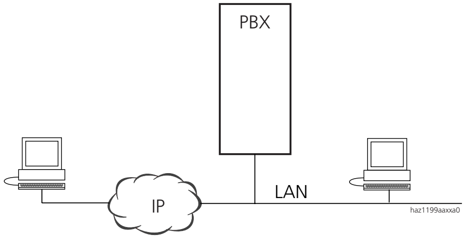


Fig. 5.4: Connection option for LAN access for PBX configuration

Cable for LAN access

- Cable (see "Ethernet Interface", page 799)

2.3 Settings

2.3.1 Settings on the PBX

Remote maintenance has to be enabled in the PBX's system Configuration. The following communication parameters have to be set in the Configuration Manager under "IP Configuration":

- ”IP configuration RDF”: TCP / IP settings for local or dial-up access
- "IP configuration Ethernet": TCP / IP settings for access via LAN

For external access the following parameters must be entered in the data service destination table under "Routing elements: data service":

- Data service: "B channel transparent"
- Destination: Call number of the remote maintenance access (initialisation value: "898")

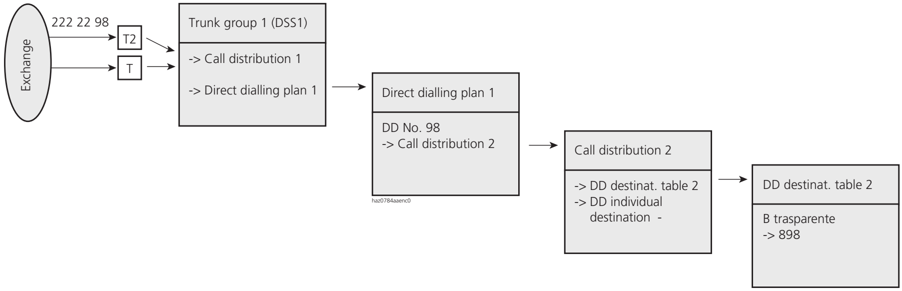


Fig. 5.5: Configuration example for external access

See also:
"Data services", page 353.

2.3.2 Settings on the Pocket Adapter

The following PA settings are required if an Ascotel IntelliGate is configured using a Pocket Adapter (PA) connected in parallel with an Office terminal:

Tab. 5.2: Setting on the Pocket Adapter

S1	S2	S3	S4	S5	S6	S7	S8	Meaning
ON		OFF						PC connection (together with S3 = OFF)
	ON							Pocket Adapter as second terminal
			ON	OFF				RTS / CTS (important for AIMS)
					ON	OFF	ON	9600 (basic speed setting)
					OFF	ON	ON	19200 (max. speed)

The transmission parameters for configuration via PA must match the parameters set in the PBX. 9600 is set as the standard data transfer rate in the PBX.

2.3.3 Settings on the PC

AIMS runs under all the usual Windows operating systems.

The services Dial-Up Networking and TCP / IP network protocol must be installed for local access and dial-up access. For LAN access the TCP / IP network protocol has to be installed.

Note:
The following settings apply to all Windows versions. The procedure may vary from one Windows version to the next. More detailed instructions can be found in the Operating Instructions "Setting up the PC-PBX connection". They are stored in AIMS in the Information Manager.

LAN configuration

The LAN access configuration (e.g. the PC-IP address) depends on the local network. The relevant instructions can be found in the Connection Operating Instructions.

Modem configuration

The following modem configuration has to be carried out for internal and external access via an ISDN card or ISDN Terminal Adapter or for access via Pocket Adapter or V.24 interface:

1. Install a modem with modem driver for PPP connection.
   - For Pocket Adapter PA or V.24 interface use the driver "Ascotel Direct PPP Connection".[1)]
   - For ISDN TA and ISDN cards, use the manufacturer's drivers.
2. Set the other parameters (depending on the Windows version):
   - Select the COM port for connection to the PC
   - Max. speed (V.24 interface): 115200 (initialization: 9600)
   - Max. speed (via Pocket Adapter): 19200 (initialization: 9600)
   - Max. speed (via Terminal Adapter): depends on the manufacturer
   - Activate hardware flow control

Setting Up dial-up networking for local access

The following configuration is required for local access:

1. Create a new dial-up connection (e.g. "Ascotel local").
2. Select the "Ascotel Direct PPP Connection" modem.
3. Select client protocol "TCP / IP".
4. Set the other parameters (depending on the Windows version):
   - Deactivate software decompression.
   - Deactivate LCP expansion

1) The driver can be obtained on the Ascotel home page https://pbxweb.aastra.com and in the AIMS installation path.

5. Set TCP / IP for dial-up networking:
   – activate "Obtain an IP address automatically".
   – Deactivate IP header decompression.
   – Deactivate standard gateway.

Setting Up dial-up networking for dial-up access

The following configuration is required for internal or external dial-up access.

1. Create a new dial-up connection (e.g. "Ascotel ISDN").
2. Select the configured ISDN modem.
3. Enter the connection number:
   – Internal access: 898
   – External access: Local network code, call number and if necessary country code of the PBX requiring maintenance
4. Select client protocol "TCP / IP".
5. Set the other parameters (depending on the Windows version):
   – Deactivate software decompression.
   – Deactivate LCP expansion
6. Set TCP / IP for dial-up networking:
   – activate "Obtain an IP address automatically".
   – Deactivate IP header decompression.
   – Deactivate standard gateway.

2.4 Configuration with AIMS

With AIMS the configuration of a PBX can be directly loaded and processed. The complete configuration of a PBX can also be created on the PC without access to the PBX and stored in the database. The prepared configuration can then be loaded on to the PBX.

Setting Up the connection

1. Open the AIMS Shell.
2. Select the PBX.
3. Under "PBX: Online Connection" select "Local Access", "Dial-up Access" or "LAN".

4. Select the dial-up networking connection under "Online Connection" in the menu "View" "Options":
   – Under "Local AIMS PBX Connection" select the dial-up networking connection for local access or
   – under "AIMS PBX Connection via ISDN" the dial-up network connection for dial-up access.
5. Log on to the PBX via the login (password).
6. Select "Connection" from the "File" menu.
7. Open the Configuration Manager and configure the PBX.

Clearing down the connection

1. Save the database.
2. From the "File" menu select "Disconnect".

See also:
Part of the configuration settings is also accessible via the System Assistant function of the Office 45. A separate set of User's Guide is available for this.

3 Enabling local access

3.1 Authorization

Access to the System Configuration is regulated via 4 authorization levels:

- Authorization level 1: Attendant
- Authorization level 2: System Manager
- Authorization level 3: Installer
- Authorization level 4: Support / Service Centre

Each authorization level allows access to a subset of all the possible settings and functions.

The following condition applies to access with AIMS:

- Authorization level 1 provides access to the fewest settings and functions; authorization level 4, to the most.
- A higher authorization level also provides access to the settings and functions of subordinate authorization levels (exception: The remote maintenance access setting can only be modified at authorization level 1).
- Access at authorization level 4 is reserved for Support and requires a special AIMS set.

Tab. 5.3: Access authorizations depending on the configuration tool

	System Assistant Office 45[1]	Client Management Set	Configuration Set
Access authorization to...			
• Authorization level 1, 2, 3	✓	✓	✓
• Authorization level 4	✓	–	–
Modify the password for the...			
• Authorization level 1	✓	✓	✓
• Authorization level 2, 3	–	✓	✓
• Authorization level 4	–	–	–

[1] Only a subset of the settings and functions possible with AIMS can be operated

The following condition applies to access with the System Assistant on Office 45:

All System Assistant menus are available with all authorization levels. (Exception: The menu for remote maintenance access is available only at authorization level 1)

3.1.1 Passwords

To ensure that the PBX can only be configured by authorized personnel, local access and the option of remote maintenance are password-protected. An access log records all denied access attempts made with erroneous or incorrectly typed passwords.

Tab. 5.4: Passwords after initialization

Authorization level	Initialization password	Access for
1	1ascotel	Attendant / System Assistant
2	2ascotel	System Manager
3	3ascotel	Installer
4	4ascotel	Support / Service Centre

Note:
Once the first System Configuration has been completed, the initialization passwords should be replaced with new passwords (see "Updating passwords", page 847). This ensures that unauthorized personnel cannot manipulate the PBX from a distance once access for remote maintenance has been enabled.

3.1.2 Password syntax

The following rules apply to the selection and format of passwords:

- 4 to 10 alphanumerical characters (upper and lower case irrelevant)
- no German "umlauts", typographical or special characters, blank spaces, etc.

Only one person can have access to the PBX configuration at any given time.

3.1.3 Updating passwords

Any password on any given authorization level can be replaced by a new password on that same level.

If a password on levels 1 to 3 is lost, it is possible with AIMS via V.24 to log into the configuration using the password-free access via the HEX rotary switch – see section below – and then configure a new password.

The password for authorization level 4 can only be changed by Support.

3.2 Access via the HEX rotary switch on the mainboard

The HEX rotary switch on the mainboard provides a password-free local access (only via AIMS and V.24) to the menus of authorization level 3.

- Allow access:
  Turn HEX rotary switch to 5.
- Bar access:
  Turn HEX rotary switch back to 0 (or 2 or B).

Password-free access with the System Assistant on the Office 45 is also possible.

There is no password-free access for remote maintenance.

3.3 Automatic exit from the configuration

The PBX will interrupt access to the system configuration if you do not make any changes to a parameter value or do not make use of the navigation system during the set disconnect time ("View: Options" under "Online Connections").

3.4 Access log

To be able to retrace access activities and access attempts made to the configuration, the PBX records the access log in accordance with authorization level 1 to 4.

The entries contain the following information:

- Date, Login time and duration of maintenance
- CLIP (Calling Line Identification Presentation)

- Access type / login name
- Interface card number, port number, MSN
- Type of access (remote / local)
- CLIP requirement
- Indication of the HEX rotary switch position
- Number of log entries
- Status indication of whether modification(s) has(have) been carried out

This log can be read by each and every password holder.

There are five access logs (i.e. one for each authorization level and one for failed access attempts), each of which stores the last twenty entries:

- Access attempts
- Attendant
- System Manager
- Installer
- Support / Service Centre

3.4.1 Retrieving the log data

The PBX monitors and saves all the accesses as well as failed access attempts. These lists can be retrieved locally or remotely.
For access requirements, see "Connection options for internal access", page 838 and "Connection options for external access", page 839.

In the case of a local access, the data field "Subscriber Name" differentiates between access to the configuration menu via PC with ISDN card, Pocket Adapter or V.24. In AIMS the login name is also recorded, where available.

CLIP verification

If the setting "CLIP required?" is set to "Yes" in the configuration, remote retrieval is possible only if the retrieving party logs in using a CLIP. The CLIP number is also recorded by the access log.

Identification of the AIMS Manager

The CLIP can be used subsequently to determine whether the maintenance or retrieval process was carried out via the "Support / Service Centre" using "AIMS Remote Maintenance" or via "AIMS Configuration".

Retrieving the log via AIMS

All access logs can be retrieved via AIMS.

3.4.2 Entering the processes in the log

Each access attempt generates an entry in the corresponding list.
In the case of a remote maintenance access an entry will not be generated if remote maintenance is barred or if "CLIP = required?" is set to "Yes" in the configuration and no CLIP is received.

Access attempts in accordance with the authorization level

The Access Attempts List logs all the failed attempts to access the PBX.

- If an access attempt via AIMS was denied due to the fact that another user is already carrying out a PBX configuration, the value "0" is entered in the corresponding list as "duration of the maintenance".
- If a change in the authorization level with password prompt occurs during the configuration phase, the change will also generate an entry in the list with the new authorization level.
- If the HEX rotary switch is on position "5", each access attempt will be entered in the access log under the "Installer" list

4 Enabling remote access

4.1 Access enabled by local users

Remote access can be enabled in three ways:

- via */# procedures (see page 851)
- via the AIMS Configuration Manager at the Attendant authorization level
- Via the System Assistant on the Office 45 at the Operator authorization level

It can be revoked again automatically or manually.

Both enabling procedures have equal authorization status. This means that remote access can be enabled using, for example, an */# procedure and then barred again using the "Remote maintenance" setting in the menu of the Attendant authorization level.

When remote maintenance is activated, the event message "Remote Maintenance Enabled" is sent to the local printer and to all the terminals registered in message group 8 (System 2025 / 2045) and 16 (System 2065).

Remote access can be enabled or barred using the */# procedures both from the idle state and the talk state, e.g. after an enquiry.

The authorization to activate / deactivate remote access using */# procedures can be allocated in the subscriber configuration.

When the system is initialized, the authorizations of all subscribers are restricted.

Note:
It is advisable not to keep the remote access authorization permanently activated (signalled by the active status of the LED on a function key configured for remote maintenance on an Office terminal or on the menu of the Attendant authorization level). This ensures that the PBX cannot be manipulated from a remote location by unauthorized persons.

4.2 */# procedure for remote access

Tab. 5.5: */# procedure for remote access authorization

Enable / bar a one-off remote access	*754 / #754
Enable / bar a permanent remote access	*753 / #753

When remote access is enabled using procedure *754, access will automatically be barred again once the remote configuration process has been completed. It is possible to bar access manually using #754 before a remote configuration process has been initiated.

Remote access can be enabled permanently using the procedure *753. To bar access, the authorized subscriber must enter the procedure #753 manually.

The enabling or barring of remote access authorization using the */# procedure is signalled in each case by an acknowledgement tone.

Remote access authorization can also be enabled or disabled in the AIMS Configuration Manager or with the System Assistant function on the Office 45, in each case at the Operator authorization level.

Note:
In a QSIG network it is important to make sure that the authorization to change the remote access is also denied to unauthorized PISN subscribers. Otherwise a PISN subscriber would be able to use an abbreviated dialling number defined for the destination PINX and containing the appropriate procedure to change the remote access authorization to the destination PINX.

4.3 Function keys for remote access authorization

On terminals of the Office family – with the exception of Office 10, Office 20 – the */# procedure for enabling remote access authorization can be stored under a function key, providing the subscriber has the appropriate authorization.

The relevant LED lights up if remote access is enabled once or permanently.

The relevant LED goes off as soon as remote access is denied again, either automatically or manually, using the procedure or the configuration menu or the AIMS Configuration Manager.

Tab. 5.6: Menu example of a one-off remote access on the Office 40 and Office 45

F12:		REMOTE MAINT.	ONCE ONLY
OK	BACK		v

Tab. 5.7: Menu example of repeated remote access on the Office 40 and Office 45

F12:		REMOTE MAINT.	ON
OK	BACK		v

The menus for Office 30 and Office 35 are similarly structured except that the submenu for "Once Only" or "On" is located on a separate page.

5 Configuration Steps

The configuration steps are based on the information determined during the planning and, where applicable, the installation.

5.1 System initialization

Before work can begin, the system has to be brought into a defined state. This involves setting or deleting all the parameter values to their initialization value and carrying out a self-test.

This initialization can be achieved in the following ways:

- With the HEX rotary switch on the mainboard
- With AIMS

5.1.1 Hardware-induced initialization

1. Set the HEX rotary switch on the mainboard to "1".
2. Switch the system off and back on again. The system is restarted.
3. When a flashing "1" appears on the display, turn the HEX rotary switch to "F".
4. When an "F" appears, turn the HEX rotary switch to "0"; the PBX resets the customer data (Flash EPROM).

5.1.2 Software-induced initialization

1. Carry out an initialization in the AIMS Configuration Manager or in the Fault and Maintenance Manager under "Online – Reset PBX". If the sales channel is also to be modified, create a new PBX in the AIMS Shell and select a sales channel. Then under "Tools – Set/Change Sales Channel" carry out an initialization using the corresponding initialization values.

Note:
Carrying out an initialization requires authorization level 3 or 4 (for initialization use the "3ascotel" password for the Installer or "4ascotel" for the Support / Service Centre, or turn the HEX rotary switch on the mainboard to position "5").

2. After initialization the system is ready for configuration. Once the configuration is completed carry out a functional test (see "Part 6 Commissioning").

5.2 Activating the licences

The licence information is stored on the EIM card (Equipment Identification Module) located on the 2025 / 2045 or 2065 mainboard.

The licence information includes:

- The EID (Equipment Identification) serial number of the EIM card
- The sales channel identification CID (Channel Identification)
- The licence code LIC (activated functions, see "Licence-related System and Expansion Limits", page 593)
- The system type (2025, 2045 or 2065)

Each basic system is supplied with a licence certificate containing the above information (without licence code). Please keep the certificate in a safe place.

Tab. 5.8: Example of licence information

Licensing	
Equipment Identification (EID):	81154445474349760E5844D276000035A317
Channel Identification (CID):	0
Licence code (LIC):	0408040158F396792739
System type:	2025

The licences must be activated. The licence code can be edited both online and offline with AIMS:

1. Type in the licence code under "Licensing" in the system configuration. The licence code is stored in the EIM.
2. Upload the system configuration data and restart the system; the licensed function(s) is(are) now enabled. The enabled licences are displayed in the Configuration Manager (under "Licensing", "Licensed attributes" in the basic setup of the System Configuration).

All the features (even those subject to charges) can be configured offline without a valid licence number. When the configuration data is uploaded, AIMS will warn the user that the system does not yet have the required licences.

Each licence code can only be used for one PBX. To licence several systems, you will obtain separate licence codes to match the licence information of the individual systems.

The licence information can be viewed directly from the PBX using the Configuration Manager in AIMS ("Licensing" menu item in the basic setup of the system configuration).

5.3 PBX / PINX configuration

Preparing the system

1. Initialize the system. All the data is deleted and the initialization values are set. (see "System initialization", page 853).
2. Log into the configuration (see "Configuration with AIMS", page 843).

Numbering plan and subscribers

3. Define the numbering plan and subscribers (see "The System's Numbering Plan", page 163).
   – The DDI numbers supplied by the network operator are important for the choice of subscriber numbers.
   – Configure the PISN subscribers.
   – Define the DECT subscribers.
   – Define the pager subscribers.

   If you delete numbers already configured in the numbering plan, those numbers will also be removed from the user groups, busy display variants, announcement groups, call distribution elements.
4. Define the subscriber settings: Name, authorizations, etc.

Call distribution and direct dialling

5. Define the user groups (see "User Group", page 236).
   – Subscriber
   – Call distribution type
   – Operator Console element if required
   – General bell element if required
   – Set the delays

6. Set up the direct dialling plans, DDI numbers and call distribution elements (see "Direct Dialling Plan (DDI plan)", page 223 and "Call Distribution Element (CDE)", page 226).
   A call distribution element is automatically allocated to each DDI number created.
   You can also define ranges for DDI numbers. If so, the question is whether only the DDI numbers that match the internal subscribers are to be created (see "Call Routing", page 267).
7. Edit the direct dialling plans and call distribution elements:
   - If required, adapt the allocation of call distribution elements to the DDI numbers in the direct dialling plan menu.
   - If necessary, use the Call Distribution Elements menu to adapt the destinations in accordance with the switch groups.
8. Create other direct dialling plans and call distribution elements and define them for internal subscribers and internal PISN subscribers. Define the destinations in accordance with the switch groups. Complement the numbering plans accordingly.

Network-oriented routing elements

9. Define the network interface (see "Point-to-Point and Point-to-Multipoint Connections", page 62 and "Trunk Groups of Network Interfaces", page 213).
   - S external to other PINXs
   - Point-to-point connections (TEI management)
   - Point-to-multipoint connections (TEI management)
   - Collision detection for terminals connected in parallel
   - Assign the network interfaces to the trunk groups
10. Define the digital trunk groups (see "Trunk groups", page 212)
   - Name
   - Allocate the direct dialling plan (possibly call distribution element only)
   - Allocate the call distribution element (possibly direct dialling plan only)
   - CLIP information (TON / NPI)
   - DSS1 / PSS1 protocol
   - Private / public
   - Other settings as required
11. Define the analogue trunk groups (see "Trunk groups", page 212)
   - Name
   - Allocate the call distribution element

12. Define the routes (see "Route", page 219)
    - Name
    - Allocate the trunk group
    - Number of outgoing connections
    - NPI for the call number
    - Other settings as required
13. Edit the table of clock sources.

Key Telephones

14. Allocate the line keys of the key telephones to KT lines.
    (see "Key Telephones", page 253).
15. Carry out limited-access settings if required.

Additional settings

16. Basic setup (e.g. adapt passwords, CLIP settings, times, digit barring facilities).
17. Other settings (e.g. additional hardware, Courtesy, paging system).
18. Cordless systems (see "Ascotel DECT configuration", page 857)
19. PINX settings
    - Max. transit PINX
    - Transit route

5.4 Ascotel DECT configuration

Once the DECT radio units have been installed, complement the system configuration. If the system is being configured for the first time, steps 1 and 2 are already covered by the preceding section.

1. Complement the numbering plan with the Ascotel DECT subscriber numbers.

Note:
After the system initialization the radio unit starts in status "AD2 OK". It is operational only once at least one DECT subscriber is entered in the numbering plan (see "Self-test for the DECT system", page 936).

2. Define the subscriber data under "Subscriber settings" (names, authorizations, unobtainable destination, etc.).
3. Configuring Ascotel DECT system parameters.
4. Name the radio units.

5. Log on the DECT handsets.
6. Define the logon period for visitors' handsets (if required).

See also:
"Planning DECT systems", page 611.

5.4.1 Configuring Ascotel DECT system parameters.

Tab. 5.9: DECT parameters valid throughout the system

Parameter	Parameter value	Remarks
DECT recall duration	[10...30 s]	Recall is signalled on a handset for this amount of time if the DECT call connection was previously briefly interrupted.
Message destination	<Subscriber>	see "Logging a visitor handset on and off", page 861
DECT system state	[Active / Passive]	Deactivates the HF power supplies of all radio units. **Warning**: DECT call connections are not possible in the Passive state.
LED radio unit	[On / Off]	The LEDs are switched off only in the normal operating state (green / orange). Flashing sequences during start-up (red / green) or error codes (red) are still indicated.
Compatibility mode	[Mode 1 / Mode 2]	Mode 1: Zero-Blind-Slot mode: All the handsets on the system must support the Zero-Blind-Slot mode (Office 135/135pro, Office 155pro, includes any GAP terminals). Mode 2: Blind-Slot mode: If the handsets used on the system do not support the Zero-Blind-Slot mode (Office 100, Office 150, includes any GAP terminals).
Encryption	[Yes / No]	This parameter may have to be set to "No" if repeaters are used as not all repeaters are capable of handling encryption.

Tab. 5.10: DECT parameters configurable for each radio unit

Parameter	Parameter value	Remarks
Port 1, Port 2[1)]	[Address Port 1, Port 2[1)]]	Used for pairing up two ports of an SB-8 (see "Configuration of ports on an SB-8", page 859).
Status Port1, Status Port 2[1)]	[Not Active / Active]	
Type	[SB-4 / SB-8 / SB-8ANT]	
Registration state	[Registered / Not Registered]	The status of a radio unit is "Not Registered" if it was not possible to allocate it to any location area.

Parameter	Parameter value	Remarks
Local power supply[1)]	[Yes / No]	Specified status of local power supply
Local power supply state[1)]	[Inactive / Active]	Actual status of local power supply
External antenna[1)]	[Yes / No]	Switchover between internal and external antennas on an SB-8ANT

[1)] Only SB-8

Configuration of ports on an SB-8

If an SB-8 is operated on two AD2 interfaces, the ports have to be paired up. We recommend that you always pair up two neighbouring ports to prevent any other radio unit being configured in between. The pairing has to be cancelled again if the ports are to be used for operation with two radio units.

Event message for local power supply on SB-8

In the following cases the SB-8 radio unit deactives the HF and triggers an event message (see "Event types", page 914):

- The "Local power supply" parameter is on "Yes" and the local power supply fails.
- No local power supply is installed and the "Local power supply" parameter changes from "No" to "Yes".

5.4.2 Logging a handset onto the system

Tab. 5.11: Procedure for logging a handset on to the PBX

Procedure	Input in AIMS	Input on handset – not logged on anywhere	Input on handset – already logged on elsewhere
1. Prepare handset for logon.		(ABCD flashing on the display). Switch handset off.	Prepare handset as per Operating Instructions up to the step with the display "Home" / "GAP"
2. Prepare PBX for logon.	• Select the subscriber number in the DECT system configuration • Select "Log Ascotel DECT subscriber on"; the logon procedure is started on the PBX side.		
3. Log handset on.		Switch handset on. Logon is carried out automatically.	Press Foxkey under "New"; the handset is checked for authorization and then assigned its subscriber number and any group membership (user group).
4. Wait for confirmation.		"OK" confirmation on the display.	"OK" confirmation on the display.

Handsets defined in the PBX as Ascotel DECT subscribers but not yet logged on are identified as follows:

- With an * in the numbering plan
- With "DECT not logged on" in the subscriber configuration.

A GAP handset subscriber must also identify himself on the Ascotel DECT system with an access code. The access code must be entered in the PBX and in the GAP handset prior to the logon procedure.

5.4.3 Logging a handset off the system

In the System Configuration under "Mobile: DECT" in the system configuration.

"Select "Log Ascotel DECT subscriber off"; log-off procedure is initiated.

Tip:
The handset identification is deleted if the handset is located within the coverage range of a radio unit; if not, it must be deleted manually on the handset (see Handset Operating Instructions). The Ascotel DECT subscriber number and the data is saved in the PBX (identified with an * in the numbering plan).

5.4.4 Logging a visitor handset on and off

If a Ascotel DECTsubscriber is configured as a visitor, he can be logged on via AIMS (Attendant authorization level) or the System Assistant on the Office 45 for the duration of his visit.
The visitor's destination should be ascertained at the same time so that he can be logged on to the ideal PINX.

In addition to the procedure described above for logging a handset on and off, you also need to specify the duration of the visit in the subscriber configuration. It is advisable to enter the visitor's name.

One hour before the visiting time is due to expire, the system sends the following message to an internal subscriber: "DECT SC No. will be logged off in one hour". This subscriber is configurable. (See Tab. 5.9)
When the visiting time expires, the handset is automatically logged off from the system.

5.5 Reading initialization values

The following configuration can be set to check the initialization values:

1. In offline mode on the AIMS Shell create and open a new system of the type 2045.
2. Open Configuration Manager.

3. In the "Hardware Configuration" add the following hardware:
   - Slot 1: SC-16AD2 (default setting of the terminal type "unknown")
   - Slot 2: TC-04AB
   - Slot 3: ISDN-02PRA
   - Slot 4: (blank)
   - Slot 5: DSP-02
4. In the "Interface Configuration" set interface 0.13 from T to S and the interface type from ETSI to External S.
5. Under "Radio Unit" set up an Ascotel DECT system in slot 0.8.
6. Under "DDI Settings" set up the following direct dialling numbers:
   - Press "New Number".
   - Enter 20 as the start value and 55 as the end value.
   - Answer the prompt "Link Matching Subscribers?" with "Yes, create non-matching also". Do not modify the default settings in the other fields of this dialog box. The DDI numbers 20-55 are set up with the relevant CDE.
7. Under "Subscribers" enter the following terminals for the following user-network interfaces:
   - Interface 1.1-1: Office 10
   - Interface 1.2-1: Office 20
   - Interface 1.3-1: Office 25
   - Interface 1.4-1: Office 30
   - Interface 1.5-1: Office 35
   - Interface 1.6-1: Office 40
   - Interface 1.7-1: Office 45
   - Interface 1.8-1: Office 45pro
8. Set up five DECT subscribers with the following call numbers and handsets and one GAP handset:
   - 50: Office 100
   - 51: Office 135
   - 52: Office 135pro
   - 53: Office 150
   - 54: Office 155pro
   - 55: GAP handset

9. Under "Terminal Data" configure the following key assignments for the defined subscribers, where permissible with each terminal:
   – Number key: On the first level store the call number; on the second, the name.
   – Function key: Save with Call Forwarding Unconditional (CFU).
   – Set up the Team key (all set to subscriber 20).
   – Set up the line key on the key telephone.

Once all the parameters have been entered, the settings in all the AIMS Managers can be checked step by step to detect any inadmissible configurations.

Part 6 Commissioning

1 Overview of Chapters

General Checks and Function check

Before the PBX is commissioned and handed over to the customer, the system is checked to ensure it complies with the project specifications and operates correctly. Chapters 2 and 3 list the main checkpoints for verification.

Customer induction

If the system is found to be working perfectly, the customer is acquainted with the main features and instructed how to operate the terminals. Chapter 4 contains references to the type and scope of customer training courses.

Handover to the customer

Chapter 5 features checklists with all the main points to be taken into account when the system is officially handed over to the customer.

2 General Checks

2.1 Checking the Configuration

When checking the configuration, proceed as follows:

- Compare printouts by the AIMS Project Manager (block diagram, layout) with the project planning data (expansion card configuration, terminals).
- Check whether any customer requests for modifications have been taken into account (for the system configuration, see "Part 5 Configuration").
- If changes were made to the DECT system, check the locations of the radio units. Carry out follow-up measurements where necessary.
- If changes were made to the terminals (e.g. other terminal types), check performances, line lengths, S-bus configuration.

2.2 Visual inspection with the system out of operation

Carry out the following visual inspections before connecting the PBX to the power supply:

- Is the installation correct? (wire colours, S bus terminating resistors, loop resistors, see "Checking the installation", page 833).
- Is the protective earth and equipotential bonding in place (no earth loops)? (See "Checking the installation", page 833.)
- Are the expansion cards correctly fitted?
- Are the jumpers correctly fitted (e.g. on the 2025 / 2045 mainboard, ISDN-04ST ISDN card, OI-2DOOR special card)?
- Is the uninterruptible power supply fitted?
- Are the dummy covers fitted?
- Is the PBX set up for the required supply voltage?
  (e.g. operation with 48 VDC requires the DC-48V module).
- Have the correct connecting cables being used for the various applications? (See "Special Interfaces", page 788.)

2.3 Restart test

When the PBX starts up, it will automatically carry out a self-test complete with RAM test if the HEX rotary switch on the mainboard is set on position 0 or 2.

The RAM can be tested in two ways:

- RAM quick-test (normally approx. 3 seconds):
  HEX rotary switch in position 0.
- RAM thorough test (up to 1 minute, recommended with every new installation):
  HEX rotary switch in position 2.

The system's self-test including the RAM test is signalled by vertical bars moving up and down on the 7-segment display.

Once the startup process has been successfully completed, the decimal point on the 7-segment display flashes regularly (see "Basic system self-test", page 934).

Any errors during the startup process are signalled on the 7-segment display (see "Operating state and error displays", page 930).

If a RAM error occurs, replace the mainboard (see "Replacing the 2025 / 2045 or 2065 mainboard", page 910).

Note:
Never remove any cards while the system is still connected to the power supply.

Putting the PBX into operation

1. Remove the PBX cover.
2. Set the HEX rotary switch on the mainboard to position 0 or 2.
3. Reconnect the PBX to the power supply.
   The PBX starts up.

See also:
Information on the functions of the HEX rotary switch and error codes, see "Part 7 Operation and Maintenance".

2.4 Visual inspections and function tests in normal operation

With the PBX in normal operation, carry out the following visual inspections and function tests:

- Is the decimal point on the mainboard's 7-segment display flashing regularly?
- Are all the expansion cards registered with the system?
- Are all the individual configurations on the terminals working in accordance with the planning specifications and the wishes of the customer (key configurations, phone book entries, menus on the terminal's display, output text language, ringing tones, hands-free mode, etc.)?
- Is the terminal bus addressing correct?
- Do the displays on all the terminals indicate the idle-state display incl. the time?
- Do you obtain a normal dial tone whenever a terminal is seized?
- Is the LED on the radio unit of the DECT system flashing normally?
- Are the handsets of the DECT system synchronized with the radio units (aerial symbol on the display of the Ascotel DECT handset)?

3 Function check

These test instructions are used to carry out a rough check of the system. The system as a whole is tested, i.e. the function test of the in-house wiring is included in the test.

At least two people are needed to be able to check the system efficiently.

If a service or journal printer is available, it can also be used for printing out the test report.

General preparation

Connect two Office terminals to the PBX or main distribution board.

3.1 Check the power supply with the UPS-12V / 48VDC modules

Visual check

- Are the modules correctly fitted?
- Are the modules correctly connected?

3.1.1 Testing the UPS-12V module

Testing 230V mains operation

- The battery charge current, in accordance with the jumper position (see "UPS with UPS-12V module", page 725), should display a value of 1 or 2 A with a tolerance of +/- 20 %.
- The battery current is measured using a clip-on DC ammeter on one of the two wires of the battery cable.
- If there is no current or if the current is quite low, the PBX has to be operated on the battery for approx. 5 minutes before measurement (i.e. without mains power).

Testing in battery operation

- Function test:
  The system must continue to function uninterrupted after disconnection from the 230 V mains. This can be verified by establishing connection.

- Event message:
  On the system terminals configured for this, the event message "PBX Power Supply: Battery" must be displayed after the predefined time.
- Current measurement:
  Depending on the PBX type and level of expansion, the current in the battery cable (current direction reversed relative to mains operation) is between 2 and 15 A.
- Mains failure bridging time:
  The ratio of the battery capacity to the measured current gives an indication of the expected bridging time in the event of mains failure. Example:
  If the battery capacity is 20 Ah and the measured battery current is 5 A, the calculated bridging time is theoretically 20 Ah / 5 A = 4 hours. In practice the actual time is about 75% of this value, i.e. approx. 3 hours.

3.1.2 Testing the 48VDC module

Testing 230 V mains operation (if this operating mode is available)

The current in the DC power supply cable should be less than 0.1 A. This checks that no power is drawn from the DC source when there is a mains power supply.

Test in DC operation

If no mains operation is provided, the following tests is irrelevant except for current measurement.

- Function test:
  The system must continue to function uninterrupted after disconnection from the 230 V mains. This can be verified by establishing connection.
- Event message:
  On the system terminals configured for this, the event message "PBX Power Supply: Battery" must be displayed after the predefined time.
- Current measurement:
  Depending on the PBX type and level of expansion, the current in the DC power supply cable is between 0.5 and 4 A. If the PBX is the only power consumer, the ratio of the battery capacity to the measured current gives an indication of the expected mains failure bridging time.

3.1.3 Testing the external Auxiliary Terminal Power Supply (ATPS)

1. This test requires that the 230 V PBX power supply is not looped via the ATPS auxiliary power supply but connected to a separate socket outlet.
2. Set up a call. At least one terminal has to be connected to an AD2 interface of an expansion card (not the mainboard).
3. Disconnect the ATPS auxiliary supply from the 230 V mains. (Do not disconnect the connecting cable from the ATPS to the PBX connector strip.)
4. The call connection should be interrupted. This proves that the expansion card was being powered not by the internal power supply but by the ATPS auxiliary supply.

3.2 Checking internal connections

Sampling test on the user-network interfaces on the corresponding expansion cards and on the 2025 / 2045 mainboard.

Checkpoints:

- Ringing
- Call connection
- Caller's identity

3.3 Checking external connections

Tests the operation of the network interfaces on the 2025 / 2045 mainboard and the expansion cards.

3.3.1 Outgoing routes

Tests the operation of the routes by dialling the configured route number without the exchange access prefix.

3.3.2 Outgoing Traffic

Sets up the maximum possible number of connections for a trunk group and tests the network interfaces trunk group by trunk group.

Note:
Take note of the following:
- All non-connected network interfaces must be barred.
- Analogue network interfaces reserved for incoming traffic can be tested only by calling the corresponding lines.
- Check the connection quality between analogue network and analogue / digital subscribers in both directions and optimize the attenuation setting of the analogue network interfaces (see "Attenuation on analogue network interfaces", page 604).

3.3.3 Incoming traffic

Sampling test of the configured call distribution by calling main and DDI numbers.

Checkpoints:

- Are the calls distributed in accordance with the System Configuration?
- Are the destinations depending on the switch position of a switch group correctly routed?
- Is the ERC (External Remote Control) working?

3.4 Testing terminals

System terminals must be installed in their definitive locations. A second person is required to carry out the tests.

3.4.1 Office 45

Checkpoint:

- Full-duplex handsfree operation (with expansion card DSP-02 only)

Additional checkpoint for Office 45pro

- Power supply unit for illuminated display

Additional checkpoints for Office 45 VA

- Queue
- Operator number
- Special subscriber authorizations
  - Attendant menu: Remote access yes / no
  - Substitution authorization
  - Operate switch position
- Call charge display (depends on the network provider)

3.4.2 Other system terminals

Testing special configurations:

- Line Keys
- Team keys
- Function keys
- Key inscription

Testing supplementary equipment

- Alpha keyboard Office AKB for Office 35 and Office 45
- Additional keypad Office (ZTF) for Office 30 and Office 40
- Expansion keypad Office (EKP) for Office 35 and Office 45

3.4.3 Pocket Adapter (PA)

- PC connected to COM1
- Terminal connected
- DIP switch settings
- TAPI driver (PC Dial) loaded
- PA Version ≥ V2.4

See also "V.24 interface on the Pocket Adapter (PA)", page 790.

3.4.4 PC Operator Console Office 1550

The PC Operator application is very simple to install and test. See PC Operator documentation.

3.4.5 Ascotel DECT

Testing the radio units

All the radio units must be installed and activated via the configuration. At least one handset must be logged on the system.

- Is the DSP card fitted?
- Does the radio unit start up correctly?
  Do the LED flash sequences during the start-up process correspond to Tab. 7.11?
- Is the radio unit active (LED check)?
  Does the LED flash green at intervals of 1/8 s On, 7/8 s Off (all B channels free)?
- Is the field strength sufficient for the handsets?
  - Set up an internal connection with the handset:
    During the connection setup the LED first lights up orange, then green at a rate of ½ second on, ½ second off.
  - Long-click key 4;
    Check the bar indicator for field strength:
    4 bars = "very good", 3 bars = "good", 2 bars = "adequate",
    1 bar = "poor", no bar = "no signal".
    (If the bar indicator does not appear after long-click 4, the display can be changed with further long-clicks of key 4.)
  - Clear down the connection.

Testing the handsets

- Does the Hotkey work?
- Does Twin Mode and / or Twin Comfort work?
- Does the Take function work?

3.5 Function test of the Ascotel Voice Mail System

Checkpoints:

- Auto Attendant
- Announcement texts
- Notification on the terminal

3.6 Checking the logging of call data and call charges

The logging of call data (OCL / ICL) and call charges (ICC) is enabled only when a terminal is connected.

Checkpoints:

- Are the call charges displayed on the terminal?
  This feature depends on the network provider.
- Is the output format (PC5, 4, 3, etc.) correct?
  (see "Output formats", page 385).

3.7 Checking the data service

3.7.1 Data transmission on the B channel

Test procedure

- Set up connections to various data services via the ISDN network using known data service numbers.
- Test services that allow both outgoing and incoming communication (e.g. fax, TA) in incoming mode also (with and without direct dialling).
- Check the configuration of the data service table.

3.7.2 Data transmission X.25 in the D channel

Test procedure

- If the service provides for X.25 transmission via the D channel, set up the appropriate test connections via PC and the connected TA X.25-D adapter to the X.25 network via ISDN (for information on operation and connection to the S bus, see "X.25 connection in the D channel", page 757).
- Test the X.25 service during simultaneous phone traffic on the allocated B channels.

3.8 Testing the PISN

There are several possibilities for networking a PISN:

- Leased-line networking
- Virtual networking
- Virtual networking and virtual Centrex
- IP networking

The tests for the first three types of networking are described further below. For information on testing Networking over IP (Internet Protocol) see the System Manual AIP 6400.

3.8.1 PISN with fixed networking

The following test lists relate to a specimen network as shown in Fig. 6.1.

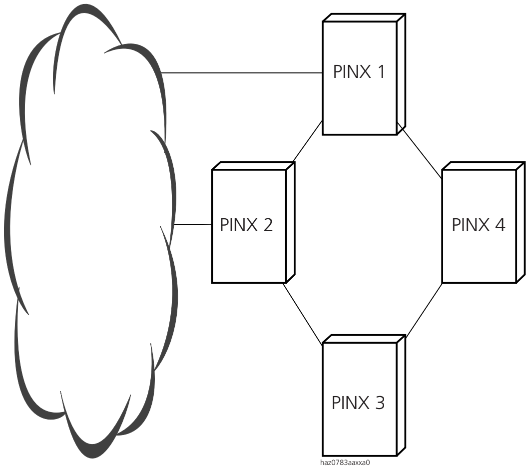


Fig. 6.1: PISN with leased-line networking

1. In PINX 1, only allocate to the route pointing to PINX 2 the trunk group containing the lines to be tested (e.g. trunk group 5).
2. Move all the network interfaces in trunk group 5 - bar one - to another, empty trunk group (in this example trunk group 20).
3. Call a subscriber on PINX 2.
4. Move the tested network interface to trunk group 20.
5. Move the next network interface to be tested from trunk group 20 to trunk group 5.
6. Repeat steps 3 to 5 until all the lines have been tested.
7. Move all the network interfaces back into trunk group 5.
8. Repeat the procedure for all trunk groups with lines to PINX 2.
9. Allocate all the trunk groups back to the route.

Repeat steps 1 to 9 for all PINXs.

Function test for PISN – internal traffic

1. Set up at least one connection to each PINX.
2. Check the ringing pattern.
3. Check the display on the terminal.

Test the overflow routing from PINX 1 to PINX 2 via PINX 3 and PINX 4

1. In PINX 1, assign to the route pointing to PINX 2 (route 5 in the example here) only the trunk group with the lines to be tested (trunk group 6 in the example here).
2. Call a subscriber on PINX 2.
3. Allocate all the trunk groups back to the route.
4. Repeat steps 1 to 3 with all the PINXs.

Testing outgoing traffic to the public network

1. Set up an external connection from a PINX using an exchange access prefix (0 in this example).
2. Check the display on the terminal.
3. Test the overflow routing to the public network, proceed as described above.

Testing incoming traffic from the public network

1. Set up a connection from the public network to subscribers in the PISN.
2. Check the display on the terminal of the PISN subscribers.
3. Repeat the test with numbers which are meant to ring with a delay on the Operator Console OC.

3.8.2 PISN with virtual networking

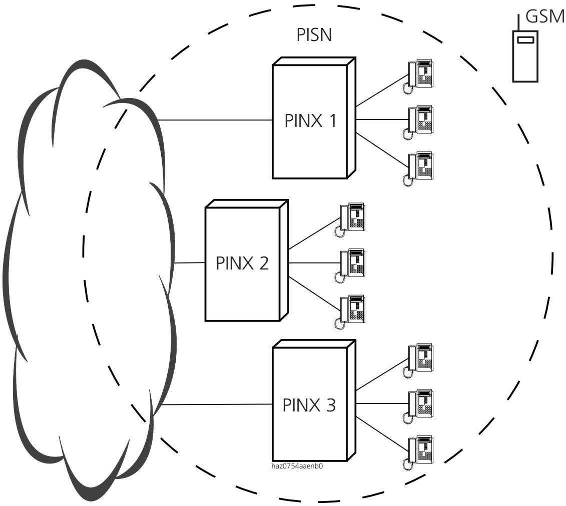


Fig. 6.2: PISN with virtual networking

Testing the connections in the PISN with virtual networking

1. Dial an internal number in the PISN.
2. Activate automatic callback.
3. Test the incoming connection of a virtual subscriber.

Testing into the public network (break-out)

1. Connect all the lines to the public network from various PINXs in the PISN and enable them via the configuration.
2. Dial a public subscriber who is geographically close to a PINX.
3. Set up the connection.
4. Check whether the connection was actually set up from the PINX closest to the public destination.
5. Dial a public subscriber who is geographically close to a different PINX.
6. Set up the connection.

7. Check whether the connection was actually set up from the PINX closest to the public destination.

See also:
Information on break-out see page 100 and page 348.

Testing the substitution

1. Dial the destination subscriber in the PISN who activated the substitution.
2. Set up the connection.

Testing the Call Forwarding Unconditional (CFU)

1. Dial the destination subscriber in the PISN who activated call forwarding (internal or external).
2. Set up the connection.

3.8.3 PISN with virtual networking and virtual PBX (Centrex)

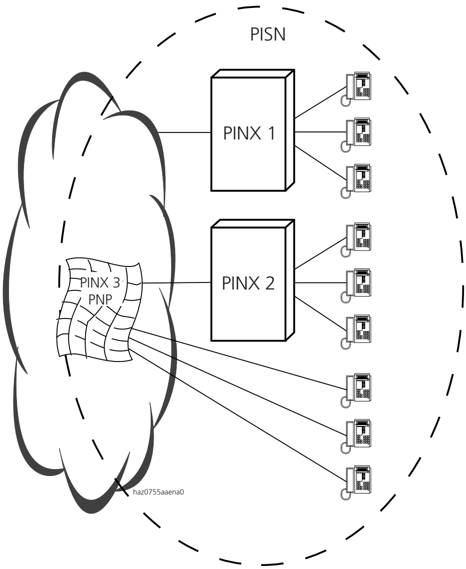


Fig. 6.3: PISN with virtual networking and virtual PBX (Centrex)

Testing the connections in the PISN with virtual networking and virtual PBX

1. Dial into the public network:
   - Does exchange access with "0 - " work?
   - Is the connection number correctly displayed, preceded by an exchange access and a hyphen?
2. Dialling into the Centrex network:
   - Is it possible to set up connections with internal numbers incl. alpha-dialling?
   - Is the connection number correctly displayed, preceded by an exchange access and a hyphen (0 - number)?
3. Incoming calls from the Centrex network:
   - Are calls signalled correctly as "internal calls"?
   - Is the display effected with CLIP internal or with name, if a name is stored?
4. Incoming calls from the public network:
   - Are calls signalled correctly as "external calls"?
   - Are the calls signalled correctly with CLIP external, i.e. preceded by the network access digit ("0 - ")?

4 Customer induction

Before the PBX is formally handed over, the customer is given an induction course on the system's main features. It includes an overview of the system's performance with practical demonstrations as well as an introduction into how to operate the terminals involved. Particular emphasis is to be placed on the features that are important to the customer and also those that are new to him.

The scope and type of customer training depends essentially on

- customer experience with PBX systems
- User requirements (Attendant, System Manager)
- Complexity of the system and its supplementary equipment (with reference for example to the system limits, handover thresholds, radio gaps with cordless systems)

The basic customer training includes, among others,

- Using the switching functions
- Using the feature functions
- Using the hotel features
- Using functions in the PISN

The customer must also be instructed on how to proceed in the event of faults, event and error messages; on the possibilities available with remote maintenance; and on how access authorization to the configuration / remote maintenance is regulated.

5 Handover to the customer

The following checklists contains the main points to be taken into account when the system is handed over to the customer.

Documentation

- Are all the terminals clearly and intelligibly labelled?
- Does each terminal have its own operating instructions?
- Are additional operating instructions required (e.g. operating instructions for Office system terminals, operating instructions for the System Assistant, etc.)?
- In what form should the System Configuration be archived:
  - in paper form next to the PBX?
  - or as software stored on the PC or on diskette?
- Are the passwords stored in a suitable place?
- Is the distribution list filled out (stored with the distribution frame)?

Customer-specific agreements

- Are all customer-specific applications fully documented?

Phone book setup / takeover

- Is the internal phone book set up on the system in accordance with the customer’s wishes?

Other checkpoints

- Has the customer received the brochures and price lists on:
  - Expansion / extension possibilities (to increase performance)
  - Additional terminals (add-on products; to improve user convenience)

Part 7 Operation and Maintenance

1 Overview of Chapters

Data Maintenance

Chapter 2 describes how to save, update, load and service data. First it explains which memories are used to store the system software and configuration data and where those memories are located on the components.

The next topic is how to update the system software on the PBX and on the terminals. The handling of configuration data (Upload / Download, Backup/Restore) is menu-driven using AIMS.

The final section deals with backing up and deleting system configuration data and terminal configuration data. With terminals it is possible to use AIMS to delete the configuration data for all the terminals simultaneously or for just one specific terminal.

Hardware updates and system expansion

Chapter 3 looks at how to adapt an existing system to customer requirements. It explains which configurations are subject to licensing, the scope covered by licensing in each case, the processes that influence licensing, and how to modify an existing licence.

When replacing expansion cards or Office terminals with terminals of a different type, it is possible under certain circumstances to transfer the configuration data. Information on the procedure for replacing hardware can be found in the second half of this chapter.

Operations supervision

The subject of Chapter 4 is the AIMS System Event Manager (SEM) for recording event messages generated by the system. The event message concept with detailed event messages, which can be specifically configured and analysed, supports both local and remote maintenance. For remote maintenance the Fault & Maintenance Manager is used in addition.

At the end of the chapter you will find numerous error tables, which systematically list occasional faults. They provide optimum support on how to determine the causes of errors, how to deal with errors, and how to target and replace defective components.

2 Data Maintenance

2.1 What data is stored where

The PBX's data storage system consists of 3 system cards:

- The Flash card stores the system software, the boot software and the configuration data.
  The contents of the memory are retained even when there is no power supply.
- The RAM card contains the main memory.
  The main memory stores volatile data that cannot be saved. It is available only when the PBX is in operation.
- The EIM card (Equipment Identification Module) contains the system-specific data (system ID, system type, sales channel, licensing, DECT identification numbers).

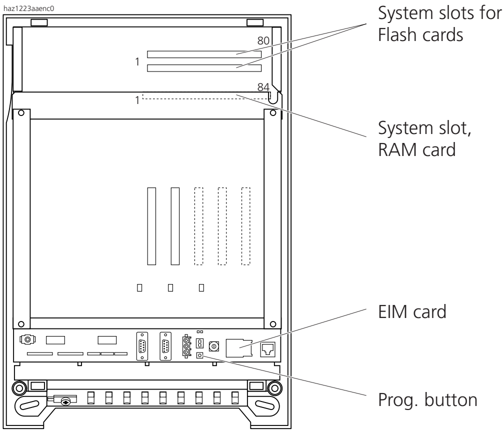


Fig. 7.1: Memories on the mainboard 2025 / 2045

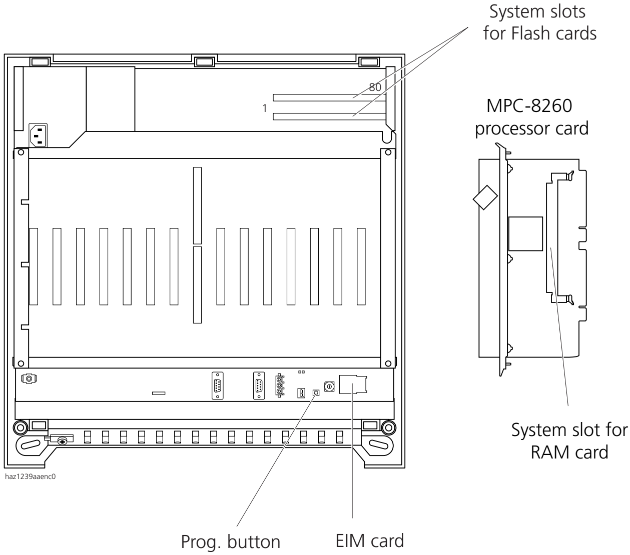


Fig. 7.2: Memories on the mainboard 2065 / MPC-8260 processor card

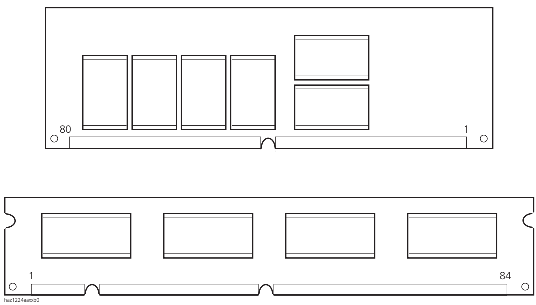


Fig. 7.3: Flash (top) and RAM (bottom) memory cards

The RAM and Flash memory cards are the same for both basic systems.

2.1.1 System software

The PBX's entire system software and the software of a number of system terminals (terminals with software download possibility) are stored in compressed form on the Flash card in a serial Flash memory.

The main memory for program and data is on the SDRAM card. When the PBX starts up, the system software on the serial Flash memory is decompressed and loaded into the main memory.

2.1.2 Boot software

The boot software is stored on the Flash board in a parallel Flash memory.

2.1.3 Configuration data

There are two categories of configuration data:

- System configuration data
- Terminal configuration data

This data is stored in parallel memories on the Flash card.

2.1.3.1 System configuration data

The System Configuration data contains all the settings for the System Configuration with the exception of the terminal-specific settings.

The system configuration data is stored on the Flash card in a parallel Flash memory and saved during AIMS backup.

System configuration data can only be modified using AIMS.

2.1.3.2 Terminal configuration data

The terminal configuration data includes all the terminal-specific settings such as

- private phone books
- key assignments
- terminal settings such as volume, ringing melody, etc.

Terminal configuration data is available only for system terminals of the Office family. It is stored in a parallel Flash memory and saved during the AIMS backup.

Terminal configuration data can be modified either using AIMS or directly from the system terminal.

2.1.4 System-specific data

The system-specific data (system ID, system type, sales channel, licensing, DECT identification numbers) is stored on the EIM card (chip card).

2.2 Updating system software and terminal software

The PBX's system software is loaded using the AIMS Upload Manager (software upload).

The PBX's system software also contains the software for the Office 45 system terminals, the DECT radio units and the Office 135 and Office 155pro handsets.

There are several possibilities for establishing a communication link between the PBX and the AIMS Upload Manager (see "PBX connection", page 891).

The type of access to the PBX is also determined by the type of upload. A standard upload replaces an already functional system software (see "Standard upload", page 892).
An Emergency Upload has to be initiated if the PBX does not have a functional system software (see "Emergency Upload of the system software", page 896).

2.2.1 AIMS Upload Manager

The Upload Manager called up via the AIMS Shell is a convenient and reliable way of loading a new system software on to the PBX.

All the information and settings required for the software upload are accessible via the main menu of the AIMS Upload Manager, namely "PBX Remote Upload".

The top left-hand portion of the screen contains pull-down menus and a toolbar. They contain all the main standard functions.

Context-sensitive help information is available in many of the menus.

The "Configuration" tab

The "Upload" tab contains information, input and selection possibilities for the system type, software version and the PBX connection.

"PBX" Tab

This tab is used to define when the newly loaded system software is to be activated and how long it should be monitored to ensure it is operating without fault.

"Optionen" pull-down menu

The "Options / Settings" menu item is used to configure the communication link.

Monitor

The Upload Monitor is an Upload Manager function and is called up using "Options / Monitor". The Upload Monitor is used to call up detailed information on the PBX (e.g. hardware expansion, event messages, upload configuration). The Upload Monitor is not available while a software upload is running.

2.2.2 PBX connection

The communication link for a software upload is selected in the "PBX connection" field of the "Upload" tab. The possibilities for the PBX are as follows:

- Local access:
  This operating mode is designed for a software upload via the V.24 interface of the mainboard or of a Pocket Adapter via PPP.
- Dial-up access:
  A software upload is possible via an ISDN connection (PPP protocol) in the public network (external dial-up access) or in the private network (internal dial-up access). The call number under which the PBX can be reached via remote maintenance has to be entered.
- Local access Xmodem port:
  This PBX connection type is designed only for an Emergency Upload (no functional system software available on the PBX) (only via COM1).
  With this operating mode the software is loaded via the V.24 interface on the mainboard using the Xmodem protocol.
- LAN:
  This operating mode is designed for a software upload via LAN.

Setting

The connection parameters for a software upload are set using "PBX connection / Setting".

- Connection parameters for local access:
  - IP address of the PBX for dial-up networking connections (redundant if the IP address has already been configured in dial-up networking; see IP configuration, IP setup PPP/dial-up networking server)
  - Dial-up networking; information on dial-up networking (see "Settings on the PC", page 841)
  - Terminal; "Ascotel Direct PPP Connection"
- Connection parameters for dial-up access:
  - IP address of the PBX for dial-up networking connections (redundant if the IP address has already been configured in dial-up networking; see IP configuration, IP setup PPP/dial-up networking server)
  - Dial-up networking; information on dial-up networking (see "Settings on the PC", page 841)
  - Device; ISDN card or TA connection parameter LAN
- Connection parameters for local access Xmodem port:
  - Communication port number of the PC (COM1 initialization value)
  - Transmission rate (initialization value 9 600)
  - Number of data bits (initialization value 8)
  - Party (initialization value None)
  - Stop bits (initialization value 1)
  - Flow control (initialization value RTS / CTS)
- Connection parameters for LAN
  - IP address of the PBX for LAN connections (redundant if the IP address has already been configured in CM under ”LAN IP configuration”)

See also:
For further information on parameter configuration see "Settings", page 840).

2.2.3 Standard upload

Loading a PBX with new system software takes places in several phases. The Upload Manager monitors the system software copying process from the PC to the PBX. The PBX monitors the version transfer.

Sequence phases for the standard upload:

- Preparation phase:
  The Upload Manager prepares the PBX for the transmission of the new system software.
- Upload and backup phase:
  The new system software is transmitted to the PBX in compressed form and stored in the dedicated Flash memory on the Flash card. The previous system software remains stored as a backup.
- Software update phase:
  The current system software is replaced by the freshly loaded system software.
- Restart phase and version transfer:
  The PBX is restarted automatically. The boot software starts the new system software and runs a version transfer at the same time.

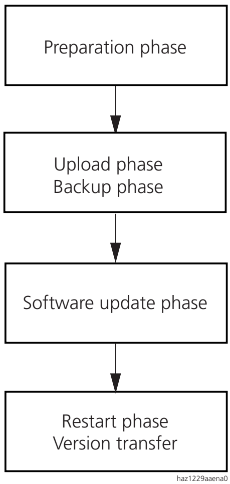


Fig. 7.4: Software upload sequence

During a standard upload the PBX remains operational during the upload phase. After a successful software upload, the PBX runs an automatic warm-start at the set time and restarts with the newly loaded system software.

If for whatever reasons the software upload is not possible or if a fault occurs during the upload, the previous system software with the previous configuration data remains activated.

If errors are detected in the system software during the time in which the newly loaded system software is being monitored for fault-free operation, the previous system software is reactivated.

If the previous software is no longer available, an Emergency Upload has to be initiated using the AIMS Upload Manager. An triggered Emergency Upload is signalled by "EUL" on the 7-segment display (see "Emergency Upload of the system software", page 896). The PBX is not operational during that time.

Successful and failed software uploads are stored as event messages in the PBX and output at the set signal destinations.

See also:
For information on event messages and signal destinations, see "Event message concept", page 913.

Initiating an upload process

To ensure a successful software upload, carry out the following preparatory steps:

1. Recommendation: Use AIMS to save the configuration data (see "Saving configuration data with AIMS", page 900).
2. Call up the AIMS Upload Manager.
3. Select the "Configuration" tab.
4. Enter the password for access to the PBX.
5. Select the system type.
6. Click the "Add" button in the "Software Version" field and specify the directory containing the zip-file with the system software you want to install.
   The software version you have added appears in the list box.
7. Select the software version you want from the list box.
8. Select the PBX connection (see "PBX connection", page 891):
   – Local access (PC connection to the V.24 interface of the mainboard or PA)
   – Via dial-up access (ISDN, PC connection on the S/T interface)
   – Via LAN (PC connection to the Ethernet interface on the basic systems)
9. If required, use the "Settings" button to set the connection parameters (see "Setting", page 892).
10. Select the "PBX" tab.

11. Set the time at which you want the newly loaded system software to be activated.
12. Set the period of time during which you want the newly loaded system software to be monitored for fault-free operation.
13. Click the "Upload" button.
    The upload process is now initiated.

Bar indicator

During the software upload a dialog box with a horizontal bar indicates the time progress of the upload process.

If you need to stop the upload at any stage, click the "Cancel" button. The software upload is then stopped and the current system software remains in operation.

Status display

The status display provides information with date and time indications on the current software upload, including all the event messages output in connection with the current software upload.

Upload log

Once an upload process is completed or if it is terminated prematurely, the settings of the software upload including the data automatically entered in the log directory are printed out on the system printer. The data is displayed in the following sequence:

Tab. 7.1: Software upload

dd/mm/yy, hh:mm:sec	
Remarks	
Software version	
PBX type	
Connection type	
Event messages	

2.2.4 Emergency Upload of the system software

An Emergency Upload has to be activated whenever a standard software upload is not possible or has proved faulty.

To ensure a successful Emergency Upload, proceed as follows:

1. Call up the AIMS Upload Manager.
2. Select the "Configuration" tab.
3. Enter the password for access to the PBX.
4. Select the system type.
5. Click the "Add" button in the "Software Version" field and specify the directory containing the zip-file with the system software you want to install.

The software version you have added appears in the list box.

6. Select the software version you want from the list box.
7. In the "PBX connection" field select "Local access Xmodem port" (see "PBX connection", page 891).
8. Use the "Settings" button to set the connection parameters (see "Setting", page 892).
9. On the "PBX" tab click the "Upload" button.

If a connection was successfully set up with the PBX,a prompt will appear to ask whether the loaded system software is to be activated with or without PBX initialization after the software upload.

If the newly loaded system software is activated without initializing the PBX, any existing configuration data will be retained. If the newly loaded system software is activated with a PBX initialization, all existing configuration data will be deleted or reset to the initialization values.

10. Select the process you want.

The Emergency Upload is started.

The Emergency Upload sequence can be monitored using the 7-segment display.

Tab. 7.2: Sequence of the 7-segment display during an Emergency Upload

7-segment display	Meaning
EUL	Once Emergency Upload is initiated, the letters EUL appear one after the other.
P flashing	Software waiting for the program to be transferred from the V.24 interface via Xmodem.
Segments rising slowly	Receiving the system software via V.24 Xmodem.
Segments rising quickly	Writing the system software from the RAM into the Flash memory.
S	A rotating "8" appears while the system software is being decompressed and loaded into the RAM memory on the RAM card.
G	The upload or copy was successful (‘GOOD’).
E-XXX-	An error has occurred (XXX stands for error code, see "Error coding", page 944).

2.2.5 Software upload for hard-wired system terminals

The software for the system terminals is contained in the PBX's system software and is therefore always updated along with the PBX's system software.

With the exception of the Office 45 system terminal, hard-wired system terminals of the Office family do not have their own memory.

Office 45 system terminal

The Office 45 system terminal contains a Flash memory. The Flash memory contains an area that cannot be modified. This boot area is used for starting the terminal and receiving the software.

The software for the Office 45 is contained in the PBX's system software. The loaded software is tested when the terminal starts up. If the loaded software is not identical to the version in the system software, the software will be downloaded from the PBX on to the terminal.

The terminal's software is stored in the Flash memory.

2.2.6 Software upload for DECT systems

Ascotel DECT radio unit

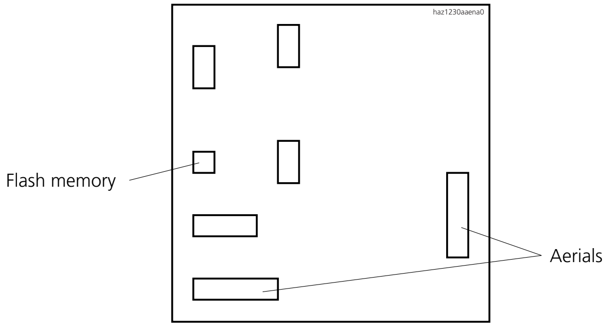


Fig. 7.5: Ascotel DECT radio unit

The Flash memory on the Ascotel DECT radio unit contains an area that cannot be modified. It is used for starting the radio unit and receiving the software for the radio unit.

The actual software for the radio unit is contained in the PBX's system software. The loaded software is tested when the radio unit starts up. If the loaded software is not identical to the version in the system software, the software will be downloaded from the PBX on to the Ascotel DECT radio unit and stored in the Flash memory of the Ascotel DECT radio unit.

Office 135/Office 155pro handsets

The software for the Office 135/Office 155pro handsets is updated via radio (AIR upload). This requires the handset to be logged on to system A.

The memory in the handsets is a Flash memory. The Flash memory contains an area that cannot be modified. This area contains the handset's boot software.

The handset software is contained in the PBX's system software. The loaded software is tested when the handset starts up. If the loaded software is not identical to the version in the system software, the PBX will initiate an AIR update. The software is loaded from the PBX onto the handsets via radio and stored in the Flash memory.

To be able to run an AIR upload, you need to ensure that the handset contains a functional software.

The handset remains fully functional while an AIR upload is in progress. The new loaded software is activated only once the upload has been successfully completed. A reset is carried out on the handset.

If the handset does not have a functional software, the software has to be loaded via the Iris box using the Office 135 adapter in the case of the Office 135, or via the Doris adapter in the case of the Office 155pro .

Office 100 and Office 150 handsets

The software for the Office 100 and Office 150 handsets is stored in the Flash memories.

- The software for the Office 100 handset is updated via the Iris adapter.
- The software for the Office 150 handset is updated via the Doris adapter.

Software upload via radio (AIR upload) is not possible on the Office 100 and Office 150 handsets.

2.3 Updating the PBX boot software

The boot software is stored on the Flash board in a parallel Flash memory (see Fig. 7.3). To update the boot software, you need to replace the Flash card:

Procedure:

1. Preliminaries: Use AIMS to save the configuration data (see "Exchanging data between PBX and PC", page 153).
2. Replacing the Flash card: Carry out steps 2-11 as described in "Replacing the Flash Card", page 961.
3. Use the Upload Manager to load the system software (see "Updating system software and terminal software", page 890).
4. Use AIMS to reload the configuration data (see "Exchanging data between PBX and PC", page 153).

2.4 Updating configuration data

System configuration data can only be modified using AIMS (see also "System configuration data", page 889).

Terminal configuration data can be modified using both AIMS and the system terminal (see also "Terminal configuration data", page 889).

2.4.1 Saving configuration data with AIMS

The procedure for saving configuration data with AIMS
(Upload / Download, Backup / Restore) is described in "Exchanging data between PBX and PC", page 153.

2.4.2 Deleting system and terminal configuration data

Deleting system configuration data

To delete the system configuration data from a system terminal, delete the terminal's subscriber number from the numbering plan in the system configuration (Configuration Manager) using the Delete key. All the data is then reset to its initialization values.

Terminal configuration data of all system terminals

To delete the terminal configuration data of all the Office system terminals, use the "Delete Office data" configuration menu.

A warning message appears before deleting begins and you can cancel the delete process without any change to the data. The system configuration data of all the terminals is retained.

Terminal configuration data of one system terminal

To delete the terminal configuration data of an individual Office system terminal, delete the terminal's subscriber number from the numbering plan in the system configuration (Configuration Manager) using the Delete key, then re-enter a new number. All the data is then reset to the initialization values.

3 Hardware updates and system expansion

Here is described how licenses are adapted, how cards and terminals are replaced, expanded and removed, and how a defective mainboard is replaced.

3.1 Licenses and EIM cards

License information and the IP address of the PBX are stored on the EIM card. This chapter describes how licenses are adapted as well as when and how an EIM card is replaced.

3.1.1 Licences

This available licenses are described in Part 3 under "System expansion". The following chapter describes how licenses are adapted.

3.1.1.1 Adapting licences

To expand a system already in operation or to re-order a licence for a new system (see "Licence-related System and Expansion Limits", page 593), proceed as follows:

1. Order the licence package you want from your authorized dealer. Also Specify the system's licence information (EID, old licence code, system type). Your dealer will then send you a new licence code.
2. Overwrite the old licence code with the new code (see "Activating the licences", page 854). The new licence code is stored in the EIM.
3. Upload the system configuration data and restart the system; the licensed function(s) is (are) now enabled.

3.1.2 EIM card

If the EIM card has to be replaced if:

- A licence is transferred to another system of the same type
- The mainboard is defective
- The EIM card is defective

A licence is transferred to another system of the same type

A licence can only be transferred to a system of the same type. To do so you need to replace the EIM card with the licence information.

The mainboard is defective

If you need to replace a defective mainboard, transfer the EIM card from the defective mainboard onto the new one. For instructions on how to replace the mainboard, see "Replacing the 2025 / 2045 or 2065 mainboard", page 910.

The EIM card is defective

In the unlikely event of a defective EIM card, contact your authorized dealer to discuss the procedure.

3.1.2.1 Replace EIM card

The EIM card is located in a chip-card housing with a sliding lock that is secured directly on the mainboard (see Fig. 7.1 and Fig. 7.2).

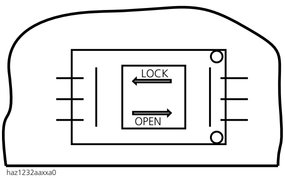


Fig. 7.6: EIM card

Note:
The EIM card can also be replaced when the PBX is in operation. Current calls and data connections are interrupted.

Warning:
EIM cards can be damaged by electrostatic discharge (ESD).
Make sure you observe the measures to protect against electrostatic discharge.

Remove EIM card

1. Remove the PBX housing cover.
2. Touch the earthed housing (equipotential bonding).
3. Open the chip-card housing (slide the lock toward ”Open”).
4. Remove EIM card.

Fit the EIM card

1. Fit a new EIM card.
2. Close the chip-card housing (slide the lock toward ”LOCK”).
3. Press the Prog. pushbutton (see Fig. 7.1 and Fig. 7.2).
4. Mounting the PBX housing cover.
5. After the system has started, log on all DECT handsets again. This is necessary because the DECT identification numbers are stored on the EIM card.

Note:
The EIM card must be fitted before the system is put into operation. The PBX will not start without the EIM card.

3.2 Expansion cards

Basic systems are expanded by equipping the expansion slots with expansion cards. This chapter describes how and when expansion cards are expanded, moved and replaced.

The combination of card type and the the maxium expansion is determined by system limits. The system and expansion limits of the various basic systems are described in "Licence-related System and Expansion Limits", page 593.

Expansion card types

There are three types of expansion cards: Subscriber cards, trunk cards and special cards. An overview of the different cards with their interfaces can be found in "Equipping the PBX", page 733.

Expansion card slots

The 2025 / 2045 mainboard contains 5 expansion slots for expansion cards. The 2065 mainboard contains 14 expansion slots (7 expansion slots on either side of the processor card).

Expansion cards can be fitted to any expansion slot. Exception: On the Ascotel 2065 the system slot in the middle is reserved for the MPC-8260 processor card.

Unused expansion card slots are fitted with dummy covers (EMC).

Expansion card configuration data

Configuration data consists of system configuration data and terminal configuration data. All the configuration data is stored centrally in a non-volatile memory on the Flash card. This means that configuration data is preserved whenever a defective subscriber card or trunk card has to be replaced by a new one.

3.2.1 Installing and removing cards

3.2.1.1 Preparations

First step before expansion cards or system cards are removed or added:

- Inform users
- Prebar the system
- Disconnect system power supply

Prebar the system

Prebarring the system prevents setting up new connections. Ongoing calls are not cleared down. If a user tries to set up a call while prebarring is activated, he will obtain the congestion tone and the system terminal display will read "Invalid number".
A prebarred system is indicated by the 7-segment display on the mainboard.

Tab. 7.3: Displays on the 7-segment display

Horizontal bar display	Meaning
the bottom line	System prebarred
the middle line	Active internal connections
the top line	Active external connections

The system is prebarred in the AIMS Configuration Manager under "Card configuration": press the "Prebar system configuration" button.

As soon as there are no more active connections in the prebarred system, the system can be taken out of operation.

Note:
Prebarring the system can be dispensed with if all concerned are aware that existing connections will be disconnected.

Switch off the PBX

When using a UPS disconnect the battery first, then disconnect the PBX and any external Auxiliary Terminal Power Supply (ATPS) from the 230 V mains.

Hazard:
Expansion cards can be damaged by electrical voltage.
Always disconnect the power supply before fitting or removing any cards!

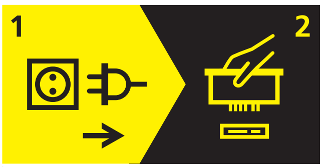


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Fig. 7.7: Warning plate on the PBX

3.2.1.2 Replacing a defective card

Required tools:

- Torx screwdriver, size Torx T 10

Instructions for changing the card:

1. Inform users, prebar the system and disconnect the system from the power supply (see "Preparations", page 904).

Hazard:
Expansion cards can be damaged by electrical voltage.
Always disconnect the power supply before fitting or removing any cards!

2. Remove the PBX cover.

Warning:
Expansion cards can be damaged by electrostatic discharge (ESD).
Always touch the earthed metal cage of the PBX before touching a card!

3. Unscrew the screws on the screening strip and pull out the expansion card complete with screening strip.
4. Wrap the dismantled expansion card in the ESD protective sleeve.
5. Take the new expansion card out of its ESD protective sleeve.
6. Fit the expansion card to the correct slot. Make sure the card is lying in the lower guide slot of the metal cage and that the ends of the screening strip are flush against the edge of the metal cage once the card has been inserted.
7. Screw the screening strip to the metal cage using the 2 screws supplied.
8. Fit the PBX cover.
9. Connect the PBX and any ATPS system to the 230 V mains, then connect the UPS battery where applicable.

Note:
Always fit dummy covers to unused expansion slots to ensure EMC.

See also:
A number of rules have to be observed when fitting the cards (see "Component mounting rules", page 588).

3.2.1.3 New card with fewer ports

A card is replaced by a similar card with fewer ports.

Procedure:

1. Change the card and put the PBX into operation again (see "Replacing a defective card", page 906).

The following data is deleted:

- The system and terminal configuration data of the terminals on the user-network interfaces that are no longer present in the new configuration.
- The system configuration data of the network interfaces that are no longer present in the new configuration.

Tab. 7.4: Example: Reducing the number of user-network interfaces

SC-16AD2 → SC-08AD2	The configuration data of user-network interfaces 9...16 is deleted.
SC-08AD2 → ISDN-04ST	The configuration data of all 8 user-network interfaces is deleted.

Note:
If the terminal configuration data of system terminals is deleted following the reconfiguration of an expansion card, a warning message will appear beforehand to give you the possibility of cancelling the process. However, this is possible only if the configuration data of the original card was not already deleted beforehand.

3.2.1.4 New card with more ports

A card is replaced by a similar card with more ports.

Procedure:

1. Change the card and put the PBX into operation again (see "Replacing a defective card", page 906).
2. Select “Accept System Configuration” in the AIMS Configuration Manager.
3. Configure new ports.

The system configuration data (SC No., SC configuration, etc.) of the terminals on the new ports is created as new data (initialization values).

Tab. 7.5: Example: Expanding the number of user-network or network interfaces

SC-08AD2 → SC-16AD2	The configuration data of user-network interfaces 9...16 is created as new data.
ISDN-04ST → SC-08AD2	The configuration data of all 8 user-network interfaces is created as new data.

3.2.1.5 Change slot

Expansion cards can be moved to a different slot. The terminal configuration data of the system terminals can be transferred.

Procedure:

1. Change the slot and put the PBX back into operation (similar procedure as described in "Replacing a defective card", page 906.
2. Connect the system terminals to the ports of the new slot.
3. Re-configure port allocation using the AIMS Configuration Manager.
4. Log on the card at the new slot and log it off at the old slot. The configuration data at the old slot location is now deleted.

3.2.2 Special cards

In principle, special cards are changed in the same way as expansion cards. Proceed as follows:

- When fitting a special card to a different slot, it has to be logged off from the old slot location with the AIMS Configuration Manager and then logged on at the new slot location.

- Voice Mail cards VM-02 / VM-04:
  The configuration and voice data is stored on the Voice Mail card and needs to be saved separately before the card is changed. For more information see the System Manual for the Voice Mail System AVS 5150.

- IP interface cards AIP-6400 and AIP-6350:
  The configuration data is stored on a Flash component on the IP interface card and can be saved in a file. For more information, refer to the System Manual AIP 6400 and System Manual AIP 6350.

- Signal processor cards DSP-01 / DSP-02 / DSP-04:
  These special cards do not contain any data that has to be saved prior to a card change.

- Door intercom card OI-2DOOR:
  This special card does not contain any data that has to be saved prior to a card change.

3.3 System cards

System cards are necessary for operating the PBX. There are different memory cards and the processor card for the 2065 basic system. This chapter describes when and whether a system card should be replaced.

3.3.1 Memory cards

Flash card

The Flash card is replaced together with the mainboard and therefore does not have to be removed separately. Exceptions:

- Updating the PBX boot software (see page 899).
- A system defect (e.g. mainboard) can make it impossible to read out unstored configuration data with AIMS. In such cases the data can be saved using a new mainboard.
- Upgrading system 2065 from I5 to I6 (see page 959)

RAM card

The RAM card is replaced together with the mainboard and therefore does not have to be removed separately (Exception: Upgrading system 2065 from I5 to I6, see page 959).

EIM card

For information on how to replace the EIM card (see page 901)

3.3.2 Processor card MPC-8260

The MPC-8260 processor card for the 2065 basic system is factory-fitted in a special system slot and cannot be moved to a different slot. The processor card is replaced together with the mainboard and therefore does not have to be removed separately.

On the 2025 / 2045 basic system the processor is integrated into the mainboard.

3.4 Mainboard and power supply modules

This chapter describes when and how mainboards and power supply modules are replaced.

3.4.1 Replacing the 2025 / 2045 or 2065 mainboard

If the mainboard or one of the system cards is defective or permanently faulty, the entire basic system will have to be replaced, complete with system cards and metal housing.

1. If still possible, save the configuration data using AIMS.
2. Switch off the PBX. If using a UPS, first disconnect the batteries and then pull out the mains plug.
3. Remove the PBX cover.
4. Touch the earthed housing of the PBX (equipotential bonding).
5. Remove the expansion cards and UPS module. For details of the procedure for removing expansion cards, see "Replacing a defective card", page 906.
6. Replace the EIM card of the defective mainboard and the new mainboard. See "EIM card", page 901 for the procedure.
7. Dismantle all the connected cables in such a way that the new basic system can be identically reconnected.
8. The mainboard is not dismantled but replaced complete with metal housing. Depending on the installation (wall-mounting / cabinet mounting) the housing with the mainboard may also have to be dismantled and the housing cover screwed back on.
9. The new basic system can now be reassembled and fitted in the reverse sequence.

10. If the configuration data was not transferred when changing the Flash card, initialize the system and reload the configuration data using AIMS.

3.4.2 Uninterruptible power supply

The UPS-12V and DC-48V modules are connected with the mainboard via two 6-pin connectors. Information on installation and removal can be found in "Uninterruptible power supply (UPS)", page 725.

3.5 Replacing system terminals

This chapter describes what to pay attention to when replacing terminals.

3.5.1 Corded system terminals

3.5.1.1 Terminals with the same level of added features

Replacing a defective terminal

Once the defective system terminal has been replaced by an identical terminal the terminal configuration data is automatically transferred.

Relocating a terminal

The terminal configuration data of a system terminal can be copied to another terminal with the same level of added features using the AIMS Configuration Manager. The data can also be saved with AIMS and then reloaded if the system terminal is logged on to a different card.

See also:
The procedure for saving configuration data with AIMS is described in "Exchanging data between PBX and PC", page 153.

3.5.1.2 Terminals with a different level of added features

Given that every level of added features on system terminals has a certain number of features, the features are adapted (reduced or increased) to the new terminal whenever the terminal definition is changed. The features are reduced if a terminal is replaced by a terminal with a lower level of added features (e.g. Office 45 → Office 10 to Office 35) or by a predecessor model (e.g. Office 45 → Office 40).

If a system terminal is replaced by a system terminal of a different level of added features, the terminal display will display the message "Incorrect set type". On the Office 10 the LED flashes slowly. In this situation, although the terminal can be used for basic telephone operations, none of the added features will be available.

Before the added features of the new system terminal can be used, the new terminal type will have to be entered in the PBX using the AIMS Configuration Manager "Terminal data" or by configuring at the terminal.

3.5.2 DECT terminals

Replacing a radio unit

1. Dismantle the defective radio unit.
2. Fit the new radio unit.

For information on installing radio units, see "Installing the radio units", page 829.

Replacing a handset

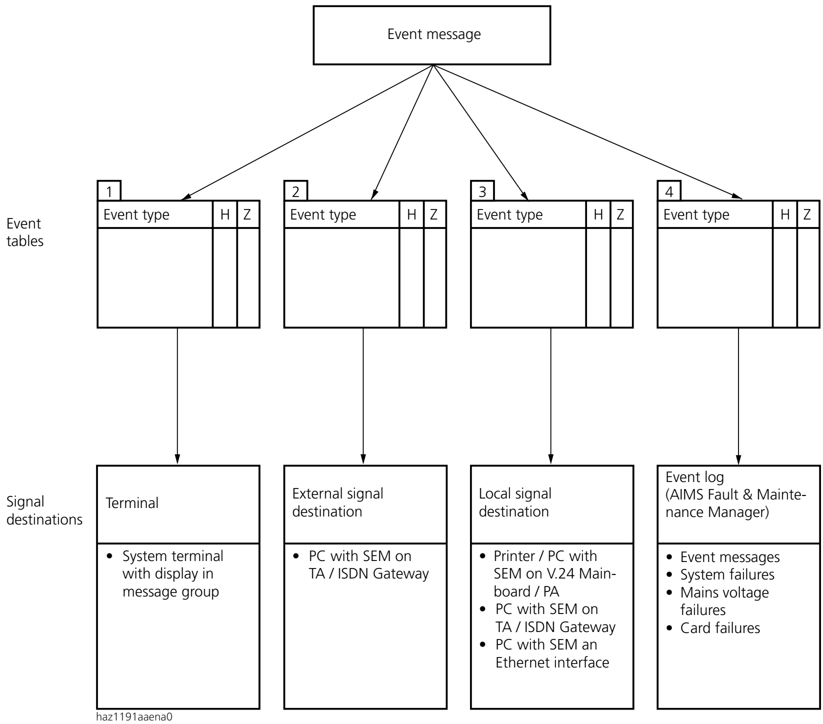
1. Log off the previous DECT subscriber via AIMS.
2. Log on the new handset (see "Configuration Steps", page 853). The handset data is preserved until the subscriber number is also deleted.

4 Operations supervision

4.1 Event message concept

The PBX generates an event message every time an event or error occurs. The event tables in the Fault & Maintenance Manager are used to specify how often an event message of a particular type may be generated over a given time period before the event message is sent to the allocated signal destinations.

The Fault & Maintenance Manager has 4 event tables that can be allocated to 4 signal destinations:



F = Frequency
T = Time period

Fig. 7.8: Distribution principle for an event message

4.1.1 Event types

Tab. 7.6: Event types, in alphabetical order

Event / error message	Trigger condition	Details
ACD server out of service	ACD server defined as destination but not responding.	Date, time
Active radio unit defective	Radio unit not responding	Card number, port number, date, time
Card out of service	A card previously in operation has stopped functioning.	Number of the expansion slot, date, time
CL Printer Jam	• No response from system printer for past 4 minutes • Printer out of paper or switched off	V.24 interface, interfaces/card number, port number, date, time
DECT channels on DSP-0x overloaded	DECT channels on DSP-0x overloaded	Date, time
ESME unobtainable	The LAN connection between the SMSC and the ESME is interrupted	Date, time
External signal destination missing	External signal destination not automatically reachable	0: Busy / 1: Not available / 2: Barred / 3: Undefined, date, time
External SMS gateway unobtainable	External SMS gateway unobtainable by network provider or incorrectly configured	Date, time
ICC overflow	Individual cumulative counter or cost centre counter overflow	Subscriber number, cost centre, exchange line, room number, date, time
Internal signal destination missing	Local output blocked or not available	0: Busy / 1: Not available / 2: Barred / 3: Undefined, date, time
LCR on alternative network provider	Automatic switch from primary network provider to secondary network provider using LCR function	Provider ID, date, time
Local network supply error on the radio unit	Local power supply of an SB-8 radio unit failed or unavailable	Card number, port number, date, time
Mains voltage failures	Event message once mains power is restored • Mains power has failed more frequently than entered in the trigger table	Date, time
Malfunction	With 3-digit error ID Hardware error during self-test (for more details see **Tab. 7.25)** With 5-digit error ID: General error during operation (for more details see **Tab. 7.7)**	Error ID, date, time
No response from network	No answer to call setup on T / T2 interface	Port number of the exchange line circuit, date, time

Event / error message	Trigger condition	Details
Numbers missing	Card(s) not full logged on • Insufficient memory reserved in the numbering plan to enable allocation of numbers to all subscribers: Type in missing numbers by hand	Date, time
Outgoing call rejected	Call rejected by the network • On any line: error code 34 • On required line group: error code 44	Port number of the exchange line circuit, cause, date, time
PBX power supply: Battery	• Warning when the UPS takes over the power supply to the PBX. (Battery parameter) • All-clear once the 230 V mains power supply is restored. (Parameter: Mains power)	0: Battery / 1: Mains power, date, time
Port out of service	A port previously in operation has stopped functioning.	Number of the expansion slot, relevant port number, date, time
QSIG licence limit reached	Maximum number of licensed outgoing connections with QSIG protocol exceeded	Route number, subscriber number, date, time
Recording error	• Card not fitted • Card not logged on • Card defective	Card number, date, time
Remote maintenance is activated	Remote maintenance has been activated	Unfiltered output to local destinations
Reset card	A reset was carried out for one card	Number of the expansion slot, date, time
Software upload	During an upload in PBX status: • "Updating…" • "Monitoring…" • "Normal operation"	• "New PBX software loaded, starting..." • "New PBX software crashed, rollback executed" • "New PBX software started, running fault-free" Date, time
Subscriber does not answer	No answer to incoming DDI call from digital subscriber on S bus or AD2	DDI No., date, time
Synchronisation loss to exchange	A T / T2 interface entered in the clock pool has lost the system clock.	Port number, date, time
System overload	Network access attempted when all lines are seized or the PBX is overloaded.	Route number, subscriber number, date, time
Terminal power supply: Overload	Rated power slightly exceeded for > 4 seconds	Date, time
Terminal power supply: Switch Off	Rated power clearly exceeded for > 4 seconds	Date, time
The System has crashed	The System has crashed	Date, time

Event / error message	Trigger condition	Details
Too many event messages	Number of message types exceeds limit entered in the table on: • "Sync. Loss on T / T2" • "Outgoing Call Rejected" • "No response from network"	Date, time
Too many network interfaces	System limits exceeded	Card No., date, time
Too much subscriber data	System limits exceeded	Date, time
Total Synchronization loss	Synchronization with network has failed on all T / T2 interfaces	Date, time
V.24 of theVoice Mail System defective	Invalid characters received on V.24 interface for Voice Mail Server	Date, time
Voice Mail System defective	Voice Mail Server does not respond to messages via V.24	Date, time
Wake-up call unanswered	Room wake-up call not answered	Room No., date, time

Malfunction with 5-digit error ID

Malfunction-related event messages, with a five-digit error ID, are sent to the signal destinations, complete with date and time. Tab. 7.7 describes the possible causes for these event messages.

Tab. 7.7: Breakdown of 5-digit error ID

ID	Possible cause:
10002	Unknown error cause on the AD2 bus. Example: Due to a loose contact an Office terminal sends a message the PBX does not understand.
12067	A defective terminal keeps trying to log on. The terminal can only be located via the I-bus tier (bus monitor). It must first be activated and then stopped shortly after the fault. Only specialists can carry out such analyses.
12097	Fault on 1 or more radio units. The radio unit(s) need to be reset or replaced. The radio unit(s) can only be located via the I-bus tier. It must first be activated and then stopped shortly after the fault. Only a specialist can carry out the analysis.
14027	A handset tries to set up a connection; the system rejects the attempt.
16046	Framing or Parity Error. Possible causes: a defective radio unit or a defective Office terminal.
24013	Text longer than expected. Recommendation: 1. Download all the configuration data using AIMS. 2. Initialize the PBX. 3. Upload the configuration data.

4.1.2 Event tables

Event tables list all the event messages the PBX is capable of generating (see Tab. 7.6).

The frequency of event messages can range between "0" and "20".
The time period is indicated in hours, ranging between "0" and "672". The longest time period "672" corresponds to 28 days or 4 weeks.

If the frequency of event messages is set to "0", the time period will also automatically be set to "0". No event message is sent to a signal destination.

If the frequency of event messages is set to "1", the time period will automatically be set to "0".
The event message will immediately be sent to the signal destinations.

If the time period is set to "0" hours, the frequency of the event message will automatically be set to "1".
The event message will immediately be sent to the signal destinations.

There are 4 event tables in the Fault & Maintenance Manager. Each event table can be individually configured and allocated to one of the 4 signal destinations. This means it is possible to decide which event message – if any – should be sent to a particular signal destination either immediately, with a delay or not at all.

Example

Tab. 7.8: Example of event table

Event type	Frequency	Time period
Total sync. loss	10	1
Sync. Loss on T / T2	1	0
System Overload	1	0
Outgoing Call Rejected	1	0
No response from network	0	0

In this example with event type "Total Sync Loss", an event message is sent to the signal destinations if the PBX generates the event message "Total Sync Loss" 10 times within one hour With event types "Sync Loss on T / T2 to "Outgoing Call Rejected",an event message is sent to the signal destinations immediately; with event type "No Response from Network", no event message needs to be sent to the signal destinations.

4.1.3 Signal destinations

4 signal destinations can be configured in the Fault & Maintenance Manager. Any one of the four event tables can be assigned to each signal destination.

When the PBX is first started, one event table with its own number is assigned to each signal destination.

Signal destinations include:

- Terminal (system terminals of the Office family, with alphanumeric display)
- External signal destination (signal destination PC (SEM) via ISDN or LAN / WAN to T interface)
- Local signal destination (e.g. PC (SEM) / printer on V.24 interface, PC (SEM) on S interface / Ethernet interface)
- Event log (event protocols in the Fault & Maintenance Manager)

There are several possibilities for connecting the signal destinations with a PBX:

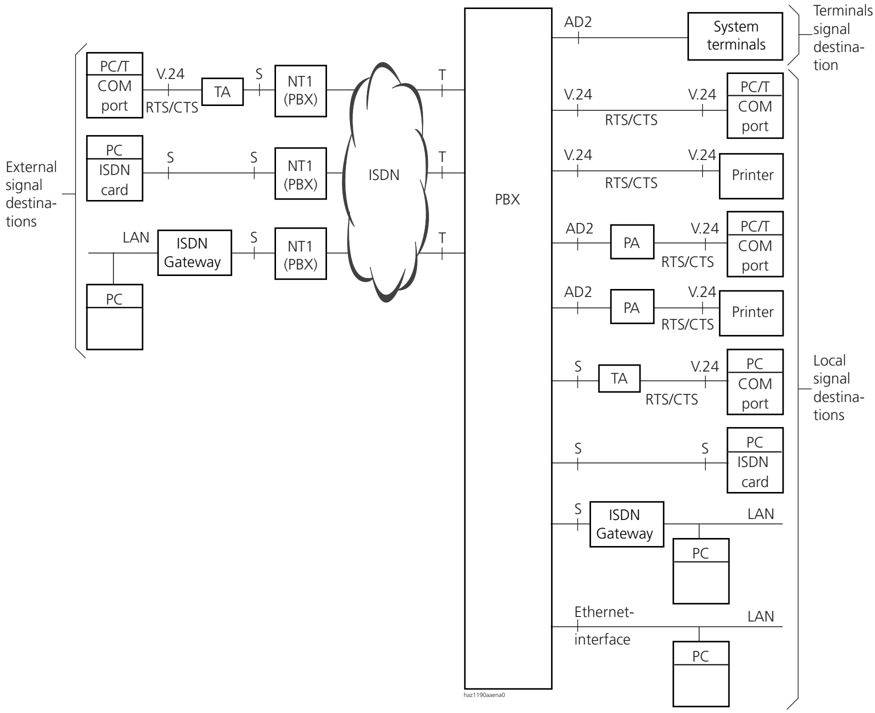


Fig. 7.9: Overview of connection possibilities for the various signal destinations

4.1.3.1 Signal Destination: terminal

Depending on the allocated event table, event messages are sent to all the system terminals of the Office family that have a display and are entered in message group 8 (System 2025 / 2045) and 16 (System 2065) respectively.

4.1.3.2 External signal destinations

Depending on the event table allocated, event messages are sent to a specified external signal destination. Two external signal destinations can be specified:

- 1 preferred external signal destination

- 1 alternative external signal destination

Signalling an event message to an external signal destination

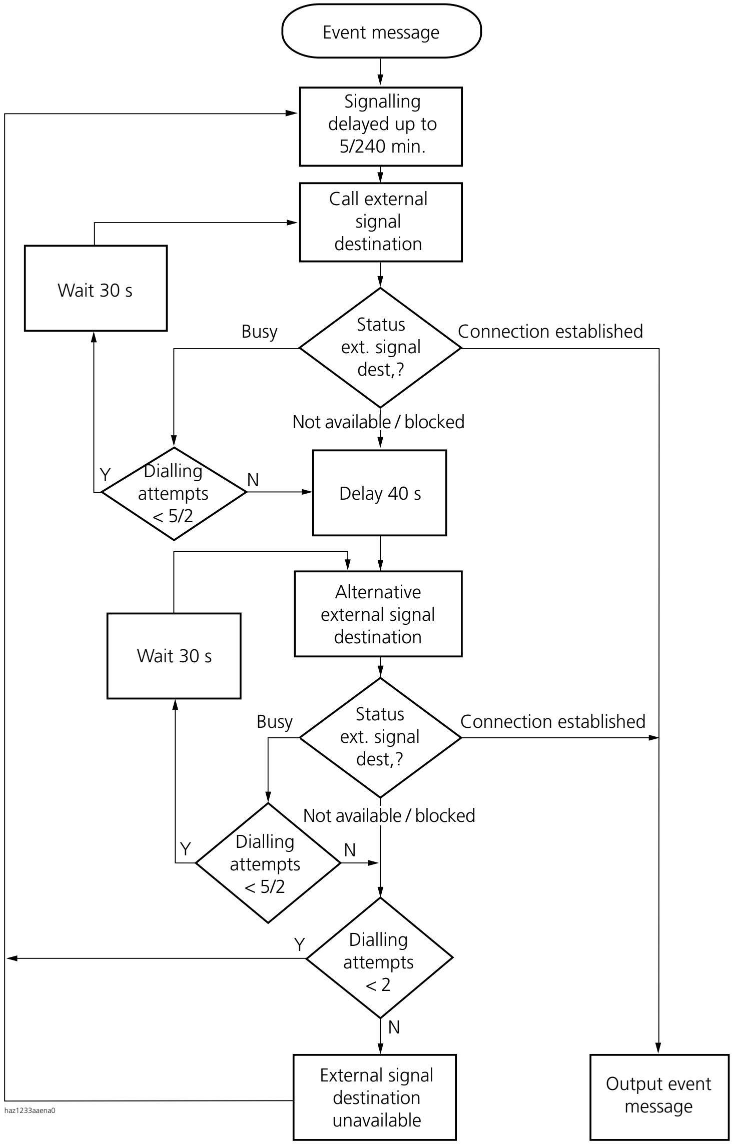


Fig. 7.10: Flowchart of the signalling of an event message to an external signal destination

The following principles govern the way event messages are signalled to an external signal destination:

- Individual event messages are not signalled if they occur at short intervals. The event messages are stored temporarily for 5 minutes and then sent together to the external signal destination.
- If over a period of one hour an attempt is made unsuccessfully to send the event messages to the external signal destination, the signalling period is extended from 5 minutes to 4 hours. As soon as the event messages are successfully output at the external signal destination, the time period is reset to 5 minutes.
- If over a period of 1 hour an attempt is made unsuccessfully to send an event message to an external signal destination, the number of dialling attempts is reduced from 5 to 2. As soon as an event message has been successfully sent, the number of dialling attempts is increased to 5 again.
- If the attempt to send an event message to an external signal destination was unsuccessful, the PBX will generate the event message "Text. No Response from External Destination".

Note:
Event tables and signal destinations should be configured so that the event message Text: "No Response from External Destination" is immediately signalled to a still available signal destination.

Routing an external signal destination

The following points are to be taken into account when specifying the routing to external signal destinations:

- If the external signal destination is dialled up via an exchange access prefix followed by the call number, the call will be routed via route 3. To use a different route, you need to configure a route selection.
- Digit barring for external calls and printer faults (in the case of call logging) do not affect outgoing event messages.

Configuring external signal destinations

If a PBX sends an event message, the event message opens a PPP communication channel via the public network from the PBX to a Terminal Adapter, connected either directly to a PC with the System Event Manager (SEM) software program or indirectly via a LAN / WAN (ISDN gateway). Once the event has been confirmed, the PBX clears down the PPP connection.

There are several possibilities for connecting external signal destinations to a PBX:

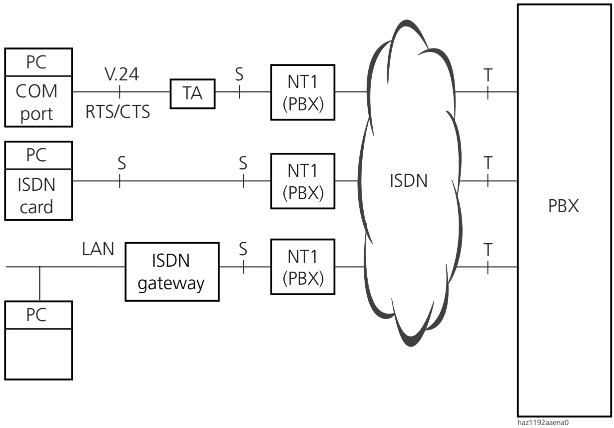


Fig. 7.11: Overview of connection possibilities for external signal destinations

Two external signal destinations (1 preferred and 1 alternative external signal destination) can be configured in the Fault & Maintenance Manager.

The PPP via ISDN communication protocol is used for the connection between the PBX and the external signal destination.

The following parameters need to be selected or entered in the Fault & Maintenance Manager for each of the two external signal destinations:

- Call number of the external signal destination (possibly as a route selection)
  Possible external signal destinations include:
  – Ordinary exchange output (route 3 is used)
  – Route selection
  – Cost centre selection (route 3)

  After initialization, the call charges are allocated to cost centre 100.
- IP address of the PC if the PBX is to connect with the PC via an ISDN gateway. If the PBX is to connect with the PC without ISDN gateway, the "IP address" entry should remain empty.
- TCP port number (the initialization value is 1062; if the value is changed in the SEM, it will have to be altered accordingly on the PBX side.)
- User name and password of the dial-up networking of the PC or ISDN gateway, to gain access via the TA or the ISDN gateway to the PC with the SEM.

See also:
For information on dial-up networking configuration on the PC, see "Settings on the PC", page 841.

Other necessary configurations

The following parameters must also be configured.

- In the Fault & Maintenance Manager:
  System ID of the PBX. This is important so that the PBX can be identified by the SEM. The system ID must match the system ID stored in the AIMS Shell. In the system ID you can store a serial number or the DDI number for the remote maintenance of the PBX (20 digits).
- In the Configuration Manager:
  Route 3 must be allocated trunk groups with digital network interfaces ("Routes" setting).

4.1.3.3 Local signal destinations

Depending on the event table allocated, event messages are sent to a specified local signal destination.

There are several possibilities for connecting a local signal destination to a PBX:

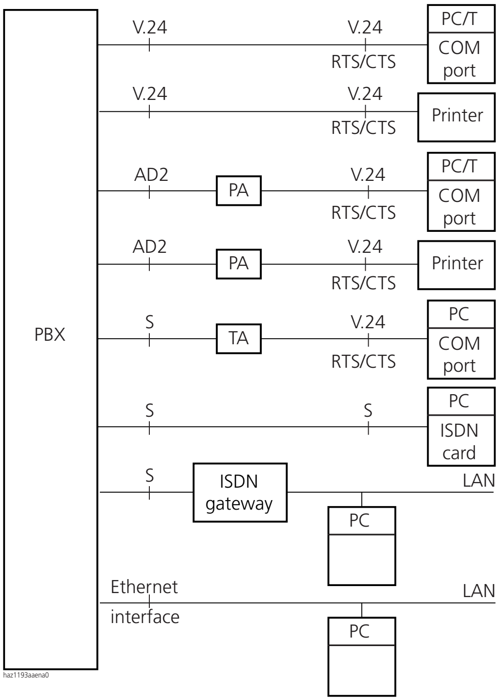


Fig. 7.12: Overview of connection possibilities for local signal destinations

Note:

Event tables and signal destinations should be set in such a way that the event message "No Response from Local Destination" is signalled immediately to any signal destination still available.

Configuring a local signal destination on a V.24 interface

The local signal destination is connected to the V.24 interface of the mainboard or of the Pocket Adapter (PA).

The "Local Output Interface" must be set on "Printer" in the Fault & Maintenance Manager.

The following parameters need to be selected or entered in the Fault & Maintenance Manager:

- Location of the V.24 interface to which the local output destination is connected (V.24 Main = Mainboard; V.24 PA = Pocket Adapter)
- Required output format "PC" or "Text" (see "Output formats", page 925)
- With "Text" output format: Language (de / fr / it[1] /en) in which the event messages are to be printed out.
- With the V.24 interface of the mainboard - port number
- With the V.24 interface on the PA - port address
- With "Text" output format and printer as the output location: Maximum page length of the printout

Output formats

There are two output formats available for local signal destinations connected to a V.24 interface.

- Text format
- PC format

Text format

```
11.03.97 10:12 LOSS OF SYNC ON T/T2 0.9
```

haz1234aaena0

Fig. 7.13: Event signal format "Text" for printer as the local output destination (example with date, time and event message)

PC format

The event signal format "PC" is required if a PC is configured as the local signal destination using the System Event Manager (SEM).

1) Depending on the country, the third language can also be a different one.
The language set here is also used for event messages in message group 8 (System 2025 / 2045) and 16 (System 2065) respectively.

Configuring a local signal destination on an S interface / ISDN

As with an external signal destination the event message opens a PPP communication channel from the PBX to a Terminal Adapter, connected either directly to a PC with the System Event Manager (SEM) or indirectly via a LAN (ISDN gateway). Once the event has been confirmed, the PBX clears down the PPP connection.

Configuring a local signal destination on an S interface

The event messages are displayed in PC format.

The "Local Output Interface" must be set on "IP Destination" in the Fault & Maintenance Manager.

The following parameters need to be selected or entered in the Fault & Maintenance Manager:

- Destination: Local PPP communication protocol.
- Call number of the local signal destination (the call number is checked by the PBX, a warning message will appear if the call number is incorrect)
- IP address of the PC if the PBX is to connect with the PC via an ISDN gateway. If the PBX is to connect with the PC without ISDN gateway, the "IP address" entry should remain empty.
- TCP port number (the initialization value is 1062; if the value is changed in the SEM, it will have to be altered accordingly on the PBX side.)
- User name and password of the dial-up networking of the PC or ISDN gateway, to gain access via the TA or the ISDN gateway to the PC with the SEM.

Configuring a local signal destination on ISDN

The event messages are displayed in PC format.

The "Local Output Interface" must be set on "IP Destination" in the Fault & Maintenance Manager.

The following parameters need to be selected or entered in the Fault & Maintenance Manager:

- Destination: PPP via ISDN communication protocol
- Call number of the local signal destination (the call number is not checked by the PBX, if the entered call number is incorrect, the PBX will issue the event message "Local signal destination does not answer").

- IP address of the PC if the PBX is to connect with the PC via an ISDN gateway. If the PBX is to connect with the PC without an ISDN gateway, the IP address entry should remain empty.
- TCP port number (the initialization value is 1062; if the value is changed in the SEM, it will have to be altered accordingly on the PBX side.)
- User name and password of the dial-up networking of the PC or ISDN gateway, to gain access via the TA or the ISDN gateway to the PC with the SEM.

See also:
Information on dial-up networking configuration on the PC see "Settings", page 840.

Configuring a local signal destination on an Ethernet interface

A PC (with the System Event Manager) connected either directly to the Ethernet interface or to the basic system via a LAN (LAN connection) can be configured as the local signal destination.

The event messages are displayed in PC format.

The "Local Output Interface" must be set on "IP Destination" in the Fault & Maintenance Manager.

The following parameters need to be selected or entered in the Fault & Maintenance Manager:

- Destination: Ethernet
- IP address of the PC
- TCP port number (the initialization value is 1062; if the value is changed in the SEM, it will have to be altered accordingly on the PBX side.)

See also:
Information on the connecting cables see "Ethernet Interface", page 799.

4.1.3.4 Signal destination Event Log

When the PBX is initialized, the signal destination Event Log is automatically allocated event table 4. In event table 4 the frequency for all event types (with the exception of the event type "Too many event messages") is set on "1" and the time period on "0". This means that all the PBX's event messages are immediately entered in the event log.

If the signal destination event log is assigned a different event table or if event table 4 is reconfigured, the event messages are entered in the event log in accordance with the new event table or the new configuration.

The Event Log consists of four protocols:

- Event messages (max. 254 entries)
- System failures (max. 80 entries)
- Power failures (max. 10 entries)
- Card failures (max. 150 entries)

If the maximum number of entries is exceeded, the oldest entry in each case is deleted.

These 4 Event Logs protocols are not printed out automatically; likewise, your attention is not drawn to any incoming event message. The protocols have to be retrieved manually in the Fault & Maintenance Manager or printed out.

See also:
Event messages entered in the protocols of the Event Logs can also be retrieved on the Office 45 using the System Assistant function (see "Maintenance menu on the Office 45", page 939).

4.1.3.5 Testing the signal destination configuration

To test the configuration, you can trigger a test event message via the Fault & Maintenance Manager. The event message is signalled without any delay, directly at the selected signal destination.

If the PBX is connected with AIMS via a TA, the test event messages will be signalled only once the connection is cleared down.

4.1.4 Suppressing event messages in the configuration mode

If the mainboard's HEX rotary switch is set on position 5 (configuration mode), no event messages will be generated when a card is replaced.

4.2 System Event Manager SEM

The System Event Manager (SEM) is a program capable of receiving and handling PBX event messages. It consists of the components:

- SEM Configurator
- SEM Server
- SEM Viewer

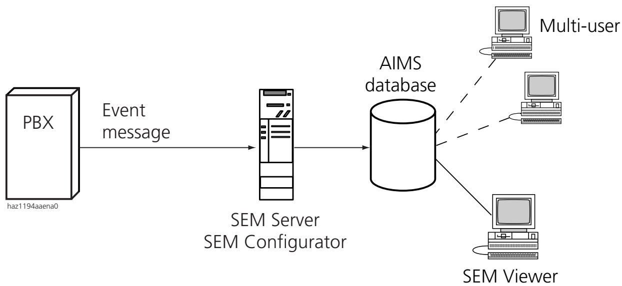


Fig. 7.14: SEM concept (general)

Event messages are sent to the SEM Server by the PBX. The SEM Server stores all the important data in the corresponding AIMS database. This data can be retrieved and edited using the SEM Viewer.

SEM Configurator

The SEM Configurator is used to start or end the SEM Server.

The relevant TCP port has to be activated for data transmission.

The event types can be assigned different priorities in the SEM Configurator.

The SEM Configurator is also used to set, for each individual AIMS database, the way in which the SEM Server should respond to each incoming event message (e.g. send e-mail, print out on a printer).

If you want a beep or melody to signal that the SEM Server has received an event message, make the appropriate setting in the SEM Configurator.

SEM Server

The SEM Server receives event messages from PBX systems and stores all the important data in the corresponding AIMS databases.

For the program to operate correctly, there has to be at least one AIMS database.

SEM Viewer

The SEM Viewer is used to edit the event messages stored by the SEM Server; they can then be filtered, sorted, printed or written into a file according to, for example , status, customer, PBX or priority.

The SEM Viewer can also be used to set, for each AIMS database, the way in which the SEM Server should respond to an event message (e.g. send e-mail, print out on a printer).

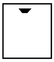
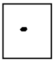
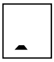
4.3 Operating state and error displays

4.3.1 Status display – basic system

The mainboard has a 7-segment display which indicates the system's current operating state, depending on the mode selected.

The main operating state displays in normal mode are listed below. The HEX rotary switch has to be on position 0, 2 or 5. A description of the various HEX rotary switch positions can be found in "Switch positions of the HEX rotary switch", page 935.

Tab. 7.9: Operating state displays of the 7-segment display in normal mode

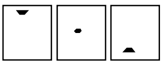
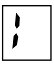
7-segment display	Detailed Description	State / Remarks
	Point is flashing[1]	System in operation
	Upper segment lit[1]	Internal port in operation
	Middle segment lit[1]	External port in operation
	Lower segment lit[1]	PBX prebarred

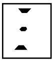
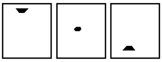
[1] Combinations of displays are possible

Note:
If the HEX rotary switch is in position B, the current software version will be displayed instead of the port states.

Besides the normal mode there are also various other modes with their own status display on the 7-segment display.

Tab. 7.10: Operating state displays other than normal mode (selection)

7-segment display	Detailed Description	State / Remarks
	All segments lit	Display test during startup
	Rapidly descending segments	RAM test during startup[1]
	Permanently displayed	Start of the startup mode
	Permanently displayed	Preparing to load the system software or searching for system software
	Rotating segments, forming the figure "8"	Loading the system software
	Permanently displayed	Action successfully completed

7-segment display	Detailed Description	State / Remarks
	Timeout symbol (permanently lit)	Timeout for turning HEX rotary switch
	Flashing	Waiting for V.24 transfer
	Descending segments	Loading application via V.24
	Rotating segments, forming the letter "o"	Data transfer to or from a Flash card
	Letter "E" followed by a three-digit figure and a bar (repeated)	An error with error code nnn has occurred[2)]

1) A thorough RAM test is carried out when the HEX switch is on position 2. In this case the segments move up and down slowly.

2) An overview of error codes can be found in "Output of error codes in the PBX", page 943.

4.3.2 State display for cordless system

4.3.2.1 LED on the radio unit Ascotel DECT

Each SB-4 and SB-8 radio unit is equipped with 3 LEDs. The radio unit operating state is indicated by different LED colours and flashing sequences in cycles of 1 s, specifically by the middle LED on the SB-4 and by the two outer LEDs on the SB-8 (separately for each AD2 bus). Each character (G, R or -) corresponds to 1/8 of a second.

Example

During the synchronization phase GGGGRRRR the LED flashes periodically 1/2 second green, 1/2 second red.

Tab. 7.11: Flashing sequences of the pilot LED on the DECT radio unit

State	Cycle	Meaning
No flashing	- - - - - - - -	Software not running / RU not connected
Red		Error:
	R R R R R R R -	AD2 bus not in order
	R - - - - - - -	Power supply error or AD2 line too long
Green / red		Startup process:
	G R R R R R R R	AD2 ok
	G R G R G R G R	Software downloading
	G G G G R R R R	Synchronizing
	G G G G G G G R	Starting DECT
	G G G G G R G R	HF Power Down / DECT System Status Passive[1)]
Green		Normal operation:
	G - - - - - - -	All B channels available
	G G G G - - - -	1 to 3 B channels busy
	G G G G G G G -	> 3 B channels busy

1) This operating state appears during a configuration data upload and after a system initialization. The radio unit is operational only once at least one DECT subscriber is entered in the numbering plan or the parameter "DECT System Status" in the AIMS Configuration Manager is on "Active".

An orange pilot LED indicates that the DECT signalling is active, i.e. DECT sequences are currently being transmitted between the handset and the radio unit. Examples:

- With each keystroke on the handset the LED briefly lights up orange.
- During a handset software download the orange LED remains lit until the download is completed.

On an SB-8ANT radio unit the middle LED indicates whether the internal or external antennas are active. If the LED is lit green, the external antennas are active.

4.3.2.2 LED on the charging bays

The LED on the charging bays indicates the charge level of the handset batteries when the handset is in the charging bay.

Tab. 7.12: LED on the charging bays

Charging bay	Office 100	Office 150 / Office 155pro
Batteries charging	LED flashing green	LED lit red
Batteries are charged	LED lit green	LED lit green
Handset is not in the charging bay or is incorrectly positioned in it	LED not lit	LED not lit

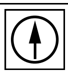
The charging bay of the Office 135 has no LED. On this model the battery charge level can be read from the handset's LCD display.

4.4 Self-tests

4.4.1 Basic system self-test

Self-tests are run automatically after a restart caused by an interruption in the power supply or after a PBX reset. The scope of the tests can be influenced during the startup process using the HEX rotary switch on the mainboard.

Tab. 7.13: HEX rotary switch

HEX rotary switch	Position	Remarks
	16 positions	Various coding options

4.4.1.1 Switch positions of the HEX rotary switch

Tab. 7.14: Functions of the switch positions

0	Normal operation; start with simple RAM test (normally takes about approx. 3 seconds). This test is triggered only after a restart when the 230 V mains supply to the system (Power Up) is switched on.
2	Same as 0, except with a thorough RAM test (approx. 1 minute) followed by an indication of the RAM size. This test is triggered only after a restart when the 230 V mains supply to the system (Power Up) is switched on.
5	Same as 0, except that the "Installer Level" configuration mode is enabled. (Only locally using the Office 45, Pocket Adapter or V.24; not in the case of remote maintenance via ISDN.)
B	Same as 0, except the installed version of the system software is displayed (e.g.: U.0.0.9.0<Pause>U.0.0.9.0… indicates Version 0.90).
Others	Reserved (same function as position 0).

Once the servicing phase is completed, turn the HEX rotary switch back to position 0.

4.4.1.2 Entering rotary Switch sequences using the example of the initialization function

Rotary switch sequences triggered during startup must be entered in a particular sequence.

Example

Tab. 7.15: Rotary switch sequences for initialization function

1-F-0	Forces a system initialization (also possible via the menu)

Starting position: PBX is in operation.

1. Turn the rotary switch to "1".
2. Press the Prog. pushbutton (see Fig. 7.1 and Fig. 7.2). Various self-tests are carried out.
3. Wait for the digit "1" to start flashing on the 7-segment display.
4. Turn the rotary switch to position "F".
   If the timeout symbol "≡" appears on the display, start again with step 1.
5. Wait for the character "F" (permanently lit) to appear on the 7-segment display.

6. Turn the rotary switch to position "0" (the "0" position completes a sequence).
7. Wait for the digit "0" (permanently lit) to appear on the 7-segment display. The corresponding function is automatically executed; if this is not possible, an error message will appear.

Note:
An incorrectly entered sequence can be corrected by repeating the input as of step 1.

See also:
"Hardware-induced initialization", page 853.

4.4.2 Self-test for the DECT system

Self-test for the Ascotel DECT radio unit

Once the AD2 interface has been connected, the radio unit carries out the following checks automatically:

- Is a plug-in power supply unit connected?
- Is the power available via the AD2 interface sufficient?
  This test is carried out via an internal load circuit if no plug-in power supply is connected. The test can take up to 10 seconds.

If the test was unsuccessful, the radio unit startup process will be halted; the pilot LED will flash red until the PBX resets the radio unit.

If the test was successful, the radio unit startup process will proceed and be completed.

Note:
After the system initialization the radio unit starts in status "AD2 OK". It is operational only once at least one DECT subscriber is entered in the numbering plan.

4.4.3 Sequence of self-tests for corded terminals

Once the terminal has been checked with regard to connections, unlocking, etc., the self-test can be used to detect possible sources of error.

Note:
Terminal self-tests cannot be carried out on the Office 10 system terminal.

Test procedure for Office 25

1. Unplug the terminal connecting line from the connection socket; with the Foxkey on the outer left pressed down, reconnect the terminal connecting line. Once the self-test mode has started, release the Foxkey.

   All the display symbols appear.
2. Test keys: Pick up the handset and press one key after the other.
   Each keystroke generates a tone in the loudspeaker.
3. To exit the self-test mode, unplug then reconnect the terminal connecting line.

Should any irregularities occur during the self-test, contact a specialist.

Test procedure for Office 35, Office 45 with alpha keyboard (AKB) and expansion keypad (EKP)

1. Unplug the terminal connecting line from the connection socket; with the Foxkey on the outer left pressed down, reconnect the terminal connecting line. Once the self-test mode has started, release the Foxkey.

   The LEDs go on and off, and the display symbols alternate periodically between lit and off.
2. With the handset off-hook, blow into the microphone.

   The microphone and earpiece are connected with each other; the noise can be heard in the earpiece.
3. Press the loudspeaker key.

   The hands-free microphone is briefly connected with the earpiece and the ambient noise is audible.
4. Test keys: Pick up the handset and press one key after the other.

   Each keystroke generates a tone in the loudspeaker.

5. Test the AKB and EKP supplementary equipment:

   All the LED displays on the connected supplementary equipment are flashing. Test keys: Press one key after the other.

   Each keystroke generates a tone in the loudspeaker. Pressing the keys "Shift", "Control" and "Alt" does not generate a tone.
6. To exit the self-test mode, unplug then reconnect the terminal connecting line.

Should any irregularities occur during the self-test, contact a specialist.

Test procedure for Office 20, Office 30, Office 40

1. Unplug the terminal connecting line from the connection socket; with the Foxkey on the outer left pressed down, reconnect the terminal connecting line. Once the self-test mode has started, release the Foxkey.
2. Office 20 automatic test: All the display symbols appear.

   Office 30, Office 40 automatic test: The LEDs go on and off, and the display symbols alternate continuously between lit and off.
3. Key test (Office 20): Do not pick up the handset.
   Key test (Office 30, Office 40): Pick up the handset and press one key after the other.

   Each keystroke generates a tone in the loudspeaker.
4. Microphone and earpiece are connected with each other: Blow into the microphone; you should be able to hear the noise in the earpiece.
5. Press the loudspeaker key (Office 30, Office 40 only):

   The hands-free microphone is briefly connected with the earpiece. The ambient noise is audible.
6. To exit the self-test mode, unplug then reconnect the terminal connecting line.

Test procedure for the additional keypad (ZTF)

1. Unplug the terminal connecting line, then plug the connector back in again while keeping the topmost key pressed down.

   The ZTF LEDs go on and off.
2. Press any key on the ZTF.

   Only the LED of the pressed key lights up.

3. Press one key after the other.

   Only the LED of the key pressed in each case lights up.
4. To exit the self-test mode, unplug then reconnect the terminal connecting line.

Should any irregularities occur during the self-test, contact a specialist.

4.5 Other aids

4.5.1 Maintenance menu on the Office 45

The System Assistant function under the "Maintenance" menu item on the Office 45 can be used to retrieve system information which in the event of a malfunction provides important clues as to the cause of the fault:

Tab. 7.16: "Maintenance" menu selection:

1: View	3: Delete
2: Print	4: Both

You can select from the following menu items:

1. System status
2. System failures
3. Mains voltage failures
4. Event messages

System status menu item

Tab. 7.17: Display of the system status lines

== SYSTEM STATUS		
BCS: 00000 SUBS: 0011 DIST: 0001 Back with [<-- |]	CC: 00000 NSUB: 0000 DDIN: 0000	 LINE: 0001 ABB: 1000

The system status lines provide useful information for a more in-depth fault diagnosis. They can be printed out and sent to customer support on request.

Tab. 7.18: The displayed data and what it means

Display	Description	Normal value / idle state	Note
BCS: xxxxx	Number of existing BCS references	BCS: 00000	Each active connection needs 2 BCS references
CC: xxxxx	Number of existing call controls	CC: 00000	For each BCS reference there is one or more CC
SUBS: xxxx	Number of subscribers in the system	SUBS: 0000	0000: No ports busy
NSUB: xxxx	Number of PISN subscribers in the system	NSUB: 0000	0000: No PISN SC in the system
LINE: xxxx	Number of lines in the system	LINE: 0000	0000: No lines defined
DIST: xxxx	Number of call distributions in the system	DIST: 0000	0000: No call distribution defined
DDIN: xxxx	Number of DDI numbers in the system	DDIN: 0000	0000: No DDI numbers defined
ABB: xxxx	Number of abbreviated dialling numbers in the system	ABB: 1000	1000: Initialization value unchanged

System failures menu item

Tab. 7.19: System failures display

== SYSTEM FAILURES				52
W 15.09.00	13:32	011A59F2,	011A5A8C,	01156FFE
W 06.12.00	13:32	011A59F7	011A5A82	01156FF1

The system's last 80 system failures (resets) are displayed. The resets are incremented in the counter in the top right (0...255).

Significance of the display: Error type W = restart (watchdog), date, time

When printing, only the last 4 addresses are printed out.

Power failures menu item

Tab. 7.20: Power failures display

== POWER FAILURES	
01.12.98	16:13

Only the restart time is recorded.

Event messages menu item

Tab. 7.21: Event messages display

== EVENT MESSAGES		
01.12.98	00:01	OUTGOING CALL REJECTED TO LINE: 12.25
02.12.98	09:15	TOO MANY EVENT MESSAGES

The event messages are identical to the displays obtained when entering the configuration with System Assistant (Office 45). They are stored in event table 4 (see "Event message concept", page 913). The entries in the Maintenance menu remain stored until they are deleted with the "Delete" command.

Remarks

- The display on the event message header line ("1") indicates the number of event messages that have occurred (max. 255). You can use the cursor keys to scroll through the lines (not visible)(e.g.4 events 1/2: 2 events on page 1,2 events on page 2)
- The last 254 entries can be displayed.
- The event entries record only the time of the error incident, not the time at which it was remedied.
- More detailed information on triggering and printing event messages can be found in "Event message concept", page 913.

See also:
Event messages can also be called up in the AIMS Fault & Maintenance Manager (see "Signal destination Event Log", page 928).

4.5.2 AD2 Monitor pro

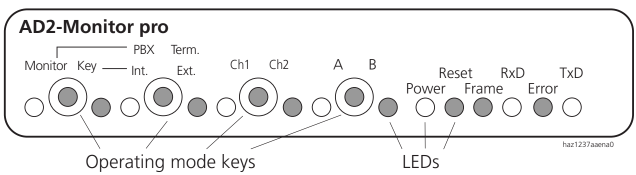


Fig. 7.15: Front view of the AD2 Monitor pro

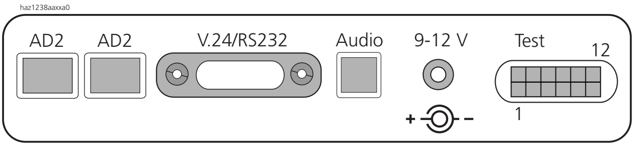


Fig. 7.16: Rear view of the AD2 Monitor pro

The AD2 Monitor pro is used for the following tasks:

- Monitoring the AD2 interface including addressing, installation and power supply, and recording the data flow in both directions with output on a V.24 terminal.
- Simulating a terminal with an AD2 interface that answers data sent by the PBX or is controlled via V.24.

The AD2 Monitor pro is looped into the connecting line as near as possible to the terminals. It can be set for several operating modes:

- MM Monitor Mode: Passive line monitoring
- MT terminal mode: AD2 terminal simulation incl. line monitoring
- Audio Int. / PBX:
  - With MN, the PBX's B channel is switched to handset.
  - With MT, the internal PCM Codec is activated.
- Audio Ext. / PBX:
  - With MN, the PBX's B channel is switched to handset.
  - With MT, the externally connectable PCM Codec is activated.
- Channel 1/2 M1/M2: Selects the AD2 channel used
- Switch A/B: MA: Controls the output for the test connector

The AD2 Monitor pro offers user-friendly settings on the device itself, with LED displays for the transmit and receive channels as well as operation via V.24, including data logging with the setting of breakpoints. Updated AD2 Monitor pro software can be loaded directly onto the Monitor from a PC, as an Intel Hex file.

The Quick User's Guide supplied with the Monitor provides an overview of all the possibilities and applications for the AD2 Monitor pro.

4.5.3 Fault & Maintenance Manager

For information on the Fault & Maintenance Manager see "Event message concept", page 913.

4.5.4 System Event Manager SEM

For information on the System Event Manager (SEM) see "System Event Manager SEM", page 929.

4.5.5 Measuring equipment for cordless systems

The aids required for measuring out DECT systems are described in "Measuring Equipment", page 635.

4.6 Error causes and error handling

The following tables contain a systematic list of occasional errors. These tables are designed to help you pinpoint and remedy errors more easily.

When the system is operating correctly, the decimal point on the mainboard's 7-segment display flashes regularly.
In the event of sporadic errors check the installation for earth loops (see "Checking the installation", page 833).

4.6.1 Output of error codes in the PBX

Whenever the system detects an error, it displays the corresponding error code on the mainboard's 7-segment display (providing the system is still powered and the display is working).

There are two types of error:

- Unrecoverable errors that lead to a restart. The current system software is unable to remedy the error that has occurred. The system is reset and the initialization restarted.
- Errors following the failure of partial functions. These errors occur when the user carries out a function incorrectly or when non-critical system components fail.

Errors that concern partial functions overwrite the standard output of the mainboard's 7-segment display for 10 seconds. Errors that lead to a restart are displayed permanently.

Error messages are characterized by the letter E (= error), followed by a three-digit error code. The individual digits appear one after the other on the single-digit status display of the mainboard (7-segment display with decimal point).

In the example below (system error message 022) each individual digit appears for 0.5 seconds. The hyphen marks the start of
the 3-digit error message

E.0.2.2.-.E.0.2.2. - ....

4.6.2 Error coding

The first digit indicates where the error has occurred; the two-digit code that follows classifies the error.

Tab. 7.22: Overview of error localization

Code	Description
E-1xx	Mainboard error
E-2xx	(unused)
E-3xx	(unused)
E-4xx	Error during software version transfer
E-5xx	Self-test error
E-6xx	Data transmission error

The following overview of error messages contains the error code and a column with the description of the error.

Tab. 7.23: Mainboard error messages

Identification	Error description	Error handling
105	RESA not responding	• Replace mainboard
106	Clock cannot be adjusted	• Replace mainboard
107	Clock not working	• Replace mainboard
108	EIM not responding	• Replace EIM card
109	EIM defective	• Replace EIM card

Tab. 7.24: Error message during software version transfer (version transfer configuration data organization OIM)

Identification	Error description	Error handling
402	Version transfer not possible due to incorrect mainboard	• Replace mainboard
403	Version transfer not possible: Version of the PBX software used to create the configuration data is more recent than the software currently on the PBX	• Load new software onto mainboard
404	Version transfer not possible: Country and / or sales channel on the PBX (EIM card) does not match information in the configuration data on the Flash (e.g. country code in Flash memory is CH and in the software it is D)	• Fit a different mainboard with correct country and / or distribution channel
405	Version transfer not possible: Software release unknown.	• Load new software onto mainboard

Note:
If error codes 402 to 405 occur, the system will halt the startup. Proceeding with the startup results in a system initialization.

Tab. 7.25: Self-test error messages (system unable to run)

Identification	Error description	Error handling
500	Program checksum incorrect	• Load correct software onto the PBX
503	Too many system restarts	• Replace mainboard
504	Program error after startup	• Replace mainboard or RAM card
505	DRAM SIMM defective	• Replace mainboard
520	Incompatible boot software (e.g. the 2065 system cannot load I6 system software using I5 boot software)	• New Flash card with appropriate boot software required (e.g. from the I6 "Update-Set")
521	Incompatible RAM size (e.g. the I6 system 2065 requires a 64MB RAM)	• New RAM card with appropriate RAM component required (e.g. from the I6 "Update-Set")
522	Incompatible Flash size (e.g. the I6 system 2065 requires a 8MB Flash memory)	• New Flash card with appropriate Flash component required (e.g. from the I6 "Update-Set")

Tab. 7.26: Self-test error messages in Xmodem mode (system unable to run)

Identification	Error description	Error handling
601	XMODEM transfer aborted	• Retry transfer
602	XMODEM transfer rejected	• Ensure that correct file is being transmitted

Tab. 7.27: General self-test error displays

7-segment display	Description
Horizontal bars moving up and down	Self-test in progress
3 horizontal bars	Time-out appears, i.e. an input via BCD code is expected on the HEX rotary switch.
E U L	Emergency Upload system software
P (flashing)	Software waiting for software transfer via Xmodem.
n o L I C	EIM error or missing EIM card

4.6.3 PBX will not start

Tab. 7.28: PBX will not start

Error description	Error handling
No decimal point (DP) on mainboard status display.	• Check power supply
DP not flashing, 7-segment display reads: 8, PBX will not start up.	• Check whether – software is loaded – software is correctly loaded • If basic system is defective, replace basic system

4.6.4 Malfunction during PBX configuration

Tab. 7.29: Malfunction when configuring the PBX with AIMS in online operation using internal access

Error description	Error cause / error handling
AIMS online operation cannot be activated or setup of online connection is terminated; AIMS Configuration Manager cannot therefore be accessed.	• Check connecting table (type, connector). • Check whether internal access you want is selected in AIMS: – Start AIMS Shell – Log into PBX – Under "PBX: Activate required internal access. • Check communication parameter settings under Windows; if necessary, set identical to PBX side: – IP addresses – TCP / IP settings in the dial-up networking – Under COM Port / set local parameters to same settings as on PBX side

Tab. 7.30: Malfunction when configuring PBX with AIMS in online operation using external access (remote configuration)

Error description	Error cause / error handling
AIMS online operation cannot be activated or setup of online connection is terminated; AIMS Configuration Manager cannot therefore be accessed.	• Check infrastructure for remote configuration: – Ensure that remote configuration is enabled on the PBX. • Check whether the required dial-up access in AIMS is selected: – Start AIMS Shell. – Log into the PBX. – Under "PBX: Activate the "Dial-up access" online connection. • Checks communication parameter settings under Windows: – COM port speed – IP addresses – TCP / IP settings in the dial-up networking • Check Terminal Adapter settings in connection with the PC.

Tab. 7.31: Malfunction when configuring the PBX with AIMS in online operation via LAN

Error description	Error handling
AIMS online operation cannot be activated or setup of online connection is terminated; AIMS Configuration Manager cannot therefore be accessed.	• Check whether required LAN access in AIMS is selected: – Start AIMS Shell – Log into PBX – Under "PBX: Activate the "LAN" online connection. – Check whether correct IP address is entered in the AIMS Shell. – Check whether there is an answer on the IP address (DOS command "ping", e.g. "ping 192.168.104 13")

See also:
For information on setting communication parameters under Windows see the User's Guide for setting up a PC-PBX connection in the AIMS Information Manager.

4.6.5 Malfunctions on the system as a whole

Tab. 7.32: Malfunctions on the system as a whole

Error description	Error cause / error handling
No connection to public network.	See "Malfunctions on network side", page 948
Dial tone available on all network interfaces; no power supply.	• Check -40 V power supply
No music audible on hold.	Music source not connected or not activated? Volume set too low in the menu Music on hold not activated
External outgoing analogue connections with long numbers on pulse dialling do not work.	Incorrect configuration • Activate dial tone detection in the PBX
No charge pulses detected / transmitted.	• Arrange for charge pulses to be activated at the local exchange
If all items are in order	• Trigger restart (disconnect then reconnect the power supply (system reset) • If system is not running, the following defects are possible: – Basic system defective – Flash card (configuration data, system software, boot software) defective. – Expansion cards incorrectly fitted – Connector tabs bent

4.6.6 Malfunctions on network side

Tab. 7.33: Malfunctions on network side

Error description	Error cause / error handling
When calls are made via basic accesses, the caller obtains a CP message or a congestion tone even though the figure for "Maximum incoming calls" has not been exceeded.	Subscriber to whom the external call is routed is not connected Set protocol in the exchange configuration is incorrect. Basic access incorrectly configured • Configure for point-to-multipoint or point-to-point in accordance with local exchange default.
DDI numbers reach CDE 1 (UG16) instead of the configured destinations.	Length of system's configured DDI numbers does not match length of DDI numbers sent by local exchange

Error description	Error cause / error handling
It takes more than 6 s to obtain an exchange dial tone.	Network interfaces not connected or defective • Bar outgoing calls on any network interface not connected • Test exchange lines in accordance with Part 6 "Commissioning"
Outgoing key telephone traffic is disrupted.	Incorrect authorizations configured (semi-restricted / outgoing calls barred)

4.6.7 Malfunctions on subscriber side

Tab. 7.34: Malfunctions on the subscriber side

Error description	Error cause / error handling
Terminals with configurable dialling method experience sporadic malfunctions whenever control key is pressed.	System earth must not be connected on terminals configured for MFV / DTMF (double signalling on Flash / earth key).
Analogue terminals obtain congestion tone instead of dial tone when off-hook.	No call number allocated to user-network interface
System terminals obtain congestion tone when a line is seized; display reads "Number invalid".	No call number allocated to user-network interface Terminal allocated incorrect terminal selection digit (TSD)
System terminals do not obtain dial tone (dial signal) when seizing a line; display reads "Not available".	• Check installation or connecting cable Expansion card is prebarred • Pull out expansion card, then plug it back in again Line length of an S bus longer than 150 m is defined as "short" instead of "long" in the S-bus port configuration • Replace terminal if necessary

4.6.8 Malfunctions with call logging (CL)

Tab. 7.35: Malfunctions with call logging (CL)

Error description	Error cause / error handling
Operator Console charges not transferred to next subscriber.	A different terminal selection digit was probably set on the Operator Console. This means the terminal is logged on in the PBX with the new terminal selection digit. • Unplug the Operator Console, then reconnect it after a few seconds
The event message "CL printer jam" appears and cannot be cleared. This means that after approx. 300 calls (System 2025 / 2045) or 1000 calls (System 2065) none of the subscribers will be able to make outgoing phone calls as the exchange output will be barred by the emergency digit barring.	If the printer is OK, the OCL / ICL memory may be full. To clear it, proceed as follows: • V.24 mainboard: Set "–" mode • This measure will result in the loss of stored printer data.
The event message "ICC OVERFLOW" appears and the ICC data is given the prefix "+".	Charge counter has reached maximum value of 2,147,483 and begins at zero again • Print out or clear the counter Printer is blocked (paper jam, out of paper): • Clear printer and reset

Error description	Error cause / error handling
No more ICL data is output via the V.24	Memory overflow if ICL printer is blocked (the same memory is used). • Clear the printer jam or, if necessary, • Clear the OCL / ICL memory (see above)

4.6.9 Malfunctions with the Telecourier 900 pager system (PS)

Tab. 7.36: Malfunctions with the PS

Error description	Error cause / error handling
Paging system (PS) does not work.	• Check whether the PS has been configured
The PS receivers continue to ring after answering (several search paths).	The search duration t_r is set too short in the AIMS configuration. t_r must be ≥ the search duration set in the PS. • Adjust the search duration t_r

4.6.10 Malfunctions of the DECT radio unit

Tab. 7.37: Malfunctions of the DECT radio unit

Error description	Error cause / error handling
No radio connection in a coverage area.	Check LED on radio unit. Red LED flashing: • Check power supply / line length of AD2 bus cable (LED flashes briefly at regular intervals) • Check AD2 bus cable (LED goes out briefly at regular intervals) • Unplug AD2 bus cable for one minute, then reconnect Green LED flashing long: • All B channels busy
Radio unit not activated.	• If LED on radio unit is flashing – Radio unit in startup phase – Radio unit defective • If LED on radio unit not flashing – Check trunk connections – Radio unit defective

See also:
A detailed description of the various flashing sequences for the radio unit LED can be found in "State display for cordless system", page 932.

4.6.11 Malfunctions of the DECT handset

Tab. 7.38: Malfunctions of the DECT handset

Error description	Error cause / error handling
No display.	• Switch handset on and test • Replace or charge battery
No radio link to radio unit; no aerial symbol.	Check coverage area (within range of a radio unit). • Check radio units in this section Handset not logged on to system • Log handset on
Impossible to dial.	Keypad blocked (keylock) • Reactivate keypad
No dial tone.	• Check radio units in this section (see below)
Poor connection quality (echo effect).	• Activate echo compensation
Handset beeps approx. every 10 s during a call (or in standby) while battery indicator is flashing.	• Replace battery immediately, either after or during the call (see handset operating instructions)
Call breaking up.	You are moving out of range. • Find a location with a better radio contact, see "Coverage area", page 613.
Handset cannot be reached.	System terminal calls handset and obtains the following responses: Busy tone obtained and display reads "Busy" • Handset is busy Congestion tone obtained and display reads "Connection overloaded" • All radio channels busy If congestion tone is obtained after 8 seconds and display reads "No answer" Reasons why handset could not be reached: • Handset switched off. • Handset not within reachable radio area. • No radio channels currently available. • Handset not logged on to system • Call diverted due to unobtainable
Handset not ringing.	Activate tone ringing
Handset cannot be configured; password missing (or forgotten).	Overwrite password at Attendant authorization level (order to Attendant)

4.6.12 Overload code displays

The overload code displays on the DECT handsets can be activated and deactivated using the following key combination (toggle function):[1)]
Long-click key 5 and then long-click key 3 (long-click = > 2 seconds).

The overload code display is always deactivated after system initialization.

Tab. 7.39: DECT overload code displays Office 135 / Office 155pro

Code	Name	Error description	Error handling
05 / 06	IPEI Not Accepted	Handset already logged on to system under different number.	• Check existing subscriber No. under "Config."; log that particular subscriber off • Try again
10	Authentication failed	Logon error	• Try again
51	DL 04 Expiry	Timer (on handset) has expired	• Try again
70	Timer Expired	MM timer in system has expired (during logon)	• Try again
44	Failure to set up traffic bearer	Connection cannot be set up as too many handsets are phoning within the same range	• Try again • If still unsuccessful after several attempts, "Reset Handset" (normally it is enough to press key 0 with a long keystroke and switch back on)
45	No Quiet Channel	No channel available, same as code 44	Same measures as for code 44

Tab. 7.40: DECT overload code displays Office 100 / Office 150

Code	Name	Error description	Error handling
05	IPEI Not Accepted	Handset already logged on to system under different number.	• Checking existing subscriber No • Log subscriber off under "Config." • Try again
OD	Timer Expired	Timer in system has expired	• Try again
10	Authentication failed	Logon error	• Try again
31	System busy	System overload	• Try again later
41	Unsuccessful Power Up	Power-up failed; handset remains in "sleep" mode	• Try again – If unsuccessful, "Reset Handset"

1) Office 135: Possibly only as of software version 2.00.

Code	Name	Error description	Error handling
43	LMAC Initiated Release of Original Bearer	a) Disconnection or b) Connection cannot be set up (e.g. due to a hanging link) or c) Handset already logged on to system	a) • Go back to obtain a recall b) • Try again • If still unsuccessful after 6 or 7 attempts, reset radio unit or basic system c) • Delete subscription on handset (in "System" menu) • Log handset back on to system
44	Failure to set up traffic bearer	Connection cannot be set up as too many handsets are phoning within the same range	• Try again • If still unsuccessful after several attempts, "Reset Handset" (normally it is enough to press key 0 with a long keystroke and switch back on)
45	No Quiet Channel	No channel available, same as code 44	Same measures as for code 44
51	DL 04 Expiry	Timer (on handset) has expired	• Try again
70	Timer Expired	MM timer in system has expired (during logon)	• Try again

4.6.13 Malfunctions of the DECT charging bays

Office 135

Tab. 7.41: Malfunctions of the DECT charging bay

Error description	Error cause / error handling
Handset will not charge.	• Connect power supply • Check batteries and replace if necessary About the charging process: • Battery symbol on handset flashes when battery is being charged • Check tone indicates correct contact

Office 100 / Office 150 / Office 155pro

Tab. 7.42: Malfunctions of the DECT charging bay

Error description	Error cause / error handling
Handset will not charge.	• Connect power supply Display should light up or flash when handset is in place. • Check batteries and replace if necessary About the charging process • Display flashes green (Office 100) or is lit red (Office 150 / Office 155pro) when the battery is charging. • Display is lit green when battery is fully charged • Check tone indicates correct contact

Part 8 Annex

1 Overview of Chapters

Generation Change

This chapter describes the generation change from I5 to I6. The upgrade procedures for a 2025 / 2045 system and a 2065 system are explained step by step. The procedure for turning an I6 system back into an I5 system is also described. The Chapter ends with information on which functions and products are no longer supported on I5 and I6 systems.

Systematic Designation System

Chapter 3 explains the PCB designations, rating plates and stickers, and provides an overview of the PCBs and the corresponding material numbers.

DECT Compatibility

The tables in Chapter 4 show which features are supported by Ascotel IntelliGate and the cordless terminals of the Office family in accordance with the GAP standard, and which radio units and handsets are compatible with which PBX software release.

Taking Over a Crystal Phone Book

This chapter describes how to copy the data from the Crystal private phone book into the private telephone book of any corded or cordless Office terminal.

Technical Data

Chapter 6 contains general technical data such as dimensions, weights, ambient temperatures and primary electrical data, as well as data on network interfaces, user-network interfaces and special interfaces.

PC Dial commands

The PC Dial application lets you address the PBX directly from your PC. Chapter 7 lists the commands available for this purpose.

Operation of the system terminals

Chapter 8 provides an overview in table form of the control elements and digit key assignment of system terminals. It also contains the assignment of the alpha keyboards and lists the available function commands for system terminals.

Additional documentation

A list of additional documents accessible on the Internet is to be found here.

2 Generation Change

Ever since it was first introduced in 1988, the system has undergone a number of further developments. The PBX hardware, among other things, was entirely redesigned at the time of the generational change to I5. It is therefore no longer possible to upgrade or expand an older system to I5 or I6.
New software is required for the generation change from I5 to I6. On the 2065 system the RAM and Flash cards also need to be replaced. All the existing expansion cards and terminals can still be used on the I6 system.

Note:
A 2065 system can only be upgraded from I5 to I6 on site as manual interventions are required on the system.

2.1 System Upgrade 2025 / 2045

Upgrading a 2025 / 2045 system from I5.X to I6.X requires only new system software. The system upgrade can be carried out on site or remotely:

1. Initiate the upload process as described in the paragraph "Initiating an upload process", page 894. Select "Start Now" as the activation time.
   Note: To upload the system software you first need to change the password in AIMS 5 once.
2. Once the upload is completed, the PBX resets and is then ready to operate again with the new I6 system software and the usual system configuration.

AIMS 6 is required in order to update the I6 system. Once AIMS 6 has been installed and started, the switch in PBX generation from I5 to I6 has to be carried out on the corresponding system. Note: If the PBX generation is incorrect, AIMS 6 cannot access the PBX after the software upgrade and generates the corresponding error message.

2.2 System Upgrade 2065

Upgrading a 2065 system to I6 requires AIMS 6, a new system software and additional RAM and Flash memory. An "Update-Set" consisting of a RAM card and a Flash card is provided for this purpose. The Flash card also contains an I6 system software, which simplifies the upgrade:

1. Install AIMS 6 and create a new PBX of the corresponding type. Select I5 as the PBX generation.
2. Save the configuration data using a download followed by a backup (see "Saving configuration data with AIMS", page 900). This is necessary as the current PBX data cannot be transferred to the new system automatically.
3. Change the PBX generation in AIMS 6 to I6.
   Note: If the PBX generation is incorrect, AIMS 6 cannot access the PBX after the software upgrade and generates the corresponding error message.
4. Replacing the Flash card: Carry out steps 2-9 as described in "Replacing the Flash Card", page 961.
5. Replacing the RAM card: Carry out steps 4-12 as described in "Replacing the RAM Card", page 963.
6. The PBX carries out a first start and starts with the I6 system software. To load a more recent system software than the one in the Flash memory, initiate the upload process as described in the paragraph "Initiating an upload process", page 894. Select "Start Now" as the activation time.
   Note: To upload the system software you need to change the password in AIMS 6 once.
7. Once the system has started up, load the saved configuration data using AIMS 6 under "Restore", and save the data in the PBX with "Upload".
8. The PBX reboots and is then ready for operation with the new I6 system software and the familiar system configuration.

Note:
An I6 system software on a 2065 system with insufficient Flash and/or RAM memory is unable to run, and outputs the corresponding error messages. The system hardware has to be expanded beforehand using the "Update-Set".

2.3 System downgrade from I6 to I5

A successful downgrade requires the existence of an I5 system software and valid configuration data. AIMS 6 is backwards compatible with AIMS 5, which means that a change of AIMS is not compulsory.

System Downgrade with Flash Card

The simplest way of obtaining an I5 system from an I6 system is to replace the Flash card with the loaded I5 system software (see "Replacing the Flash Card", page 961). The configuration data is also stored on the Flash card. On the 2065 system you also need to change the RAM card (see "Replacing the RAM Card", page 963). After the start-up the system is ready to operate again with the old system configuration.

System Downgrade with Software Upload

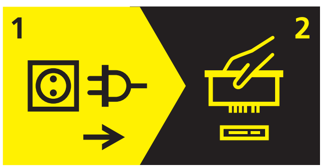
If an I5 Flash card with an I5 software is no longer available, you need to follow the steps described below:

1. 2065 system: Change the Flash card and RAM card (see "Replacing the Flash Card", page 961 and "Replacing the RAM Card", page 963).
2. Delete the PBX software: Delete the I6 system software using the HEX rotary switch sequence "1-E-0". (See "Entering rotary Switch sequences using the example of the initialization function", page 935 for the procedure). The error code "EUL" with a flashing "P" is indicated on the 7-segment display.
3. Carry out an emergency upload as indicated in "Emergency Upload of the system software", page 896.
4. Once the system has started up, load the saved configuration data using AIMS under "Restore", and save the data in the PBX with "Upload". The PBX reboots and is then ready for operation with the I5 system software and the familiar system configuration.

Note:
The I5 system software does not support the memory expansion of the "I6 Update Set" and will output the corresponding error messages. The I5 RAM and Flash cards are essential for a successful downgrade.

2.4 Replacing the Flash Card

Hazard:
Flash cards can be damaged by electrical voltage.
Always make sure the card is disconnected from the power supply before replacing it.



haz1254aaxxa0

Fig. 8.1: Warning plate on the PBX

Warning:
Flash cards can be damaged by electrostatic discharge (ESD).
Always touch the earthed metal cage of PBX before touching a Flash card.

Procedure:

1. If necessary carry out the preparations as described in "System Upgrade 2065", page 959 and "Updating the PBX boot software", page 899.
2. Switch off the PBX. When using a UPS disconnect the battery first, then disconnect the PBX and any external Auxiliary Terminal Power Supply (ATPS) from the 230 V mains.
3. Dismantle the housing cover, then the cover to the power supply area.
4. Before handling the Flash card, touch the earthed metal cage of the PBX.
5. Remove the old Flash card from the PBX.
6. Take the new Flash card out of its ESD protective sleeve.
7. Fit the Flash card into the same system slot on the PBX (Fig. 8.2). Make sure that the side notch on the connector tab is pointing towards the socket outlet.
8. Wrap the Flash card in the ESD protective sleeve.
9. Fit the cover on to the power supply area
10. Fit the housing cover.
11. Connect the PBX and any ATPS system to the 230 V mains, then connect the UPS battery where applicable.

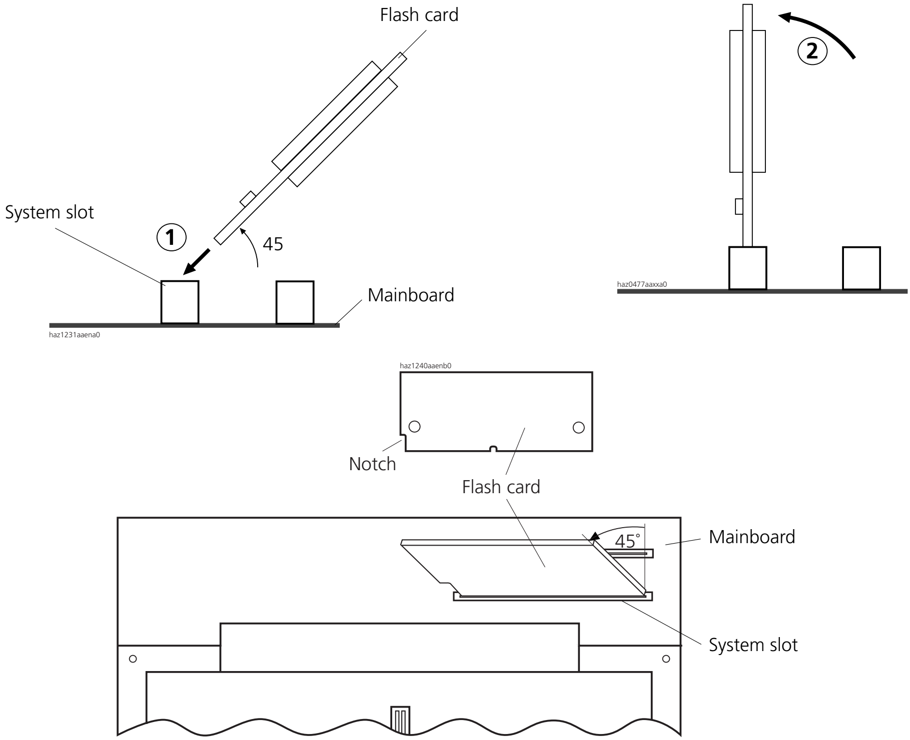


Fig. 8.2: Fitting the Flash card

2.5 Replacing the RAM Card

Required tool: Torx screwdriver, size Torx T 10

Hazard:
The processor card and/or RAM card can be damaged by electrical voltage. Always make sure the cards are disconnected from the power supply before replacing them.

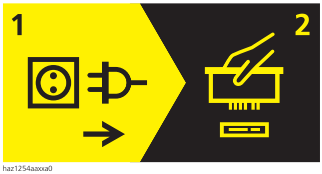
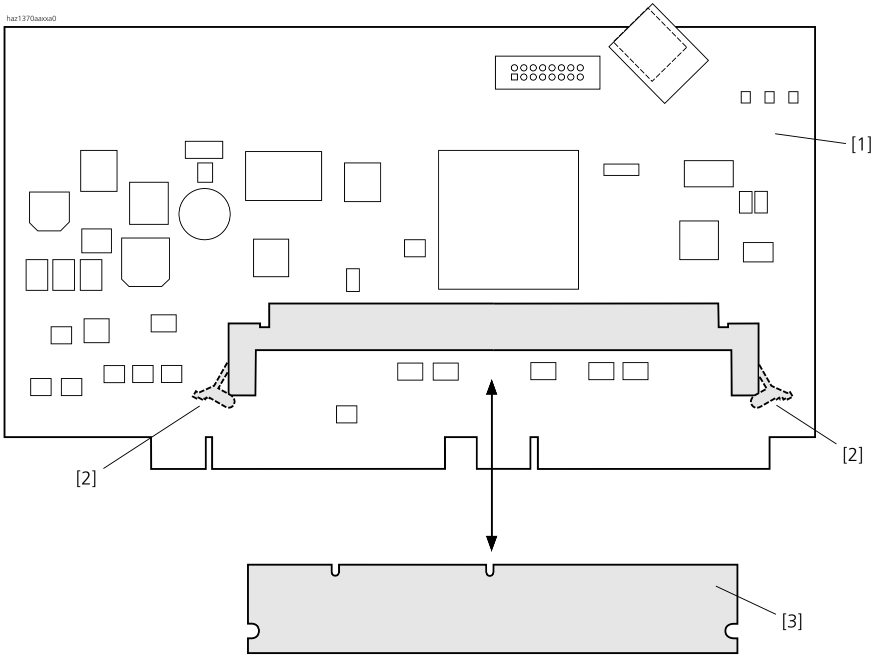


Fig. 8.3: Warning plate on the PBX

Warning:
The processor card and/or RAM card can be damaged by electrostatic discharge. Always touch the earthed metal cage of PBX before touching the cards!

Procedure

1. If necessary carry out the preparations as described in the Chapter "System Upgrade 2065", page 959.
2. Switch off the PBX. When using a UPS disconnect the battery first, then disconnect the PBX and any external Auxiliary Terminal Power Supply (ATPS) from the 230 V mains.
3. Remove the PBX cover.
4. Unscrew the screws on the screening strip and pull out the processor card complete with screening strip.
5. Press the side locks on the RAM card outwards and pull out the RAM card (see Fig. 8.4).
6. Take the new RAM card out of the ESD protective sleeve and insert it. The notches on the RAM card prevent it from being inserted incorrectly.
7. Press the side locks inwards until they click into place.
8. Wrap the RAM card in the ESD protective sleeve.
9. Insert the processor card. Make sure the card is lying in the lower guide slot of the metal cage and that the ends of the screening strip are flush against the edge of the metal cage once the card has been inserted.
10. Screw the screening strip to the metal cage using the 2 screws.
11. Fit the housing cover.
12. Connect the PBX and any ATPS system to the 230 V mains, then connect the UPS battery where applicable.



[1] Processor card
[2] Base locks
[3] RAM card

Fig. 8.4: RAM card on the processor card

2.6 Functions and Products No Longer Supported

The following functions and products are no longer supported compared with version 4.6:

- 65 Volt AC ringing voltage
- 40 Volt DC (socket for powering external equipment)
- House key on the Operator Console
- Digital emergency circuit (replaced by UPS module)
- Analogue emergency relay
- Parallel interface for pager systems
- bcs cordless system
- CTO terminals (Crystal[1]), Topaz, Opal)

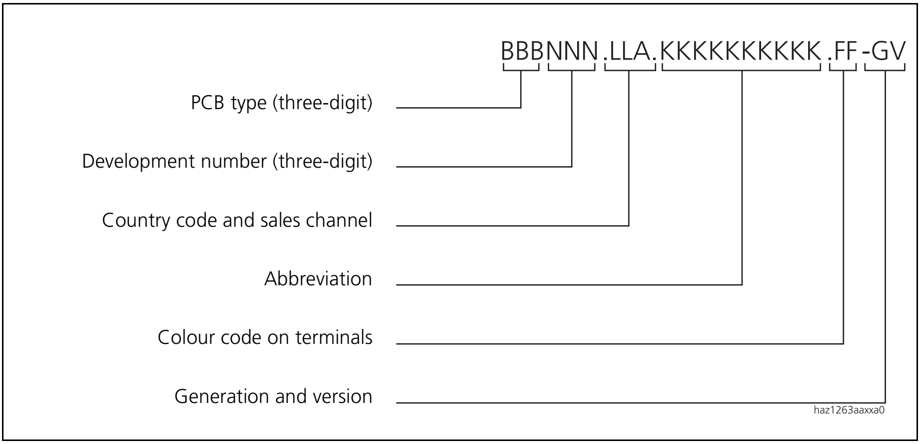
- V2 protocol on S interface
  (now only supported for PC Operator and teleCOURIER 900)
- Direct access with V.110 communication protocol
- Xmodem communication protocol (now only emergency upload via V.24)

The I6 generation supports all the I5 functions and products.

1) For transferring Crystal data, see page 972.

3 Systematic Designation System

Tab. 8.1: PCB Designation



Tab. 8.2: Explanation of the PCB Designation

Part of the PCB designation	Remarks and examples
PCB type (three-digit)	LPB = Printed circuit board fitted KAB = Cable fitted PBX = Complete system SEV = Set packed EGV = Terminal packed
Project number (three-digit)	955 (System 2025/45/65)
Country code and sales channel (one to three-digit, with full stops)	Two-digit country code as per ISO 3166, Sales channel (1...9) for various sales channels, Examples: AT = Austria BE = Belgium CH = Switzerland SWISSCOM CH2 = Switzerland private market DE = Germany EXP = Export channels (not country-specific) Space = No country code
Abbreviation	SC-08AD2 = subscriber card with 8 AD2 interfaces ISDN-02PRA = ISDN card with 2 primary rate accesses
Colour code on terminals	Colour designation in accordance with EU directive
Generation and version	Example: –3C = 3rd generation, version C New PCBs: –1

3.1 Equipment Overview

Tab. 8.3: Equipment Overview

Designation	Description
KAB955 16pin.3m	16-wire cable, 3 m
KAB955 16pin.6m	16-wire cable, 6 m
KAB955 16pin.20m	16-wire cable, 20 m
SEV955 BAD1	Dummy cover 1 BE
SEV955 BAD3	Dummy cover 3 BE
SEV955 BAD5	Dummy cover 5 BE
SEV955 RMS19"	Mounting set, 19" system cabinet
SEV955 KLS20	WAGO terminal set
LPB955 SC-04AD2-x	AD2 card with 4 interfaces
LPB955 SC-08AD2-x	AD2 card with 8 interfaces
LPB955 SC-16AD2-x	AD2 card with 16 interfaces
LPB955 SC-24AD2-x	AD2 card with 24 interfaces
LPB955 SC-32AD2-x	AD2 card with 32 interfaces
LPB955 SC-04AB-x	Analogue subscriber card with 4 interfaces
LPB955 SC-08AB-x	Analogue subscriber card with 8 interfaces
LPB955 SC-16AB-x	Analogue subscriber card with 16 interfaces
LPB955 ISDN-04ST-x	ISDN card with 4 basic access interfaces T and S
LPB955 ISDN-08ST-x	ISDN card with 8 basic access interfaces T and S
LPB955 ISDN-01PRA-x	ISDN card with 1 primary rate access interface T2
LPB955 ISDN-02PRA-x	ISDN card with 2 primary rate access interfaces T2
LPB955 TC-04AB-x	Analogue trunk card with 4 interfaces
LPB955 IPI-6400-x	IP Interface Card AIP-6400
LPB955 IPI-6350-x	IP Interface Card AIP-6350
LPB955 DSP-01-x	DSP card 12 DECT channels
LPB955 DSP-02-x	DSP card; 18 DECT channels, 6 full duplex hands-free channels
LPB955 DSP-04-x	DSP card; 36 DECT channels, 9 full duplex hands-free channels
LPB955 VM-02P-x	Card for Voice Mail, 2 ports
LPB955 VM-04P-x	Card for Voice Mail, 4 ports
LPB955 OI-2DOOR-x	Special card for announcement equipment or Door intercom system
LPB955 UPS-12V-x	12 Vdc uninterruptible power supply module
LPB955 DC-48V-x	Module for 48 Vdc power supply
PBX955 2025 / 2045-x	2025 / 2045 basic system

Designation	Description
PBX955 2065-x	2065 basic system
PBX955 2025R / 2045R-x PBX955 2065R-x	Basic system 2025 / 2045, rack version Basic system 2065, rack version

3.2 Rating Plate and Designation Stickers

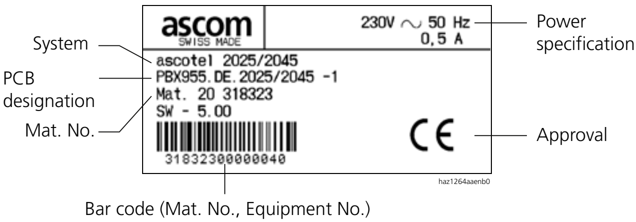


Fig. 8.5: Rating plate outside the housing on the power-cord side

Generational and Version Change:

- A generational change is effected following substantial changes to the functionality of a PCB.
- A change of version is effected following small changes to functions or once faults have been remedied. Backward compatibility is guaranteed.

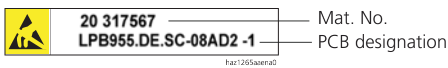


Fig. 8.6: Designation sticker on expansion card

4 DECT Compatibility

4.1 GAP functionality

The following table contains the network features as defined in the GAP standard. For each feature it indicates in a separate column whether it is supported by Ascotel IntelliGate and Office 100 / Office 135 / Office 150 / Office 155pro.

Tab. 8.4: Features supported as per GAP standard

No.	Feature	PP	In Office 100 / Office 135 / Office 150 / Office 155pro	FP	In Ascotel IntelliGate
1	Outgoing call	M	✓	M	✓
2	Off hook	M	✓	M	✓
3	On hook (full release)	M	✓	M	✓
4	Dialled digits (basic)	M	✓	M	✓
5	Register recall	M	✓	O	✓
6	Go to DTMF signalling (defined tone length)	M	✓	O	✓
7	Pause (dialling pause)	M	✓	O	—
8	Incoming call	M	✓	M	✓
9	Authentication of PP	M	✓	O	✓
10	Authentication of user	M	✓	O	—
11	Location registration	M	✓	O	✓
12	On air key allocation	M	✓	O	✓
13	Identification of PP	M	✓	O	—
14	Service class indication / assignment	M	✓	O	—
15	Alerting	M	✓	M	✓
16	ZAP	M	✓	O	—
17	Encryption activation FP initiated	M	✓	O	—
18	Subscription registration procedure on-air	M	✓	M	✓
19	Link control	M	✓	M	✓
20	Terminate access rights FP initiated	M	✓	O	✓
21	Partial release	O	✓	O	✓
22	Go to DTMF (infinite tone length)	O	—	O	—
23	Go to Pulse	O	—	O	—
24	Signalling of display characters	O	✓	O	—
25	Display control characters	O	—	O	—
26	Authentication of FP	O	✓	O	✓

No.	Feature	PP	In Office 100 / Office 135 / Office 150 / Office 155pro	FP	In Ascotel Intelli-Gate
27	Encryption activation PP initiated	O	—	O	—
28	Encryption deactivation FP initiated	O	—	O	—
29	Encryption deactivation PP initiated	O	—	O	—
30	Calling Line Identification Presentation (CLIP)	O	✓	O	✓
31	Internal Call	O	✓	O	—
32	Service Call	O	—	O	—

PP: Portable Part
FP: Fixed Part
M: Mandatory (this feature must be supported by GAP-compliant equipment)
O: Optional
—: Office 100 / Office 135 / Office 150 / Office 155pro and Ascotel IntelliGate do not support this feature.

4.2 Radio units and handsets

Tab. 8.5: Compability of radio units and handsets

Devices	can be used as of / on	Restrictions
SB-4	I4 V4.30	Does not support the Zero-Blind-Slot mode; therefore only 4 simultaneous call connections possible per radio unit.
SB-8	I4 V4.30	• On < I6.1 systems the SB-8 responds in the same way as an SB-4, i.e. it is connected via an AD2 bus only. • Only 4 simultaneous call connections possible per radio unit.
SB-8ANT	I4 V4.30	• As for SB-8 • Plus: Operation with external antennas is not immediately possible on < I6.1 systems (installation of a hardware switch required)
Office 100, Office 150	SB-4, SB-8, SB-8ANT	• Does not support the Zero-Blind-Slot mode; therefore only 4 simultaneous call connections possible per radio unit. • The entire DECT system has to operate in Blind-Slot mode (see Tab. 5.9).
Office 135/135pro, Office 155pro	SB-4, SB-8, SB-8ANT	

5 Taking Over a Crystal Phone Book

The data in the Crystal private phone book can be copied into the private telephone book of any corded or cordless Office terminal.

If the private telephone book on the Office terminal is smaller than the Crystal private phone book, the number of entries exceeding capacity are not copied and are lost.

Copying overwrites all data present in private telephone books of the Office terminal.

Saving Crystal Data to a PC

1. Connect Crystal V.24 to PC.
2. Set transmission rate on Crystal.
3. Start Hyperterminal on PC and set transmission rate.
4. Initiate ”Record text file” command.
5. Activate ”Export private file” on Crystal.

The private file is saved as a TXT file on the PC.

Convert data

1. On the PC initiate the file ”CrystalPrivateDataconverter.xls” located in the AIMS directory.
2. Activate macros.
3. Enter the interface number and MSN of the Office terminal.
4. Start ”Import” and select the TXT file.
5. Initiate the ”Convert” command.

The file "OfficeData.csv" is created.

Import data into AIMS

1. Start Configuration Manager.
2. Import the "OfficeData.cvs" file.
3. Load data onto PBX.

6 Technical Data

6.1 Basic systems

Tab. 8.6: Dimensions and weights of the basic systems

	2025 / 2045 basic system	2065 basic system	2025R / 2045R basic system	2065R basic system
Height	460 mm	460 mm	222 mm	
Width	320 mm	445 mm	485 mm	
Depth	187 mm	187 mm	515 mm	
Weight (excl. mains cord, expansion cards and packaging)	5.3 kg	6.9 kg	15.8 kg	17.4 kg

Tab. 8.7: Electrical isolation of interfaces

Interface	all systems	
Analogue network interfaces	0.2 kV	Operating isolation
Digital network interfaces PRA	1.5 kV	
Digital network interfaces BA	65 V	Operating isolation
V.24 on the Pocket Adapter	0.5 kV	
Control inputs interface switch group 1		no isolation, but input impedance > 8 kΩ
Freely connectable relays on OI-2DOOR	0.5 kV	
Door intercom interface on OI-2DOOR	0.5 kV	
Ethernet	1.5 kV	
DC power supply UPS-12V (batt. connection)	50 V	Operating isolation
DC power supply DC-48V	0.5 kV	

Tab. 8.8: Ambient conditions

Condition	all systems
Ambient temperature	5 °C to 35 °C
Relative air humidity	30% to 80%, non-condensating

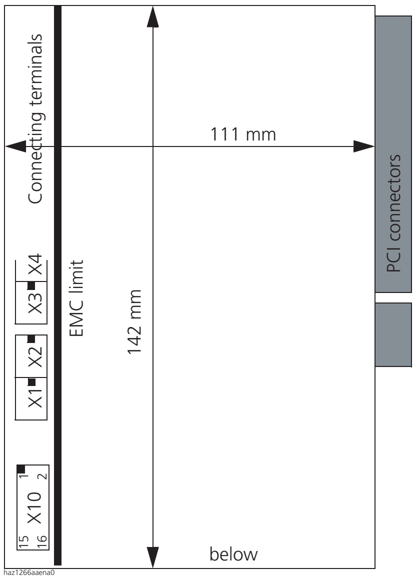
Tab. 8.9: Electrical data, basic systems

	Basic system 2025 / 2045(R)	Basic system 2065(R)
Class of protection	1	
Input voltage	230 VAC (195 V…253 V, 48…62 Hz)	
Input current	0.5 A	1.0 A
Mains fuse-unit	630 mAT G	1.6 AT G
Resistant to:		
• Voltage breaks	< 30 ms	
• Voltage dip to < 80%	< 30 ms	
Power input with min. configuration	approx. 14 W, 22 VA	approx. 20 W, 31 VA
Power input with max. configuration	approx. 70 W, 110 VA	approx. 135 W, 210 VA
Undervoltage limit (system reset, data backup)	< 190 V	

Tab. 8.10: Head dissipation, PBX and ATPS

	Basic system 2025 / 2045(R)	Basic system 2065(R)
Minimally expanded system without ATPS	approx. 60 W = 220 kJ/h	approx. 100 W = 360 kJ/h
Fully expanded system with ATPS	approx. 100 W = 360 kJ/h	approx. 160 W = 580 kJ/h

6.2 Expansion cards



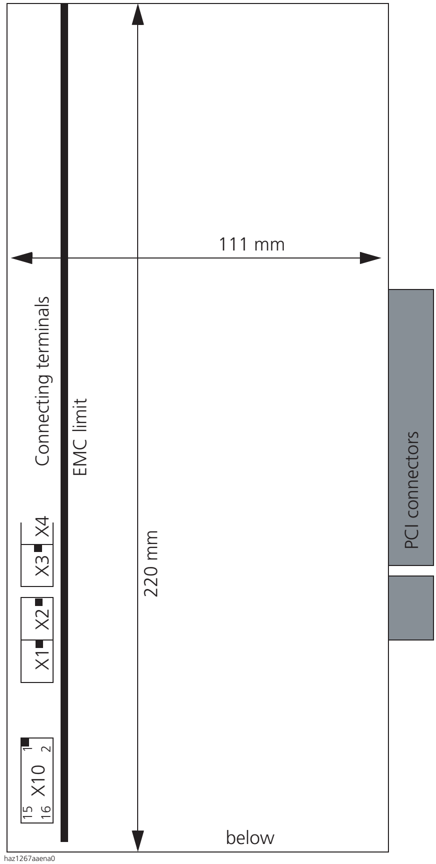


Fig. 8.7: Dimensions, expansion card, type A (left) and type B (right)

Tab. 8.11: Type and weights of the expansion cards

Expansion card	Type (A/B)	Weight
DSP-01 / -02 / -04 ISDN-04ST / -01PRA / -02PRA SC-04AD2 / -08AD2	A	approx. 150 g
ISDN-08ST OI-2DOOR AIP -6400 AIP -6350 SC-16AD2 / -24AD2 / -32AD2 SC-04AB / -08AB / -16AB TC-04AB VM-02P / -04P	B	approx. 240 g

6.3 Power supplies

Tab. 8.12: Technical data, UPS-12V module

	2025 / 2045 basic system	2065 basic system
Input voltage	12 VDC (10 V…15 V)	
Input current	max. 8.2 A with 12 V	max. 15.8 A with 12 V
Fuse cartridge	20 A (Wickmann FK1 20 A)	
Floating charging voltage	13.5 V ± 2%	
Charge current limiting	1.0 A or 2.0 A (adjustable)	
Battery cable	4 mm^2, max. 7 m	4 mm^2, max. 3,5 m
Weight	520 g	
Dimensions	278 x 98 x 40 mm	

Tab. 8.13: Technical data, DC-48V module

	2025 / 2045 basic system	2065 basic system
Input voltage	48 VDC (43 V…58 V)	
Input current	max. 2.0 A with 48 V	max. 3,8 A with 48 V
Fuse cartridge	5 A (Wickmann FUN125 5 A)	
Power supply cable	2.5 mm^2, max. 60 m	2.5 mm^2, max. 30 m
Weight	450 g	
Dimensions	278 x 98 x 40 mm	

Tab. 8.14: Technical data, Auxiliary Terminal Power Supply (ATPS)

	Auxiliary Terminal Power Supply (ATPS)
Input voltage (mains)	90 - 254 V, 47 - 63 Hz
Input current	max. 2 A with 230 V
Output voltage	40 +/-1 V
Max. power output	250 W
Cables supplied	• DC connecting cable to the PBX (1.2 m, permanently fitted) • 230 V ATPS mains cord to the PBX, (2 m, pluggable)
Weight	1630 g
Dimensions (excl. mains cord)	307 x 130 x 75 mm

6.4 System Terminals

Tab. 8.15: Accessories for system terminals

Accessories	**Office 1600IP**	**Office 45**	**Office 45pro**	**Office 35IP**	**Office 35**	**Office 25**	**Office 40**	**Office 30**	**Office 20**	**Office 10**	**Office 135**	**Office 135pro**	**Office 155pro**	**Office 150**	**Office 100**
Pocket Adapter	–	✓	✓	–	✓	✓	✓	✓	✓	✓	–	–	–	–	–
Additional keypad ZTF	–	–	–	–	–	–	1	1	–	–	–	–	–	–	–
Expansion keypad EKP	–	2/3[1)]	2/3[1)]	2/3[1)]	2/3[1)]	–	–	–	–	–	–	–	–	–	–
Alpha keyboard AKB	–	✓	✓	✓	✓	–	–	–	–	–	–	–	–	–	–
Headset	✓	✓	✓	✓	✓	✓	✓	✓	✓	–	–	✓	✓	✓	–
Plug-in power supply for direct charging	–	–	–	–	–	–	–	–	–	–	–	✓	–	–	–

1) 2 with alpha keyboard, 3 without alpha keyboard

Tab. 8.16: Ascotel DECT

Duplex method	Time-division multiplex, 10 ms frame length
Frequency range	1880 MHz to 1900 MHz
Frequency bands (carrier)	10
Channel spacing (carrier distance)	1,728 MHz
Transmission rate	1152 kbit/s
Duplex channels per carrier SB-4 / SB-8	6 / 12
Number of channels (duplex channels) SB-4 / SB-8	60 / 120
Modulation	GFSK
Data transfer rate	32 kbit/s
Voice encoding	ADPCM
Transmit power	250 mW peak value 10 mW, average power per channel
Range	30 to 250 m
Max. line length to radio unit - power supply via AD2 bus (0.5mm) SB-4 / SB-8 - with power supply unit (9–15 VDC, 400 mA)	 660 m / 1200 m[1)] 1200 m
Ambient temperature, radio unit in operation:	0 °C to 40 °C
Dimensions: Radio unit W x H x D:	165 x 170 x 70 mm
Weight: Radio unit	320 g
Local power supply to radio unit (optional)	Plug-in power supply unit (Euro-plug)

1) 1200 m with SB-8 on two AD2 interfaces or on one interface with a maximum of 4 active channels

Tab. 8.17: Ascotel DECT handsets

Standard	Office 135	Office 155pro	Office 150	Office 100
• Handset operating time (with fully charged battery) in standby	120 hours	approx. 100 hours	approx. 80 hours	approx. 33 hours
• Handset operating time, in talk time	approx. 12 hours	approx. 10 hours	approx. 8 hours	approx. 5 hours
Batteries	Ni MH battery pack	Ni MH battery pack	Ni MH battery pack	2 x 1.2 V type A-A Ni MH
Power input Charging bay	6 VA	9 VA	9 VA	9 VA
Ambient temperature in operation				
• Charging bay	5 °C to 40 °C	5 °C to 40 °C	10 °C to 40 °C	5 °C to 35 °C
• Handset	0 °C to 40 °C	0 °C to 40 °C	0 °C to 40 °C	0 °C to 45 °C
Admissible storage temperature	-25 ° to 45 °C	-25 ° to +70°C	-25 ° to +75 °C	-10 ° to +60 °C

Tab. 8.18: Dimensions and weights for system terminals Office 40, Office 30, Office 20, Office 10, Office ZTF and Office Pocket Adapter

System terminal	Height	Width	Depth	Weight
Office 40	94 mm	278 mm	233 mm	approx. 935 g
Office 30	94 mm	213 mm	233 mm	approx. 740 g
Office 20	84 mm	213 mm	216 mm	approx. 640 g
Office 10	55 mm	82 mm	200 mm	approx. 360 g
Office ZTF	61 mm	91 mm	216 mm	approx. 260 g
Office PA	26 mm	61 mm	121 mm	approx. 180 g

Tab. 8.19: Dimensions and weights for system terminals Office 45, Office 35, Office 25, Office AKB, Office EKP

Terminals on AD2 interface	Height	Width	Depth	Weight
Office 45	97 mm	336 mm	203 mm	approx. 960 g
Office 35	75 mm	254 mm	203 mm	approx. 680 g
Office 25	56 mm	224 mm	203 mm	approx. 500 g
Office EKP	44 mm	82 mm	133 mm	approx. 115 g
Office AKB	21 mm	190 mm	82 mm	approx. 150 g

Tab. 8.20: Dimensions and weight of Ascotel DECT terminals

	Handset		Charging bay	
	H x W x D	Weight	H x W x D	Weight
Office 100	35 x 52 x 162 mm	165 g	37 x 129 x 180 mm	250 g
Office 135	23 x 49 x 138 mm	130 g	70 x 82 x 160 mm	105 g
Office 150 / Office 155pro	30 x 56 x 142 mm	195 g	68 x 72 x 105 mm	105 g

6.5 Network Interfaces

The following technical data applies to the network interfaces:

Basic Access T

- Standard Euro ISDN interface as per CTR-3
- Configurable for point-to-point or point-to-multipoint operation

Primary rate access T2

- Standard Euro ISDN interface as per CTR-4
- Can be used as a network interface (T2)

Analogue network interfaces

- Voice path with A/D and D/A conversion (standard PCM, A-law)
- Transmission as per ES 201 168 (level country-specific)
- Signalling as per TBR 21
- Pulse or DTMF dialling, Flash signal
- Loop current detection
- Call charge receive 12 or 16 kHz (frequency and level setting country-specific)

6.6 User-network interfaces

The following technical data applies to the user-network interfaces:

Digital user-network interface AD2

- Proprietary interface, two-wire
- 2 system terminals per interface
- Power supply min. 75 mA, limiting at approx. 80 mA, terminal voltage 36...41 V
- Line termination in the terminal
- Transparent transmission of 2 PCM channels

Digital user-network interface S

- Standard Euro ISDN interface
- Phantom power supply min. 140 mA, limiting at approx. 170 mA, terminal voltage 36...41 V

Analogue user-network interface a/b

- Voice path with A/D and D/A conversion (standard PCM, A-law)
- Transmission as per ES 201 168 (level country-specific)
- Constant current loop supply approx. 25 mA (with loop resistances ≤ 1000 Ω)
- No-load voltage, interface 37...40 V
- Receive pulse or DTMF dialling
- Ringing supply 40...43 V 50 Hz at load 4 kΩ; no DC voltage overlay (country-specific versions also with 25 Hz)
- No control key detection
- No charge signalling pulses

7 PC Dial commands

Tab. 8.21: Activating the PC Dial application

ATPC 1 <cr>	Activates the PC Dial mode
ATPC0 <cr>	Deactivates the PC Dial mode

Tab. 8.22: Dial command

ATD nnn...<cr>	Writes nnn... on the terminal display
ATDT nnn...<cr>	Equivalent to ATD
ATDP nnn...<cr>	Equivalent to ATD

If the number contains macro characters, use the "/" character to separate it from "ATD". The "@" character is the same as "/A".

Tab. 8.23: Examples of dial command

ATD@ 351 <cr>	Seizes and dials the number 351
ATD/|*21205 PX/<cr>	Seizes internally and initiates a Call Forwarding Unconditional

Tab. 8.24: Signalling an incoming call

CALL V FROM nnnn <cr>	The PC Dial application displays a call with
CALL V FROM name/nnn <cr>	one of the following messages,
CALL V FROM name/<cr>	depending on whether name or number
CALL V FROM <cr>	information is available.

Tab. 8.25: Other commands

ATA <cr>	Answers call with hands-free mode
ATH <cr>	Terminates call connection (goes on-hook)
ATE1 <cr>	Activates echo
ATEO <cr>	Deactivates echo
ATH?	Enquires about connection status

Tab. 8.26: Status reports in response to ATH

IDLE <cr>	The terminal is idle (free)
DIALLING <cr>	The terminal is sending dialling pulses
CONNECT <cr>	The connection is set up
RING <cr>	The terminal is being called
CALLING <cr>	The terminal is receiving the ring-back tone
TRANSPARENT MODE <cr>	The terminal is sending voice-frequency signals

BUSY <cr>	Busy
DISCONNECT <cr>	The connection is being disconnected
OK <cr>	Response to all other valid commands

8 Operation of the system terminals

8.1 Overview of control elements

Tab. 8.27: Overview of control elements

Control elements	Office 45/45pro	Office 35/35IP	Office 25	Office 40	Office 30	Office 20	Office 10	Office 135/135pro	Office 155pro	Office 100	Office 150
Function keys:											
• Foxkey 🦊	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
• Menu key	✓	✓	✓	✓	✓	–	–	✓	✓	✓	✓
• Display keys	✓	–	–	–	–	–	–	–	–	–	–
• Cursor key	✓	–	–	–	–	–	–	–	–	–	–
• Info key	✓	✓	✓	🦊	🦊	–	–	✓	✓	✓	✓
• Absence key	✓	✓	(✓)	✓	✓	✓	–	–	–	–	–
• Alpha key	✓	✓	✓	✓	✓		–	🦊	🦊	🦊	🦊
• Code key	✓	🦊	🦊	✓	🦊	🦊	–	🦊	🦊	🦊 .	🦊
• Setting keys (+, -)	✓	✓	(✓)	✓	✓	(✓)	–	✓	✓	–	✓
• Phone book key	✓	–	–	✓	–	–	–	–	–	–	–
• Correction key)	✓	✓	(✓)	✓	✓	–	–	✓	✓	✓	✓
• Loudspeaker key	✓	✓	(✓)	✓	✓	✓	–	✓	✓	–	✓
• MIC key	✓	🦊	–	✓	✓	–	–	🦊	🦊	🦊	🦊
• Notepad key	✓	–	–	✓	✓	–	–	–	–	–	–
• END key	✓	✓	(✓)	✓	✓	✓	✓	✓	✓	✓	✓
• Redial key	✓ [10]	✓ [10]	✓ [4]	✓ [10]	✓ [4]	✓ [4]	✓ [1]	✓ [10]	✓ [10]	✓ [10]	✓ [10]
Freely configurable keys:											
• with LED	10	5		12	6	–	–	–	–	–	–
• without LED / Hotkey	–	–	4	–	–	1	3	1 (6)	1 (6)	1 (6)	1 (6)
Charging bay contact	–	–	–	–	–	–	–	✓	✓	✓	✓
Twin functions:											
• Twin Mode	–	–	–	–	–	–	–	✓	✓	✓	✓
• Twin Comfort	–	–	–	–	–	–	–	✓	✓	✓	✓
Trembler (trembler model only)	–	–	–	–	–	–	–	✓[1)]	✓	–	(✓)
Illuminated keypad	–	–	–	–	–	–	–	✓[1)]	–	–	–

Control elements	**Office 45/45pro**	**Office 35/35IP**	**Office 25**	**Office 40**	**Office 30**	**Office 20**	**Office 10**	**Office 135/135pro**	**Office 155pro**	**Office 100**	**Office 150**
Key telephone:											
• Line keys (LT)	9	4	–	12	5	–	–	–	–	–	–
• LT with additional keypad (ZTF)	–	–	–	32	25	–	–	–	–	–	–
• LT with expansion keypad(s) (EKP)	39	34	–	–	–	–	–	–	–	–	–
• Internal key	✓	✓	–	✓	✓	–	–	–	–	–	–
Operator Console:											
• Line Keys	5 / 9	–	–	–	–	–	–	–	–	–	–
• Personal key	1	–	–	–	–	–	–	–	–	–	–
Display elements:											
• Alphanumeric display	✓	✓	✓	✓	✓	✓[2]	–	✓	✓	✓	✓
– backlit	✓[3]	–	–	–	–	–	–	✓	✓	–	✓
• Attention LED	✓	✓	✓	✓	✓	–	✓	✓	✓	–	✓

1) only Office 135pro

2) num.

3) Office 45pro only

8.2 Digit key assignment on system terminals

Digit key assignment depends on the family of system terminals and the language set for the PBX.

The following Latin script assignment for the digit keys applies to the system terminals Office 35 / Office 45 / Office 135 and Office 155pro for all PBX languages with the exception of Greek:

Tab. 8.28: Latin-script digit key assignment

1	- . ? 1 ! , : ; ’ " ¿ ¡ - . ? 1 ! , : ; ’ " ¿ ¡	2 ABC	A B C 2 Ä Æ Å Ç a b c 2 ä æ å à ç
3 DEF	D E F 3 É d e f 3 é è ê	4 GHI	G H I 4 g h i 4 ì
5 JKL	J K L 5 j k l 5	6 MNO	M N O 6 Ñ Ö Ø m n o 6 ñ ö ø ò
7 PQRS	P Q R S 7 p q r s 7 ß	8 TUV	T U V 8 Ü t u v 8 ü ù
9 WXYZ	W X Y Z 9 w x y z 9	0	+ 0 + 0
*	* / ( ) < = > % £ $ € ¥ ¤ @ & § * / ( ) < = > % £ $ € ¥ ¤ @ & §	#	Space # Space #

The Office 25 terminal does not have an alphanumeric display and therefore cannot display all the characters featured (see also the Operating Instructions for the Office 25).

The following Greek-script assignment for the digit keys applies to the system terminals Office 35 / Office 45 / Office 135 and Office 155pro if the PBX language is set to Greek. Greek letters are always displayed in upper case on the terminal displays:

Tab. 8.29: Greek-script digit key assignment

Key	Assignment	Key	Assignment
1	- . ? 1 ! , : ; ' " - . ? 1 ! , : ; ' "	2 ABC	Α Β Γ 2 A B C Α Β Γ 2 a b c
3 DEF	Δ Ε Ζ 3 D E F Δ Ε Ζ 3 d e f	4 GHI	Η Θ Ι 4 G H I Η Θ Ι 4 g h i
5 JKL	Κ Λ Μ 5 J K L Κ Λ Μ 5 j k l	6 MNO	Ν Ξ Ο 6 M N O Ν Ξ Ο 6 m n o
7 PQRS	Π Ρ Σ 7 P Q R S Π Ρ Σ 7 p q r s	8 TUV	Τ Υ Φ 8 T U V Τ Υ Φ 8 t u v
9 WXYZ	Χ Ψ Ω 9 W X Y Z Χ Ψ Ω 9 w x y z	0	+ 0 + 0
*	* / ( ) < = > % £ $ € ¥ ¤ @ & § * / ( ) < = > % £ $ € ¥ ¤ @ & §	#	Space # Space #

The Office 25 terminal does not have an alphanumeric display and therefore cannot display all the characters featured (see also the Operating Instructions for the Office 25).

Note:
If only the language of the terminal and not the PBX language is set to Greek, only the static and dynamic menus will appear in Greek letters on the terminal. In such cases it is not possible to key in Greek letters or to edit texts in Greek letters (e.g. run alpha dialling, edit private phone book, etc.)

8.3 Alpha keyboard (AKB)

The alpha keyboard for Office 35 and Office 45 is available in 2 variants, which differ in the keypad printing.

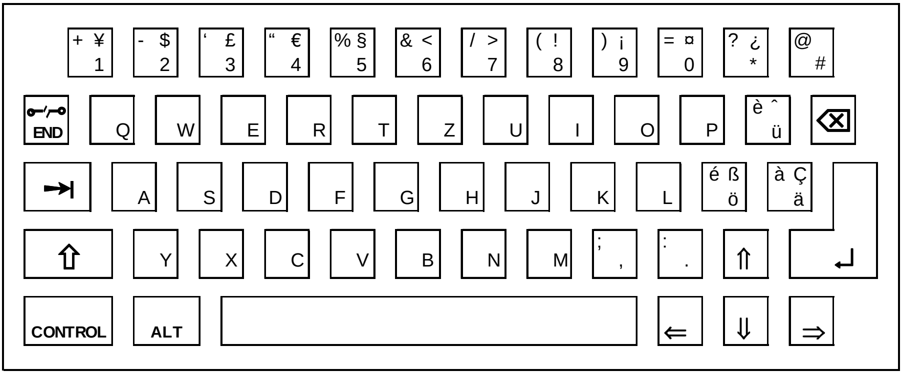


Fig. 8.8: **AKB QWERTZ**

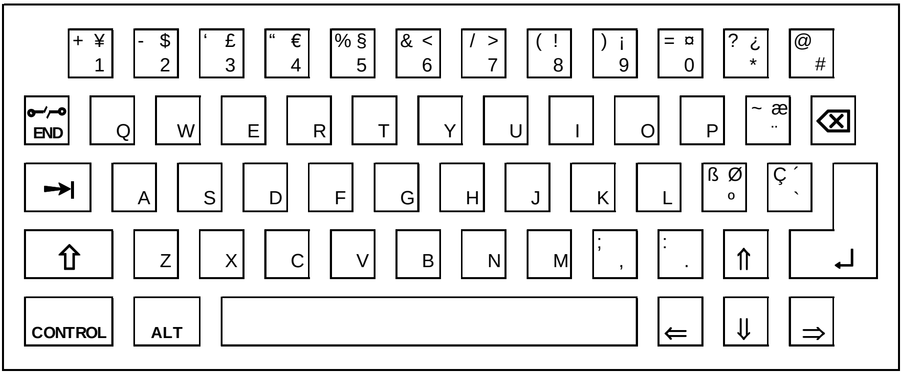


Fig. 8.9: **AKB QWERTY**

If an alpha keyboard (QWERTZ or QUERTY) is connected to a PBX that is set to Greek, the following keyboard assignment applies:

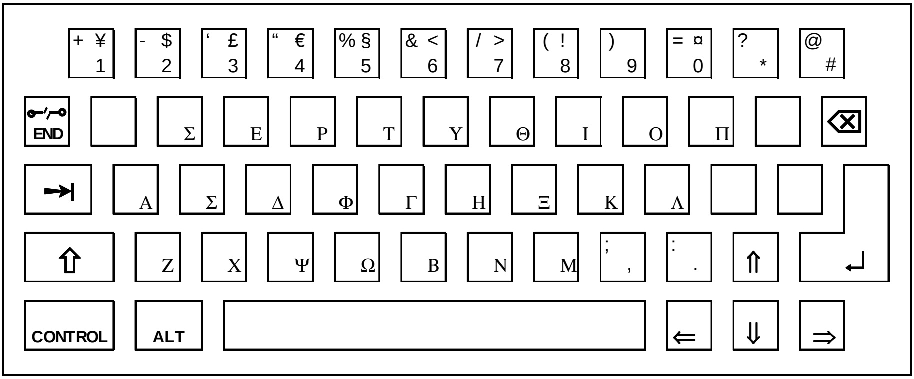


Fig. 8.10: Greek keyboard assignment

Note:
The keyboard can be switched from Greek to Latin using "Alt Tab" subject to certain restrictions with regard to special characters. This means that Latin characters can be keyed in via the AKB even though the PBX language is set to Greek.

8.4 Function commands (macros)

Function commands are used mainly for automatically activating / deactivating features using the function keys of the system terminals. The following function commands are available:

Tab. 8.30: Function commands for system terminals

Function command	Meaning
"A"	Seize line with maximum priority[1)]
"I"	Seize line
"X"	Disconnect
"P"	Pause 1 second before next action
"Lxx"	Seize line xx (line keys)[1)]
"N"	Enter call number keyed in during call preparation
"."	Control keys function

Function command	Meaning
"Z"	Activate / deactivate DTMF mode (tone dialling)
"R"	Use call number last dialled
"Y"	End call and reseize line [2]

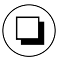
1) Available only with the Operator Console.

2) Not available for Office 10.

The function commands can be stored directly on the system terminals or on the function keys via AIMS.

Note:
As the Office 10 does not have a text mode, only 3 function commands can be stored on function keys on this terminal. The 3 function commands are entered using the following keys:

Tab. 8.31: Function commands on function keys Office 10

	Pause 1 second before next action
	Control keys function
	Activate / deactivate DTMF mode (tone dialling)

9 Additional documentation

The following documents can be found under http://www1.aastra.com/docfinder

- Environmental information for basic systems and system terminals
- Declarations of conformity for basic systems and system terminals
- Labels for system terminals and expansion keypad
- Safety instructions for system terminals
- Product information

Part 9 Abbreviations, Glossary and Index

1 Abbreviations

a/b	Analogue
ACD	Automatic Call Distribution
AD2	Ascotel digital two-wire
AIMS	Ascotel Information Management System
AIP	Ascotel IP module
AKB	Alphanumerical Keyboard
AMI	Ascotel Mobility Interface
ATPS	Auxiliary Terminal Power Supply
AVS	Ascotel Voice Mail System
BA	Basic access
BHCA	Busy Hour Call Attempts
CAPI	Common ISDN Application Interface
CC	Country Code
CCBS	Completion of Calls to Busy Subscriber
CCNR	Completion of Calls on No Reply
CD	Call Deflection
CDE	Call Distribution Element
CFB	Call Forwarding Busy
CFNR	Call Forwarding No Reply
CFU	Call Forwarding Unconditional
CID	Channel Identification
CL	Call Logging
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CNIP	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction
COLP	Connected Line Identification Presentation

COLR	Connected Line Identification Restriction
CONF	Add-on Conference
CONP	Connected Name Identification Presentation
CONR	Connected Name Identification Restriction
CT	Call Transfer
CTI	Computer Telephony Integration
CTX	Centrex
CW	Call Waiting
DAY	Subscriber Line Group
DDI	Direct Dialling In
DDO	Direct Dialling Out
DECT	Digital Enhanced Cordless Telephony
DSS1	Digital Subscriber Signalling 1
DTE	Data Terminal Equipment
DTMF	Dual Tone Multi-Frequency
ECMA	European Computer Manufacturers' Association
ECT	Explicit Call Transfer
EID	Equipment Identification
EIM	Equipment Identification Module
EKP	Expansion keypad
EMC	Electromagnetic compatibility
ERC	External Remote Control
ETSI	European Telecommunications Standards Institute
FP	Featurephone
GAP	Generic Access Profile
GB	General Bell
HA	Handset (cordless)
HL	Hotline

ICC	Individual Charge Counting
ICL	Incoming Call Logging
IMP	Pulse dialling
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISO	International Standardisation Organisation
ISP	Internet Service Provider
ITU	International Telecommunication Union
KT	Key telephone
LAN	Local Area Network
LCD	Liquid Crystal Display
LCR	Least Cost Routing
LED	Light Emitting Diode
LIC	Licence Code
MBL	Mainboard Large (2065)
MBS	Mainboard Small (2025 / 2045)
MDB	Main Distribution Board
MSN	Multiple Subscriber Number
NDC	National Destination Code
NP	National prefix
NPI	Numbering Plan Identifier
NT	Network Terminator
OC	Operator Console
OCL	Outgoing Call Logging
OIP	Open Interfaces Platform
PA	Pocket Adapter
PARE	Partial Rerouting
PBX	Private Branch Exchange
PBX SC	PBX subscriber
PCC	Preset Conference Calling

PIN	Personal Identification Number
PINX	Private Integrated Network Exchange
PISN	Private Integrated Services Network
PISN SC	PISN Subscriber
P-MP	Point-to-multipoint connection
PNO	Public Network Operator
PNP	Private Numbering Plan
P-P	Point-to-point connection
PPP	Point-to-point protocol
PRA	Primary rate access (30B+D)
PSA	Paging system
PSI	Private Subscription Identity
PSI	Paging System Interface
PSS1	Private Signalling System1
PSTN	Public Switched Telephone Network
QSIG	Signalling at reference point Q as per ITU
RAS	Remote Access Service
RDT	Remote Data Transmission
RE	Recall
RU	Radio unit
S	S user-network interface
SC	Subscriber
Se	Basic access S external
SEM	System Event Manager
SMB	Small and medium-sized businesses
SMTP	Simple Mail Transfer Protocol
SN	Subscriber Number
T	Basic Access T
T2	Primary rate access T2
TA	Terminal Adapter
TAPI	Telephony Application Program Interface
TCP	Transmission Control Protocol
TEI	Terminal Endpoint Identifier

TFE	Door intercom system
TON	Type of Number
TSD	Terminal Selection Digit
TSP	Telephony Service Provider
TSPI	Telephony Service Provider Interface
UG	User group
UPS	Uninterruptible Power Supply
UUS	User-to-User Signalling
V2	Proprietary protocol at the S interface
VAC	Volts of Alternating Current
VDC	Volts of Direct Current
VoIP	Voice over Internet Protocol
VoIP SC	VoIP subscriber
WAN	Wide Area Network
WWW	World Wide Web
ZTF	Additional keypad

2 Glossary

*, 1, 2, 3, ...

*/#	Code used for activating or ending a procedure e. g. *21 to activate a Call Forwarding Unconditional.
2B+D	Channels of the basic access, 2 2 B-channels at 64 kbit/s each plus 1 D-channel at 16 kbit/s
30B+D	Channels of the primary rate access, 30 B-channels at 64 kbit/s each plus 1 D-channel at 64 kbit/s.
3PTY	Three-Party Service (three-party connection)

A

a/b	Analogue network interface or an analogue user-network interface.
Acknowledgement tone	→ Tone signalling in the handset
AD2	Ascotel digital two-wire interface
AIMS	→ Ascotel Information Management System
Alpha dialling	Dials letters via a terminal's keypad.
Alternative Routing	→ Overflow
Ascotel Information Management System (AIMS)	System for setting up, managing and changing configuration and customer data.
Ascotel Voice Mail System (AVS)	System for notifying system terminals about Voice Mails they have received.
Attention tone	→ ringing pattern
Automatic Call Distribution (ACD)	Automatic transfer of a call complete with its customer data to a predefined subscriber; call routing control via the ACD server.
Auxiliary Terminal Power Supply (ATPS)	External Auxiliary Terminal Power Supply

B

B channel	User information channel of an ISDN connection
Backup	Saves data by copying to a different data carrier.
Basic access	(ISDN-)T or S external interface (2B + D)
Break-in	An incoming connection from the outside is routed into the → PISN at the → PINX closest to the caller.
Break-out	An outgoing external connection is routed into the public ISDN only at the PINX that is closest to the call destination.
Busy Hour Call Attempts (BHCA)	Traffic volume during the busy hour.
Busy tone	→ Tone signalling in the handset

C

Call deflection (CD)	As an ISDN supplementary service: Connection transfer into the public network As a subscriber-related feature: Forwards a call during the ringing phase
Call Distribution Element (CDE)	Distributes incoming calls to one or more internal or PISN-internal destinations, depending on the switch position (switch group). A CDE is allocated to a trunk group and / or a DDI number. A CDE can also be allocated a call number and dialled from within the system using that number.
Call Forwarding Busy (CFB)	Call Forwarding Busy.
Call Forwarding No Reply (CFNR)	Call Forwarding Unconditional if no answer.
Call Forwarding on No Reply (CFNR)	The incoming call is routed in parallel to subscriber C after (0), 3, 5 or 7 ringing cycles. The terminals of subscribers B and C then start ringing. The connection is switched through to whichever subscriber answers the calls.
Call Forwarding Unconditional (CFU)	Each internal subscriber can activate a CFU to an internal or external destination. Call Forwarding Unconditional responds differently depending on the System Configuration and the procedure used.
Call Forwarding Unconditional (CFU)	Direct Call Forwarding Unconditional without specifying any conditions.
Call Logging (CL)	Call logging
Call Transfer (CT)	Call transfer
Call Waiting (CW)	Call waiting
Call waiting tone	→ Tone signalling in the handset
Calling Line Identification Presentation (CLIP)	Displays the caller's number to the called party.
Calling Line Identification Restriction (CLIR)	Suppresses the caller's number to the called party.
Calling Name Identification Presentation (CNIP)	Displays the caller's name to the called party.
Calling Name Identification Restriction (CNIR)	Suppresses the caller's name to the called party.
Centrex (CTX)	The designation Centrex, for Central Office Exchange Service, is a product name which some network providers use for the services provided by the → virtual PBX.
Channel Identification (CID)	Sales channel identification: Part of the → Licensing information
Common ISDN Application Interface (CAPI)	TAPI interface
Completion of Calls on No Reply (CCNR)	Callback if free, no answer.
Completion of Calls to Busy Subscriber (CCBS)	Callback if busy

Computer Telephony Integration (CTI)	Programs and equipment that provide PC-based telephony properties and data services, e. g. call identification, ACD, Voice Mail.
Conference tone	→ Tone signalling in the handset
Conference, Add-on (CONF)	Variable conference
Congestion tone	→ Tone signalling in the handset
Connected Line Identification Presentation (COLP)	Displays the called party's number to the caller.
Connected Line Identification Restriction (COLR)	Suppresses the called party's number to the caller.
Connected Name Identification Presentation (CONP)	Displays the called party's name to the caller.
Connected Name Identification Restriction (CONR)	Suppresses the called party's name to the caller.
Continuous ringing	→ ringing pattern
Country Code (CC)	Country code
Courtesy Service	Announcement service for incoming external calls, if the call is not answered.

D

D channel	Control and signalling channel of an ISDN connection.
Data Terminal Equipment (DTE)	Equipment featured on data terminals
Destination PINX	A PINX acts as a destination PINX for the duration of a connection if the connection’s destination subscriber is one of its subscribers.
Digital Enhanced Cordless Telephony (DECT)	Standard for digital radio transmission in cordless systems.
Digital Subscriber Signalling 1 (DSS1)	Signalling protocol for ISDN networks (also called Euro-ISDN).
Direct Dialling In (DDI)	Direct dialling from digital network interfaces to user-network interfaces or terminals.
Direct Dialling Out (DDO)	Source of DDI numbers
Double ringing tone	→ ringing pattern
Download (PBX → PC)	Loads data from the PBX onto a PC.
Dual Tone Multi-Frequency (DTMF)	Dialling method used by analogue terminals, also referred to as dual-tone multi-frequency dialling (DTMF).

E

E.164	1. Numbering plan identifier of the public network as per ITU-T 2. Parameter value of parameter → NPI
Electromagnetic compatibility (EMC)	Guarantee that the system does not propagate any electromagnetic interference fields that exceed standardized limits and which could cause disruptions to other equipment and/or systems in the vicinity. Protection against such fields originating from other equipment or systems in the vicinity.
Equipment Identification (EID)	Equipment identification: Part of the → licensing information
Equipment Identification Module (EIM)	Small, replaceable card containing the → licensing information
Erlang	Unit of traffic intensity during the busy period
Exchange	Short for → public network
Exchange Access	Access to an → exchange line circuit
Exchange access prefix	Prefix of a call number that enables → exchange access
Exchange dial tone	Dial tone supplied by the public network provider → Tone signalling in the handset.
Exchange line	Designates the public network ↔ PBX transmission link; on the PBX, connected to a → network interface.
Exchange line circuit	Network interface to the connection to the public network.
Explicit call transfer (ECT)	Explicit Call Transfer (ISDN supplementary service)
External	Used for "public network" (the private network is not external but PISN-internal).
External call	Call from or into the public network.
External call number	Call number of an incoming call from the public network with NPI = E.164, also a call number in the public network.
External Remote Control (ERC)	External remote control
External subscriber	Subscriber in the public network

F

Featurephone (FP)	All system terminals are featurephones, provided they are not used as → key telephones or → Operator Consoles.

G

Gateway-PINX	A PINX is a gateway PINX for the duration of a connection if it routes that connection from the PISN to the public network or vice versa.
Generic Access Profile (GAP)	Standardized interface for mobile terminals.

H

H.323	IP telephony protocol. The H.323 definitions contain a protocol stack with different interfaces for voice, video and data transmission.
HOLD	Holds a call
Hotkey	Terminal key used for storing freely selectable numbers for abbreviated dialling.
Hotline (HL)	Sets up a connection to a predefined destination using numbers configured in the PBX. On this terminal / handset, the predefined destination is dialled automatically whenever the receiver is picked up or the seize key is pressed.

I

Incoming call	Call from the network to a PBX subscriber.
Incoming Call Logging (ICL)	Incoming call logging
Individual charge counting (ICC)	Allocates the call charges to the cost centre
Initialization	After a first start the system is reset to its factory settings. The configuration and customer data is deleted.
Initialization values	Configuration parameter values after an → initialization
Integrated Services Digital Network (ISDN)	Integrated services digital network for the digital transmission of services such as voice, data, fax, etc.
Internal	Used in general for the subscriber side of the PBX.
Internal call	Both the source and destination are internal
Internal subscribers	Subscriber on a PBX
International prefix	Digit sequence used for dialling abroad, e.g. 0044 for the UK.
Internet Protocol (IP)	The Internet Protocol (Layer 3) is designed to transport data packets from a transmitter to a receiver via several networks. The transmission is packed-oriented, connectionless and not guaranteed.
Internet Service Provider (ISP)	A company that provider access to the Internet and Internet-specific services.
IP Gateway	Converts a phone number into an IP address for forwarding via the LAN.
ISDN terminals	Terminals that comply with the standard of the European Telecommunications Standards Institute.

L

Least Cost Routing (LCR)	Routing function used to determine the network operators via which a call is to be routed. Usually the most cost-effective route is selected.
Licence Code (LIC)	Licence code: Part of the → licensing information
Licence information	Regulates the activation of licensed attributes. Consists of the → Licence Code and the → EID, and is stored on the →EIM card.
Light Emitting Diode (LED)	Light emitting diode: light source used for signalling purposes
Line key	Key on a → key telephone that is allocated to a KT line or on → Operator Console (previously: aggregate key).
Liquid Crystal Display (LCD)	Liquid crystal display
Local Area Network (LAN)	Computer network for geographically limited region.
Long distance code	→ National destination code

M

Multiple Subscriber Number (MSN)	Allocates several call numbers to one ISDN basic access (multiple subscriber number).

N

National Destination Code (NDC)	Code used for identifying national networks, the long-distance code
Network	→ Private or → public network
Network interface	Network-side PBX interfaces
Network Terminator	Network termination
Node	Branch point or end point in a communication network. The nodes of a → PISN are the → PINXs.
Numbering Plan Identifier (NPI)	1. Numbering plan type: In the public network, the numbering plan identifier used is → E.164. In the private network, the numbering plan identifier used is → PNP. 2. Configuration parameter used for specifying the numbering plan identifier. Parameter values: E.164 / PNP / unknown.

O

Offline configuration	Configuration of the PBX without being connected to the system. The data is loaded onto the system using an → Upload.
Online configuration	Direct configuration of the PBX using AIMS
Outgoing call	Call from the PBX to the network
Outgoing Call Logging (OCL)	Outgoing call logging
Output device	PC or printer on a V.24 interface.
Overflow	If the chosen line of a → PINX is not available due to overloading or due to a defect, the connection pending is set up via an alternative path determined by the configuration.

P

Paging Interface (PI)	Interface on the paging interface (S bus interface)
Paging system (PS)	Telecommunication service used for paging internal subscribers.
Partial Rerouting (PARE)	Connection transfer into the public network
PBX subscriber	Internal subscriber on the PBX as opposed to the →VoIP subscriber
Personal Identification Number (PIN)	Personal ID number
Personal Subscription Identity (PSI)	Subscriber ID for identification within the network
PISN internal	Within a PISN
PISN region	Region within a PISN
PISN Subscriber	1. Subscriber in a different → node of a private Network. 2. Category in the internal numbering plan used to replicate the subscribers in the private network.
Pocket Adapter (PA)	Terminal used for connecting a PC via the V.24 interface on the AD2 bus.
Point-to-point protocol	Protocol used for transmitting data via serial connections while making full use of the TCP / IP standard.
Port	Physical access point on the PBX for network interfaces and user-network interfaces.
Preset Conference Calling (PCC)	Preconfigured conference
Primary rate access (PRA)	30B + D channel, (ISDN-)T2 interface
Private Branch Exchange (PBX)	Telecommunication system for access switching
Private Integrated Network Exchange (PINX)	→ Node of a → PISN. Usually a PINX is an ISDN PBX.
Private Integrated Services Network (PISN)	Private network based on the ISDN standard. Characterized by the fact that all connected subscribers can communicate among one another as internal subscribers. This applies to both voice traffic and to ISDN-based data traffic.

Private leased-line network	Private network implemented using dedicated lines. In the PBX configuration a distinction often has to be made between the private leased line network and the public network.
Private Numbering Plan (PNP)	1. Service offered by the network provider. Essentially a virtual counterpart to a PBX numbering plan. Most important component of a → virtual PBX. 2. Parameter value of parameter → NPI
Private Signalling System (PSS1)	→ QSIG
Public network	Telephony and data network accessible to everyone, operated by the public network provider (→ Public Network Operator).
Public Network Operator (PNO)	Public network provider
Pulse dialling (PUL)	Dialling method used by analogue terminals

Q

QSIG networking in the IP network	Transmits voice data and QSIG signalling as IP data packets in the intranet.
QSIG protocol (PSS1)	1. ECMA-standardized signalling protocol used for networking several → PINXs. Now standardized worldwide (ISO / IEC) under the name PSS1. 2. Parameter value of the "Protocol" trunk group parameter. The system supports 2 versions of the QSIG protocol: QSIG (ETSI, 2nd edition) and QSIG / PSS1 ISO.

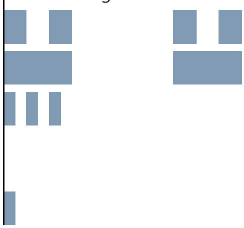
R

Radio unit (RU)	Transceiver (transmitter / receiver) for handsets.
Remote data transmission (RDT)	Data transmission from one data carrier to another via a network or by radio.
Remote maintenance	PBX maintenance via the public or private network.
Restart	Restarts the PBX (power off / on).
Ring-back tone	→ Tone signalling in the handset

Ringing pattern:

The ringing pattern provides a means of identifying whether the call originates from within the PBX or from outside.

- Double ringing tone
- Single ringing tone
- Attention tone
  – e. g. call waiting tone in a call
  – e. g. when a message is received
- Warning tone



- – Error tone whenever an invalid key is pressed (incorrect input)
- • Continuous ringing
- • Discreet ringing

Router	A communication processor between data networks that provides the routing (i.e. determines the path for data transmission and implements the transmission).

S

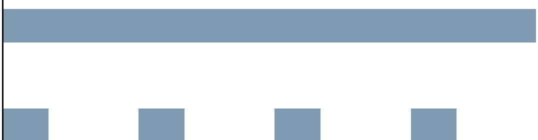
Simple Mail Transfer Protocol (SMTP)	Standard protocol for e-mails on the Internet which regulates the way in which mails are forwarded between the mail servers.
Single ringing tone	→ ringing pattern
Small and medium-sized businesses (SMB)	Companies with up 250 employees (EU definition).
Source PINX	A PINX acts as a source PINX for the duration of a connection if the connection was set up by one of its subscribers.
Subscriber Number (SN)	Subscriber's number

T

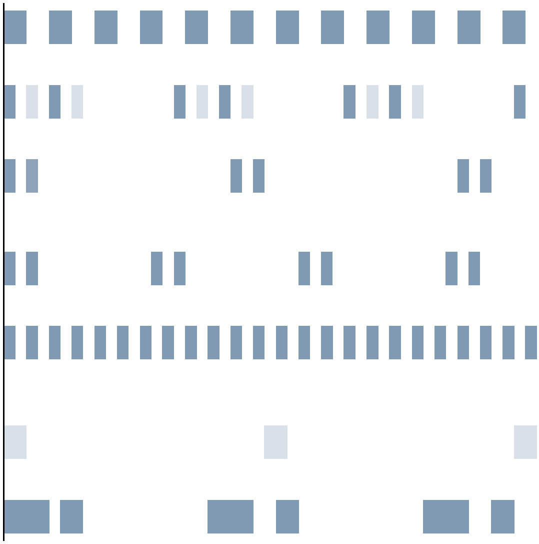
TEI management	→ Point-to-point connection, → Point-to-multipoint connection
Telephony Application Programming Interface (TAPI)	Function stack that integrates the phone into the Win32 model and provides a standardized interface for controlling telecommunications. TAPI is designed to provide software and hardware suppliers with a uniform, non-equipment-based programming model.
Telephony Service Provider (TSP)	TAPI driver
Telephony Service Provider Interface	TAPI interface
Terminal Adapter (TA)	V.24 < → (ISDN-)S interface adapter
Terminal selection digit (TSD)	Single digit used for addressing a terminal on the S interface.
The internal numbering plan	The PBX's numbering plan

Tone signalling in the handset:

- • Dialling tone / Exchange free-to-dial tone
  - – Continuous tone obtained when picking up the handset
- • Ring-back tone
  - – Wait until the call is answered



- Busy tone
  – Destination subscriber is busy
- Acknowledgement tone
  – After initiating various functions
- Call waiting tone
  – Another subscriber is signalling "call waiting" into the current conversation
- Hold tone
  – The subscriber is signalised as on hold
- Congestion tone
  – Feature or authorization not available.
  – Destination subscriber not available
- Conference tone
  – During conference calls or intrusion
- Ringing tone via a paging system



Transit PINX
A PINX acts as a transit PINX for the duration of a connection if it routes that connection from one PINX to another PINX.

Transmission Control Protocol (TCP)
TCP is a connection-oriented transport protocol used for controlling the transmission. It supports the functions of the transport layer and sets up a connection between the entities prior to data transmission.

Type of Number (TON)
Parameter used for classifying a call number
- Parameter values, if the call number corresponds to a NPI = E.164:
  unknown / subscriber / national / international.
- Parameter values, if the call number corresponds to a NPI = PNP:
  unknown / level 0 / level 1 / level 2

U

Upload (PC→PBX)
Loads the data from the PC onto the PBX

User group (UG)
In a user group incoming and internal calls are routed to a group of internal destinations in accordance with a pre-configured call distribution pattern. A user group consists of members of the subscriber group, the elements Operator Console, general bell and the cordless group. Each of these elements can be individually delayed.

User-to-User Signalling
Signalling from one user to another user

V

Virtual PBX	Network provider offer which comprises a → PNP and various ISDN supplementary services. Also known under the name → Centrex. With a virtual PBX the network provider is able to offer his customers the functionality of a PBX.
Voice over Internet protocol (VoIP)	Collective term designating all the techniques used for transmitting voice via IP networks.
VoIP subscriber	Internal telephony subscriber on the → LAN, as opposed to the → PBX subscriber

W

Warning tone	→ ringing pattern
Wide Area Network (WAN)	2 or more LANs grouped together into a network
World Wide Web (WWW)	The Internet: Distributed hypermedia system. The information base of the WWW consists of millions of HTML documents interconnected by hyperlinks.

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