

Remote Office Network Engineering Guidelines

Standard 1.2

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About this document

The *Remote Office Network Engineering Guidelines* (NTP 555-8421-103) is written for individuals who are responsible for the installation, configuration, and day-to-day management of the Remote Office series units and the Reach Line Card (RLC).

This document describes the network engineering guidelines for the Remote Office 9150 unit, Remote Office 911x series unit (9110 and 9115), Meridian Digital Telephone IP Adapter unit (internal and external), and the RLC.

This document provides:

- a clearer understanding of how you should plan your network
- a detailed description of Nortel Networks' patented QoS Transitioning Technology
- background information on Differentiated Services (DiffServ)
- instructions on how to order ISDN lines

Terminology

Throughout this document, the term “host PBX” refers to the following Nortel Networks PBXs:

- Meridian 1
- MSL-100
- Succession Communication Server for Enterprise 1000

Unless otherwise specified, the term “Remote Office units” refers to the following products:

- Remote Office 9110 unit
- Remote Office 9115 unit
- Remote Office 9150 unit
- Meridian Digital Telephone IP Adapter (Internal and External) unit

Understanding your network

A good design of the Remote Office network begins with an understanding of voice and data traffic, and the underlying network that transmits the traffic.

Your network grows and changes over time. Although you may be satisfied with the design of your network today, your needs can change quickly. You must always monitor and evaluate your network on a regular basis to make sure it is serving your needs.

Before you implement Voice over IP (VoIP) on your network, do the following:

- Measure the type, protocols, broadcast-to-unicast ratio, and overall amount of traffic currently being transmitted over your existing LAN and/or WAN. Consider the quality of your existing traffic routes and identify bottlenecks. Refer to “Implementing Remote Office in your network - LAN/WAN” on page 6.
- For Remote Office 9150 units, Remote Office 911x series units, and Meridian Digital IP Adapter units, perform tests to determine if the network is capable of supporting VoIP. Refer to “Testing your network” on page 11.
- For Remote Office 9150 units and RLCs, estimate LAN and WAN bandwidth usage per Remote Office port. Refer to the “Bandwidth usage table” on page 12.
- For Remote Office 9150 units, understand the importance of QoS on your network. Refer to “Quality of service issues” on page 13.
- For Remote Office 9150 units and Remote Office 911x series units, understand how QoS transitioning works. Refer to “Quality of service issues” on page 13.

ATTENTION

This document is a guide only. Implementing VoIP on your network and achieving good voice quality largely depend on the type and volume of traffic being transmitted over your network.

Implementing Remote Office in your network - LAN/WAN

The text in the gray boxes discusses the application of principles discussed above the boxes to the diagram on page 10.

1. Draw a network diagram showing every device on your network that could affect the traffic path between the RLC and the Remote Office units.

The “Network planning diagram” on page 10 shows a sample network containing an RLC and a Remote Office 9150 unit. There are three Ethernet segments, one radio link, and one token ring in the figure. The RLC is installed in the host PBX connected to Link 1, and the Remote Office 9150 unit is connected to the Ethernet segment labeled Link 6.

- Label each segment on the network with the link speed and media capability, such as half- or full-duplex.

In the diagram on page 10, assume all Ethernet segments are half-duplex (typical). The speed is indicated beside the link name.

- Label each device with the throughput rate in bytes and packets per second. Some software-based routers perform at significantly below the Ethernet line-speed (approximately 14K 64-byte packets per second for 10Mbits Ethernet).
- Label major traffic sources on the network with estimates of how much traffic they typically generate and to whom.

An analysis of the network on page 10 indicates that all token ring traffic is local to the token ring, so the only traffic sources are the computer labeled as Source 1 in Link 1, and one computer labeled as Source 2 in Link 5.

2. Determine the path (source and destination) between the major traffic sources on your network.

In our example network, Source 1 and Source 2 communicate along the path of Link5:Link4:Link2:Link1. The RLC and Remote Office 9150 unit communicate along the path of Link6:Link4:Link2:Link1.

3. Identify network bottlenecks and isolate them to minimize collisions. Most networks have one or two bottlenecks. The bottleneck may be a link whose throughput is slower (fewer packets per second transmitted) than the rest of the network, or a convergence point where several links funnel down to a single path.

In our example network, the devices along the common path of Dev3:Link4:Link2:Link1 are the sources of collision/backup. The link speed indicates that the bottleneck is the radio link at 8 Mbytes. The pps (packets per second) estimates indicate that Dev 2 is the bottleneck at 5K pps. Using the average of 14K pps (64-byte packets) for 10Mbytes Ethernet (10BaseT), Dev 2 limits transmission to around 3Mbytes/s. Source 1 can generate 10Mbytes of data that can overload both Link 2 and Dev 2.

4. Prioritize the bottleneck, if needed. If your network generates significantly more data than the bottlenecks can handle, consider putting in place a mechanism to prioritize voice traffic ahead of data traffic.

In our example network, adding priority queues to Dev 3 accomplishes little since its output rates are high compared to the bottleneck. To correct the problem, we determine that we need to do the following:

- add priority queues toward the radio link from Dev 2
- add priority queues and traffic metering toward the radio link from Dev 1.

Adding the priority queues is obvious, but great care must be taken so that our voice traffic doesn't prevent any Source 1 data traffic from reaching Source 2. The reason for the traffic metering is less obvious, but the problem is that Dev 1 and Link 2 can generate traffic faster than Dev 2 can accept it. Therefore, configure Dev 1 to limit its output to around 2 Mbits per second so that it doesn't overrun Dev 2, causing discards

Before you begin to prioritize traffic on your network, you must understand your broadcast domains. Ensure that neither the Remote Office 9150 unit nor the RLC service unnecessary broadcasts.

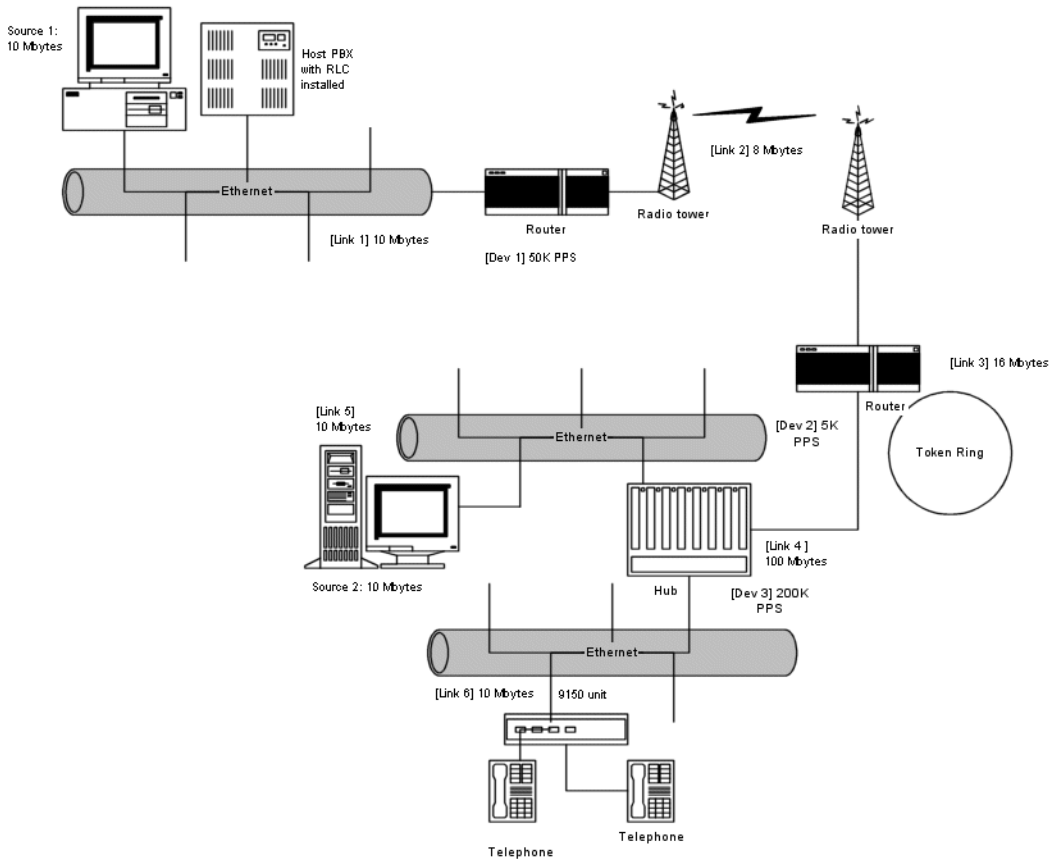
There are a variety of options that provide prioritization within routers and other WAN access devices. The simplest option is a priority-queuing mechanism based on the traffic source and destination IP addresses. Other options include, but are not limited to, the following:

- Use the port numbers as the basis for prioritization.
The Remote Office 911x series units, Remote Office 9150 unit, and RLC use UDP port numbers of 20480 (0x5000) and 20482 (0x5002). Firewall configuration requires you to use these port numbers.
- Enable the DiffServ codepoint and 802.1p header.
Routers can use the DiffServ codepoint to prioritize voice packets ahead of data packets. Ethernet switches can use the 802.1p header for the same purpose. Refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210) for additional information on DiffServ and 802.1p.
- Install an external traffic shaper to prioritize voice traffic ahead of data traffic on those portions of the network that allow prioritizing
Certain WANs recognize the concept of guaranteed bandwidth. These WANs dedicate bandwidth for certain IP addresses or DiffServ codepoint values. Guaranteed bandwidth in Frame Relay, for example, is known as Committed Information Rate (CIR). The guaranteed bandwidth must be sufficient to accommodate all of the voice traffic on the network.

- Ensure that you use the correct signaling connection
The Remote Office units use TCP port 12800 (0x3200) for signaling. Firewall configuration requires you to use this port.
- Ensure that your WAN access device marks all non-critical traffic as Discard Eligible (DE).
- If it is available for your WAN, have the user subscribe to the appropriate level of guaranteed bandwidth, as identified above.
When the link speed is significantly higher than the guaranteed bandwidth, all of the guaranteed bandwidth can be used very early in the sampling interval. With a link speed of two Mbytes per second, a congested network can use all of the guaranteed bandwidth in the first 32 milliseconds. In this situation, the next 968 milliseconds is lost. Some devices (for example, routers) implement delay queues to accommodate this issue. Some documentation refers to this concept as traffic metering.

The RLC and Remote Office 9150 unit, like all other devices, process all broadcast traffic, independent of destination, using a half-duplex 10BaseT Ethernet port. In general, the average load on the Ethernet segments should not exceed 30% of network traffic. Minimize broadcast traffic on your network so that it does not exceed 5% of all network traffic.

Network planning diagram



Testing your network

If you are going to deploy your Remote Office unit on an existing network, perform the following tests to determine if the network can support Voice over IP (VoIP) traffic.

1. For Remote Office 9150 units, use the bandwidth utilization table on page 12 to estimate WAN and LAN bandwidth.

Total BW = max number of simultaneous calls * peak bandwidth from table

2. Perform a PING test (100 pings). Perform this test at multiple times of the day to sample varying network conditions.
 - Divide the minimum delay of any ping by two to get one-way minimum delay.
 - Divide the maximum delay of any ping by two to get one-way maximum delay.

Note: Quality of Service transitioning technology is based upon three factors:

- jitter
- delay
- packet loss

The close correlation of these three factors makes the measurement extremely subjective in nature. Refer to page 14 for a detailed discussion of jitter, delay, and packet loss.

Bandwidth usage table

Packet Size	G.711 ¹	G.726 ¹	G.729 ¹	G.729 \ FAX ^{1,2}
Voice Payload (bytes/30 ms)	240	120	30	38
Voice header (bytes)	12	12	12	12
UDP header (bytes)	8	8	8	8
IP header (bytes)	20	20	20	20
IP Packet Size (bytes)	280	160	70	78

LAN Overhead

Ethernet header Size (bytes)	14	14	14	14
IP Packet Size (bytes)	280	160	70	78
LAN Packet Size (bytes)	294	174	84	92
Peak Data Rate (Kbps)	78	46	22	24
Avg Data Rate-w/silence (Kbps) ³	47	28	13	24 ⁴

IP Over Frame Relay Overhead

Frame Relay Overhead (bytes)	4	4	4	4
RFC 1490 IP Overhead (bytes)	2	2	2	2
IP Packet Size (bytes)	280	160	70	78
Frame size (bytes)	286	166	76	84
Peak Data Rate (Kbps)	76	44	20	22
Avg Data Rate-w/silence (Kbps) ³	46	27	12	22 ⁴

1. compression algorithm
2. 9600 bps (14400 bps not supported)
3. Average data rate with silence suppression is approximately 60% of Peak. Change this number to reflect the site's actual values.
4. Silence suppression, or voice activity detection (VAD) is disabled.

Quality of service issues

The following section discusses ways that the Remote Office product works to resolve Quality of Service (QoS) issues involved in transmitting Voice over IP.

QoS Transitioning Technology

When voice quality degrades on your Remote Office network, the RLC provides improved QoS. Nortel Networks' patented QoS Transitioning Technology enables the RLC to move Remote Office calls from the IP network to the Public Switched Telephone Network (PSTN). The more stable PSTN provides improved QoS until IP network QoS improves.

Once the IP network QoS improves, the RLC automatically recovers Remote Office calls to the IP network. For detailed information on how to configure QoS Transitioning Technology thresholds, refer to the *Reach Line Card Guide* (NTP 555-8421-210).

Note: Because Meridian Digital Telephone IP Adapter units do not have PSTN connectivity, they do not support Quality of Service Transitioning Technology.

How QoS Transitioning Technology works

QoS Transitioning Technology bases transition and recovery functions on the IP network's QoS level. The QoS level is a user-oriented metric that takes one of ten settings. You identify the limits of acceptable voice QoS for each remote site in your Remote Office network by choosing from among those settings using Configuration Manager. The Meridian digital telephone at the remote location and the RLC in the host PBX communicate with one another over the IP network. These communications begin with a 10BaseT Ethernet interface to the corporate intranet.

The RLC and the remote units constantly monitor the QoS level on the data paths between the host PBX and the individual remote sites. Each node calculates the prevailing QoS level based upon the following factors:

- jitter

Jitter is maximum delay minus minimum delay. When jitter on your network exceeds the configured jitter threshold (default 30 milliseconds), processing can drop voice packets, diminishing voice quality. Most customers can accept a small percentage of dropped packets (as much as 5%). However, this is subjective. The jitter buffer can grow up to 180 milliseconds without negatively effecting voice quality, but only for very brief intervals.
- delay

Excessive delay, even with very little jitter, can cause problems. For most users, acceptable one-way minimum delay falls below 100 milliseconds. Only situations with very little jitter can tolerate larger numbers.
- packet loss

Any packet loss can effect voice quality. For most users, up to 5% is acceptable.

Note: Ensure that the devices on your network support wire-speed forwarding of 64-byte packets. Almost all Ethernet switches, hubs, and routers advertise this capability (beware of those that don't). These devices generally add very little to network delay (less than three milliseconds), therefore, they are not the normal suspects for delay. Lower-speed WAN links and software-based routing can add 10s or 100s of milliseconds. This kind of WAN link and routing can significantly effect bandwidth.
- compression algorithm

The G.729 algorithm handles lost packets better than either G.726 or G.711. Because it is specialized for voice, the G.729 algorithm does not allow sharp changes in frequency or amplitude due to lost packets. The G.726 and G.711 algorithms attempt to recreate a non-speech wave-form. Packet loss causes abrupt changes in the recreated wave-form that can create audible cracks or pops.

The RLC calculates the QoS level based upon ANSI TR56 and the E-model found in ITU-T Reg. G.114. When the QoS level falls below the specified transition threshold for the specified period of time, or duration, the receiving Remote Office node rejects the call. The RLC then reroutes the call over the PSTN. When the QoS rises to a point above the specified recovery threshold for the specified period, the RLC restores calls to the IP network. You configure thresholds and durations for each remote site.

When transitioning or recovering multiple calls, the RLC moves calls between the IP network connection and the PSTN connection 64 Kbps at a time. The system waits several seconds between units of 64 Kbps to determine if the IP connection has become more stable.

Note: If using 56 Kbps connections, the RLC transfers calls in units of 56 Kbps.

The RLC moves as many calls as possible (to a maximum of 64 Kbps per B-channel) from the IP connection to the PSTN trunk connection. High priority users always move first. For information on configuring priority, refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210).

- transparent transition and recovery

Transitions can take place during a live call and are transparent to users who only notice improved voice quality. In the case of a complete IP network failure, however, a noticeable break in the call of approximately four seconds occurs. This allows the RLC to transition the call to the PSTN.

Note: When IP degradation causes the RLC to transition calls to the PSTN, a momentary degradation in voice quality can occur as the transition takes place.

- network testing

Both the RLC and the remote unit (9110, 9115, or 9150) run an IP test to determine if the QoS on the IP network meets configured standards. (For more details, refer to “Measuring QoS while offline” on page 18.) To configure QoS Transitioning Technology thresholds, use the Remote Office Configuration Manager graphical user interface (GUI). For further information, refer to the *Reach Line Card Installation and Administration Guide* (NTP 555-8421-210).

The chart on page 17 illustrates how QoS Transitioning Technology works on the RLC and Remote Office 9110, 9115, and 9150 units. The same chart depicts Z and Q as equal periods of time. However, the duration of poor voice quality required for transition to the PSTN, Z, is a period of seconds. The duration of good voice quality required for recovery to the IP network, Q, is a period of minutes. The units of time used to identify acceptable points of transition and recovery differ in an attempt to minimize *transition thrashing*. Transition thrashing is the rapid transition and recovery between the PSTN and the IP network. This can occur when QoS hovers around configured degrade and recover thresholds producing higher than normal PSTN charges.

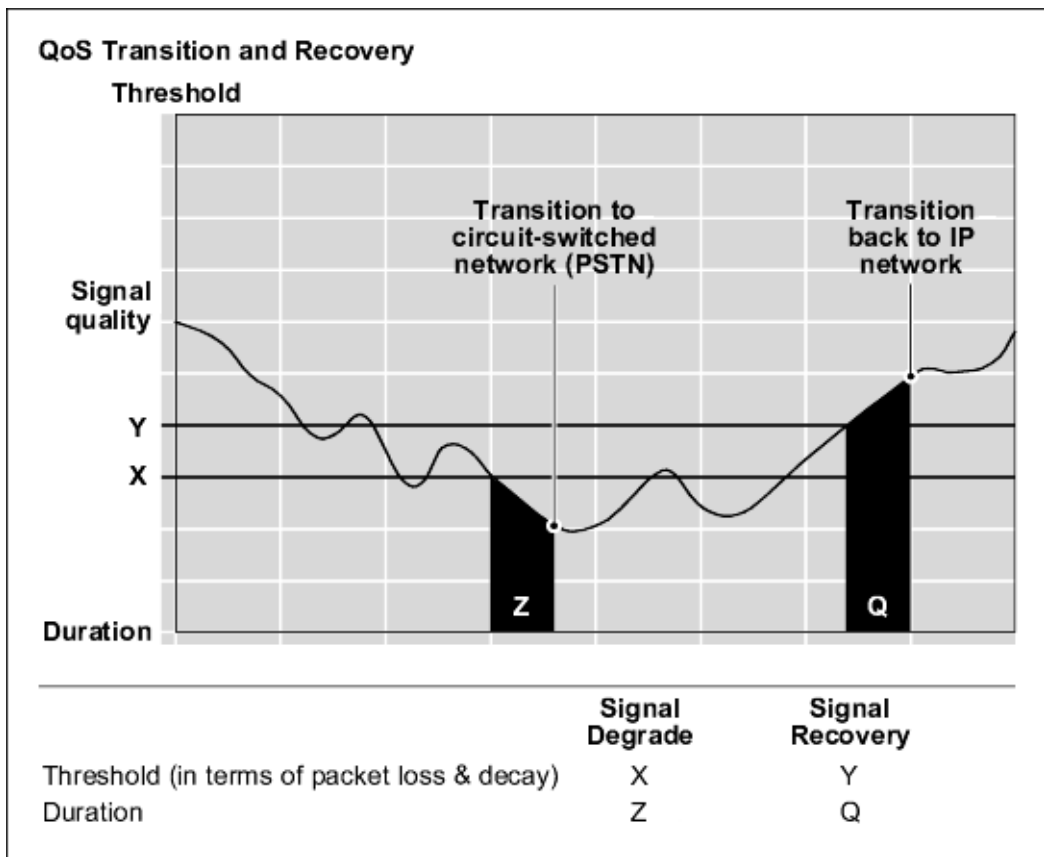
In the telephone call represented by the chart on page 17, signal quality begins in the acceptable range; that is, signal quality is above the X threshold. While signal quality remains in the acceptable range, the RLC routes calls through the IP network.

- transition

When IP QoS degrades below the X threshold, the RLC waits for duration Z to increase the likelihood of continued unacceptable voice QoS. If the voice QoS remains below the X threshold for duration Z, the RLC establishes a communication path for that call on the PSTN. The RLC then transitions the call to the newly established PSTN channel.

- recovery

After the RLC transitions a call to the PSTN, when IP QoS exceeds the Y threshold, the system waits for duration Q. Again, this increases the likelihood of continued acceptable voice QoS on the IP network. After duration Q passes, the RLC moves all current calls back to the IP network and places all new calls over the IP network.



The setting represents

- | | |
|---|---|
| X | signal quality on the IP network is unacceptable below this point. |
| Y | signal quality on the IP network is acceptable above this point. |
| Z | the amount of time that signal quality must be lower than the X threshold before the RLC transitions calls to the PSTN. |
| Q | the amount of time that signal quality must be higher than the Y threshold before the RLC recovers calls to the IP network. |
-

QoS traffic measurements

As voice packets travel across the IP network, the RLC monitors the following QoS parameters:

- average packet delay
 - The RLC calculates this delay using the following statistics gathered from its voice jitter attenuation buffer:
 - minimum packet holding time in the jitter buffer
 - maximum packet holding time in the jitter buffer
 - peak holding time in the jitter buffer
 - time-stamp values in the packet header
 - By accumulating these statistics over time, the RLC can calculate an average packet delay value through the IP network. As the system detects an increase in the average packet delay, it references the signal degrade threshold to determine when to make the transition to the PSTN connection.
 - For a further description of statistics, refer to the *Reach Line Card Installation and Administration Guide (NTP 555-8421-210)*.
- lost packets
 - The RLC calculates lost packet statistics by accumulating the following packet header and voice decoder statistics:
 - voice decoder underrun
 - voice decoder overrun
 - out-of-sequence packet reception

Measuring QoS while offline

Once the RLC reverts to using its PSTN connections, it must continually monitor the IP network. The RLC determines an appropriate time to restore voice traffic to the IP network as follows:

1. Both the RLC and the Remote Office unit (Remote Office 9110, 9115, or 9150) place pseudo-voice traffic on the IP network.

Both the RLC and the Remote Office unit send traffic with a maximum bandwidth of no more than 16 Kbps in short bursts at a higher bit rate to approximate live voice traffic.

2. Both the RLC and the Remote Office unit gather statistics based on the pseudo voice traffic to determine the congestion levels on the network. They use packet time stamps and sequence numbers to monitor the following parameters:
 - average end-to-end delay
 - average round-trip delay
 - average packet-to-packet jitter
 - average packet loss
3. When the parameters listed in step 2 fall below the predetermined threshold for the predetermined period of time, the RLC restores voice traffic to the IP network.

When restoring the connection to the IP network, the system adds hysteresis, or delay, to reduce the noise level during the transition.

Note: Hysteresis, or delay, prevents thrashing between the PSTN and the IP network (for a discussion of *transition thrashing*, refer to page 16), and ensures that acceptable voice QoS exists on the IP network for a predefined period of time.

Log reports and statistics

The Configuration Manager provides a statistics log that identifies the number of QoS transitions. For a detailed description of log and statistic reports, refer to the *Reach Line Card Installation and Configuration Guide* (NTP 555-8421-210).

Telephone display messages

Meridian digital telephones with display capability can display the following Resource Limit messages:

- Bandwidth Limit indicates that there is insufficient BRI bandwidth available to complete the requested action.
- DSP Limit indicates that there are insufficient DSP resources available to complete the requested action.

Relationship between users and services

In the context of a host PBX and Remote Office 9150 unit unit, there are two interfaces that you must consider in the relationship between users and services:

- The Remote Office node (that is, the RLC on the host PBX or the Remote Office unit at the remote site) provides an interface to the Remote Office system for end users. Voice services offered by the Remote Office node must meet user-oriented QoS objectives. Refer to “Quality of service issues” on page 13 for details.
- The Remote Office nodes also provide an interface with the intranet (or Internet in the case of a Remote Office 911x series unit). The intranet provides the “best-effort delivery of IP packets,” as opposed to “guaranteed QoS for real-time voice transport.” The Remote Office node translates the QoS objectives set by the end-users into IP-oriented QoS objectives. This document refers to these objectives as *intranet QoS objectives*.

DiffServ and 802.1p Mapping

Differentiated Services (DiffServ) - a layer 3 standard and 802.1p - a layer 2 standard allow the assignment of priorities to packets over other traffic on the data network. Enabling these two settings can help to provide increased QoS for voice traffic. In order to utilize DiffServ and/or 802.1p, the intermediate devices - hubs, switches, and routers (DiffServ only) must be DiffServ and/or 802.1p compliant. The DiffServ field has a six bit Differentiated Services Code Point (DSCP) and two currently unused bits. By enabling DiffServ, the DSCP in packets from the RLC and the Remote Office unit are given a value of hex 2E - Expedited Forwarding. For further information on DiffServ refer to RFC 2474. The 802.1p packet is tagged and the field is given a value of hex C000. For further information on 802.1p, refer to IEEE 802.1D.

Ordering ISDN Lines

The following section provides you with information regarding the ordering of ISDN lines.

What you need to tell your ISDN service provider

Inform your ISDN service provider that you need the following:

- two B-channels providing both voice and data capability (56 or 64 Kbps clear, depending on your system)
Note: Both B-channels must be PSTN Voice and Data.
- Caller Line Identification (known in the United Kingdom as Calling Line Identity Presentation–CLIP)
- Multiple Subscriber Numbering (MSN)
- two directory numbers (DNs)
- two Service Profile Identifiers (SPIDs)

Note: Multiple Subscriber Numbering (MSN) is required in situations where no SPIDs are provided.

Tell your service provider how the line should be provisioned for data, voice, and other optional services. Different service providers require this information in different ways; increasingly they are using ISDN Order Codes for simplicity, but some still require specific switch type details.

What you need from your ISDN service provider

Your service provider needs to tell you:

- the ISDN service and switch type
- the ISDN directory numbers
- associated SPIDs
- bearer capability (56 or 64 Kbps)

Supported ISDN switches and services for North America

The Remote Office 9150 unit supports the most common switch types. The following table shows the ISDN services available for each of the switch types.

Switch Type	ISDN Service
Nortel Networks DMS-100	National ISDN 1 (NI-1)
	National ISDN 2 (NI-2)
AT &T 5ESS	Custom Point-to-Point
	Custom Multipoint
	National ISDN 1 (NI-1)
	National ISDN 2 (NI-2)
Siemens	National ISDN 1 (NI-1)
	National ISDN 2 (NI-2)

Supported European ISDN services

For European installations, the Remote Office 9150 unit supports country-specific EuroISDN installations, but does not support pre-EuroISDN installations.

Using supplementary services

The following supplementary data services are not recommended for use with Remote Office 9150 and 911x units:

- call waiting
- bearer-channel bonding
- call-waiting ID
- hunt

If you need to order Hunt services, order Multiline hunting rather than Overflow (EKTS) forward. The following must be true when you order these services on the BRI circuits:

- The B-channel requested during call set-up must be the B-channel that the call presents to

— The CALLED DN must be the DN of the B-channel the call presents to Any other services, such as Forward on Busy must follow the guidelines above.

Example A

1 BRI circuit

B1: 416-555-1212, the published number

B2: 416-555-1213

A PSTN set calls B1.

The published number is currently busy.

Hunt must provide a call setup message for B2 with a CALLED number of 416-555-1213.

Example B

1 BRI circuit

B1: 416-555-1212

B2: 416-555-1213

The published number is different - 416-555-5000.

No calls are up.

When a PSTN set calls 416-555-5000, the call presents to B1 with a CALLED CLID of 416-555-1212.

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