[Standard—Nortel Networks Confidential]

Part No. NN10040-112
April, 2003

SIP Multimedia Web Client User Guide

NORTEL NETWORKS™

Nortel Networks Inc. software license agreement

This Software License Agreement (“License Agreement”) is between you, the end-user (“Customer”) and Nortel Networks Corporation and its subsidiaries and affiliates (“Nortel Networks”). PLEASE READ THE FOLLOWING CAREFULLY. YOU MUST ACCEPT THESE LICENSE TERMS IN ORDER TO DOWNLOAD AND/OR USE THE SOFTWARE. USE OF THE SOFTWARE CONSTITUTES YOUR ACCEPTANCE OF THIS LICENSE AGREEMENT. If you do not accept these terms and conditions, return the Software, unused and in the original shipping container, within 30 days of purchase to obtain a credit for the full purchase price.

“Software” is owned or licensed by Nortel Networks, its parent or one of its subsidiaries or affiliates, and is copyrighted and licensed, not sold. Software consists of machine-readable instructions, its components, data, audio-visual content (such as images, text, recordings or pictures) and related licensed materials including all whole or partial copies. Nortel Networks grants you a license to use the Software only in the country where you acquired the Software. You obtain no rights other than those granted to you under this License Agreement. You are responsible for the selection of the Software and for the installation of, use of, and results obtained from the Software.

1. Licensed Use of Software. Nortel Networks grants Customer a nonexclusive license to use a copy of the Software on only one machine at any one time or to the extent of the activation or authorized usage level, whichever is applicable. To the extent Software is furnished for use with designated hardware or Customer furnished equipment (“CFE”), Customer is granted a nonexclusive license to use Software only on such hardware or CFE, as applicable. Software contains trade secrets and Customer agrees to treat Software as confidential information using the same care and discretion Customer uses with its own similar information that it does not wish to disclose, publish or disseminate. Customer will ensure that anyone who uses the Software does so only in compliance with the terms of this Agreement. Customer shall not a) use, copy, modify, transfer or distribute the Software except as expressly authorized; b) reverse assemble, reverse compile, reverse engineer or otherwise translate the Software; c) create derivative works or modifications unless expressly authorized; or d) sublicense, rent or lease the Software. Licensors of intellectual property to Nortel Networks are beneficiaries of this provision. Upon termination or breach of the license by Customer or in the event designated hardware or CFE is no longer in use, Customer will promptly return the Software to Nortel Networks or certify its destruction. Nortel Networks may audit by remote polling or other reasonable means to determine Customer’s Software activation or usage levels. If suppliers of third party software included in Software require Nortel Networks to include additional or different terms, Customer agrees to abide by such terms provided by Nortel Networks with respect to such third party software.

2. Warranty. Except as may be otherwise expressly agreed to in writing between Nortel Networks and Customer, Software is provided “AS IS” without any warranties (conditions) of any kind. NORTEL NETWORKS DISCLAIMS ALL WARRANTIES (CONDITIONS) FOR THE SOFTWARE, EITHER EXPRESS OR IMPLIED, INCLUDING, BUT NOT LIMITED TO THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE AND ANY WARRANTY OF NON-INFRINGEMENT. Nortel Networks is not obligated to provide support of any kind for the Software. Some jurisdictions do not allow exclusion of implied warranties, and, in such event, the above exclusions may not apply.

3. Limitation of Remedies. IN NO EVENT SHALL NORTEL NETWORKS OR ITS AGENTS OR SUPPLIERS BE LIABLE FOR ANY OF THE FOLLOWING: a) DAMAGES BASED ON ANY THIRD PARTY CLAIM; b) LOSS OF, OR DAMAGE TO, CUSTOMER’S RECORDS, FILES OR DATA; OR c) DIRECT, INDIRECT, SPECIAL, INCIDENTAL, PUNITIVE, OR CONSEQUENTIAL DAMAGES (INCLUDING LOST PROFITS OR SAVINGS), WHETHER IN CONTRACT, TORT OR OTHERWISE (INCLUDING NEGLIGENCE) ARISING OUT OF YOUR USE OF THE SOFTWARE, EVEN IF NORTEL NETWORKS, ITS AGENTS OR SUPPLIERS HAVE BEEN ADVISED OF THEIR POSSIBILITY. The forgoing limitations of remedies also apply to any developer and/or supplier of the Software. Such developer and/or supplier is an intended beneficiary of this Section. Some jurisdictions do not allow these limitations or exclusions and, in such event, they may not apply.

4. General

a. If Customer is the United States Government, the following paragraph shall apply: All Nortel Networks Software available under this License Agreement is commercial computer software and commercial computer software documentation and, in the event Software is licensed for or on behalf of the United States

Government, the respective rights to the software and software documentation are governed by Nortel Networks standard commercial license in accordance with U.S. Federal Regulations at 48 C.F.R. Sections 12.212 (for non-DoD entities) and 48 C.F.R. 227.7202 (for DoD entities).

b. Customer may terminate the license at any time. Nortel Networks may terminate the license if Customer fails to comply with the terms and conditions of this license. In either event, upon termination, Customer must either return the Software to Nortel Networks or certify its destruction.

c. Customer is responsible for payment of any taxes, including personal property taxes, resulting from Customer's use of the Software. Customer agrees to comply with all applicable laws including all applicable export and import laws and regulations.

d. Neither party may bring an action, regardless of form, more than two years after the cause of the action arose.

e. The terms and conditions of this License Agreement form the complete and exclusive agreement between Customer and Nortel Networks.

f. This License Agreement is governed by the laws of the country in which Customer acquires the Software. If the Software is acquired in the United States, then this License Agreement is governed by the laws of the state of New York.

Contents

Figures

Tables

Welcome

The *SIP Multimedia Web Client User Guide* provides you with the instructions you need to get up and running with the SIP Multimedia Web Client, a feature-rich on-line user interface for multimedia communication.

Topics include:

- Getting started
- Managing your directories
- Multimedia communication
- Network settings

The SIP Multimedia Web Client uses a Web browser on your PC to deliver multimedia communication services. It enables you to efficiently perform diverse tasks in a single communication session, gives you new control over how calls are handled, puts the human face into incoming calls, and lets you manage incoming and outgoing communications in new ways.

Audience

This guide is intended for subscribers of the SIP Multimedia Web Client services and features.

Text conventions

This guide uses the following text conventions:

bold text	Indicates the command key or link you need to press or click. Examples: Press **Ok**, Click **Help**
italic text	Indicates new terms, document titles

Acronyms

This guide uses the following acronyms:

ERC	Express Routing Code
IP	Internet Protocol
JPEG	Joint Photographic Experts Group
PNG	Portable Network Graphic
SIP	Session Initiation Protocol
URL	Universal Resource Locator (internet address)

Related publications

Other publications related to the SIP Multimedia Web Client Getting Started Guide:

- *SIP Multimedia PC Client User Guide*
- *SIP Multimedia Personal Agent User Guide*

How to get help

If you purchased a service contract for your Nortel Networks product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller for assistance.

If you purchased a Nortel Networks service program, contact Nortel Networks Technical Support. To obtain contact information online, go to the www.nortelnetworks.com/cgi-bin/comments/comments.cgi URL, then click on Technical Support.

From the Technical Support page, you can open a Customer Service Request online or find the telephone number for the nearest Technical Solutions Center. If you are not connected to the Internet, you can call 1-800-4NORTEL (1-800-466-7835) to learn the telephone number for the nearest Technical Solutions Center.

An Express Routing Code (ERC) is available for many Nortel Networks products and services. When you use an ERC, your call is routed to a technical support person who specializes in supporting that product or service. To locate an ERC for your product or service, go to the http://www.nortelnetworks.com/help/contact/erc/index.html URL.

Chapter 1
Getting started

This section provides an introduction and set up information for the SIP Multimedia Web Client. Topics include:

What is the SIP Multimedia Web Client?

The SIP (Session Initiation Protocol) Multimedia Web Client is a web based access client that provides you with various multimedia and telephony features. Your service provider will give you a URL to access the SIP Multimedia Web Client through your web browser. You can also access the SIP Multimedia Web Client from a link in the SIP Personal Agent (optional program).

Figure 1 SIP Multimedia Web Client

SIP Multimedia Web Client features and services

The SIP Multimedia Web Client features and services include:

Table 1 Features and services

Telephony services	**Multimedia services**
real time call handling (answer, hold, transfer, reject, voicemail, pass)	picture caller ID
transfer, hold, mute	Instant Message (point to point, broadcast, Buddies)
calling line ID	Web page push and co-browsing
click to call (from personal network-based address book or list of Buddies)	three-way client based audio conferencing
multiple lines	server based audio conferencing determined by your service package (up to 32 ports)
calling subject delivery	video calls
	personal network-based address book
	Buddies list management

Some useful terms to know

The SIP Multimedia Web Client uses terms that may be new to you, such as:

- proxy server
- SIP address
- address book
- buddies
- presence

What is a proxy server?

A proxy server is an application that relays data between your SIP Multimedia Web Client and the network. The IP address of the proxy server is part of the URL you use to access the SIP Multimedia Web Client. When you connect to the proxy server you need to provide your username and password.

What is a SIP address?

When you enter an “address” to call someone, that can mean entering a telephone number or a SIP address. A SIP address is a unique identifier of users on the IP network. It has the same format as an email address, for example, jdoe@lab1.org, but it is not an email address. The network can track where you are and route your calls when you log into any SIP Multimedia Web Client, SIP Multimedia PC Client or i2004 Internet Telephone with your SIP address. Using SIP addresses allows you to take advantage of the more powerful features of the Web Client such as presence.

In order to make it easier to place calls, you can store addresses (SIP addresses or telephone numbers) in an address book.

Address book

Your personal network-based address book is a key tool for managing addresses. You can save your addresses for quick call access as well as organize addresses into groups. To prevent someone from seeing your online status you can add them to a ban list.

When you make changes in your network-based address book it applies to all the devices you can use to access the network (SIP Multimedia Web Client, SIP Multimedia PC Client, i2004 Internet Telephone).

Buddies

Within your network-based address book you can designate entries as Buddies. People that you contact frequently are good candidates for your list of Buddies. If you have entered the SIP address of someone in your list of Buddies then you can see their online presence status in the display area.

Presence

Presence is a feature that allows you to see the online status of other users on the network and also a way to alert others to your status. The Buddies tree area of the SIP Multimedia Web Client main page, shows you who is online, offline, or away from their desk. From the pull-down menu below the Buddies tree you can change your presence status to let other users know if you are online or away from your desk. You can also configure the SIP Multimedia Web Client to automatically alert others if you are on the telephone or your PC has been idle for a time, see "Setting up presence (optional)".

Before you begin

You need the following items to start using the SIP Multimedia Web Client:

- authorization to access the SIP Multimedia Web Client (see your System Administrator for login details)
- a PC configured with the required minimum software and hardware (as described below)

- internet access and a connection meeting the minimum transmission speed requirements (as described below)

Tip: The complete functionality of the Nortel Networks SIP Multimedia Web Client is available if you have a suitably equipped PC with microphone and web camera. For audio quality, privacy, and respectfulness in an open office environment, most users will prefer to use a headset connected to the computer.

- correct configuration of your Internet Explorer* web browser

System requirements

The SIP Multimedia Web Client can operate with the minimum hardware and operating system requirements but the recommended requirements will provide enhanced multimedia communications quality.

Minimum hardware and operating system requirements

- 550 MHz Pentium-class or equivalent processor
- Windows 98* (SE), Windows ME, Windows NT* 4.x with SP5, Windows 2000, and Windows XP
- 28.8Kbps modem
- full duplex sound card and microphone

Note: Some laptops do not come equipped with full duplex sound cards.

- 48MB free RAM. This requirement is in addition to the memory requirements of the operating system and other concurrent applications.
- 23MB free hard disk space (if Java Runtime Environment needs to be installed, otherwise 3 MB)
- Netscape Communicator* 4.77 or Microsoft Internet Explorer 5.0
  - Cookies enabled
  - Javascript enabled

- 640 x 480 @ 8bpp 256 colors VGA
- mouse

Recommended hardware and operating system requirements

- 1GHz (or higher) Pentium-class or equivalent processor
- Windows 98 (SE) or Windows XP, Windows NT 4.x with SP5, and Windows 2000
- 56Kbps modem or faster network connection (DSL, Cable, LAN, etc.)
- full duplex sound card with headset (microphone-headphone combination)
- 64MB free RAM
- 23MB free hard disk space (if Java Runtime Environment needs to be installed, otherwise 3 MB)
- Netscape Communicator 4.77 or greater, Microsoft Internet Explorer 5.0 or greater
  - Cookies enabled
  - Javascript enabled
- 800 x 600 @ 16bpp 65,536 colors VGA or better video graphics card
- mouse

Optional hardware and software requirements

- USB-based Internet Video Camera (Web Cam). See “Hardware notes” on page 61 for more information about video camera support.

Note: A 16bpp (65,536) VGA or better video graphics mode is required in order to send video.

Web browser configuration

In order to prevent your SIP Multimedia Web Client from being replaced when you click on a web link, you must disable the “Reuse windows for launching shortcuts” option. This configuration is required for Internet Explorer (IE) users.

From the IE Web browser

1. Click **Tools**> **Internet Options** and the **Advanced** tab.
2. Uncheck the Reuse windows for launching shortcuts option.
3. Click **Apply**.
4. Click **Ok**.

Starting the SIP Multimedia Web Client

To begin using the SIP Multimedia Web Client, access the SIP Multimedia Web Client URL that you received from your service provider. You may need to follow the instructions on the website in order to start the SIP Multimedia Web Client.

Or, from the SIP Personal Agent, click the SIP Multimedia Web Client icon

The SIP Multimedia Web Client home page and registration frame appear on your PC. The status window instructs you to register, which means identifying yourself to the proxy (intermediary) server that will direct your calls.

Tip: The first time you start the SIP Multimedia Web Client, you may be prompted with a security screen. If so, click Grant this session or Grant always.

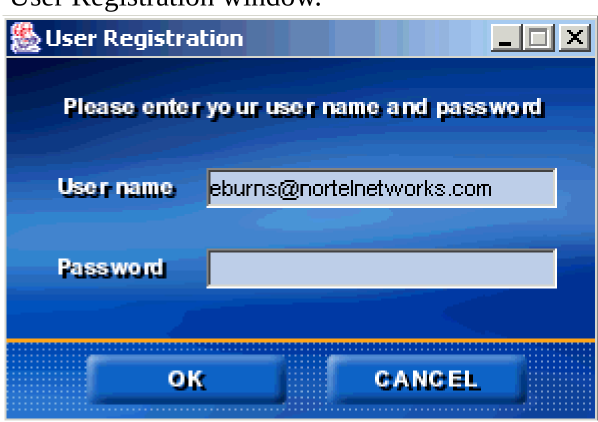
If you do not have a Java Run Time 1.3 plug-in for your browser you will be prompted to install it automatically before you can continue.

Figure 2 Security Screen

Registering with the proxy server

To register with the proxy server to use the SIP Multimedia Web Client:

1 Enter your username and password (for example, userID@mydomain.com) in the User Registration window.



2 Click **Ok**.

The SIP Multimedia Web Client checks to make sure you are authorized to use the service, then displays your username and “Registered” in the SIP Multimedia Web Client status window. (Unauthorized users will see a Not registered or Registration failed message.)

Exiting the SIP Multimedia Web Client

To exit the SIP Multimedia Web Client, click the **X** button in the upper right of the home window to close the application.

Unregister

If you want to manually disconnect without closing the SIP Multimedia Web Client main page

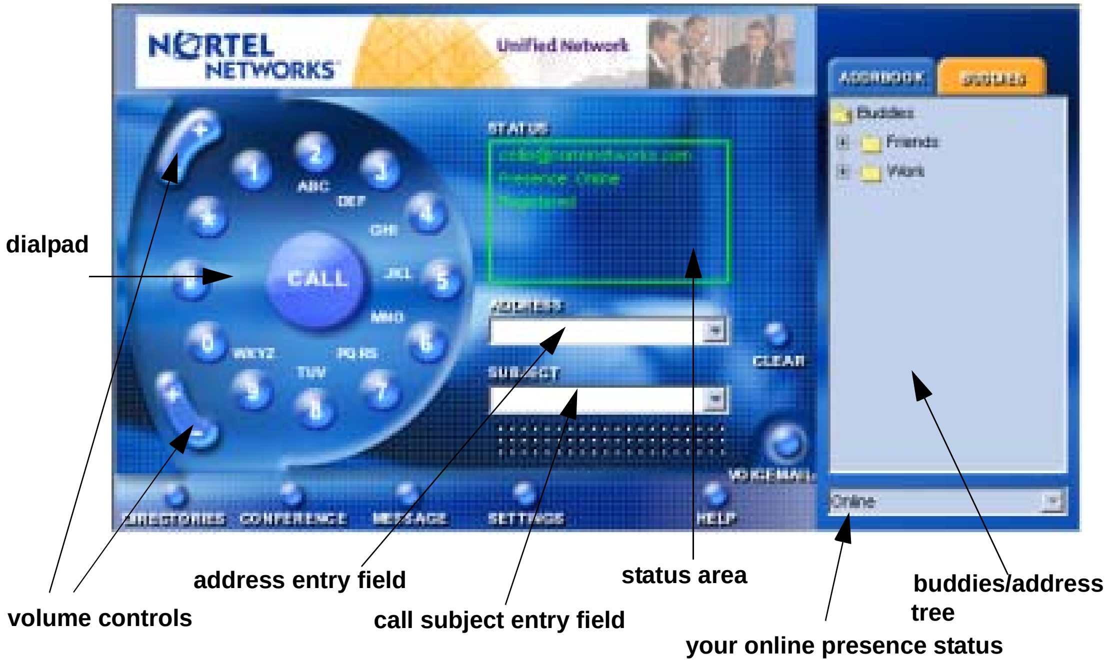
1 Click the **Settings** button. The Settings window opens at the **Proxy** tab.

2 Click the **Unregister** button. The status area displays Not Registered.

3 Click **Ok** to close the Settings window.

Using the SIP Multimedia Web Client interface

Figure 3 SIP Multimedia Web Client home page



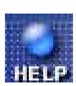
The SIP Multimedia Web Client home page that appears when you start the SIP Multimedia Web Client consists of the following areas:

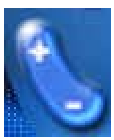
Home page area	Description
Dialpad	Allows you to dial phone numbers, initiate a call, adjust ringer volume, and adjust audio volume.
Status area	Changes based on the context (at startup, initiating a call, during a call, etc.) to display such information as user/call status, caller ID, or the dialog of an instant messaging session
Address and Subject Fields	Allows you to enter a destination (username, IP address, phone number, and so forth) and topic when initiating a call, or to display such information during an active call.
Function buttons	Buttons lining the bottom and right edges of the window provide one-click ease to user directories, conference calling, settings, on-line help, voicemail, instant message, and more.

Home page area	Description
Addrbook (tab)	Displays all of the network-base address book entries. Right-click on an entry to call or send an instant message.
Buddies (tab)	Displays a list of your Buddies and their online presence status. Right-click on an entry to call or send an instant message.
Presence	Below the Buddies tree area is a drop down list to set your own presence status (Online, Busy, Show Offline, Be Right Back, Out to Lunch, Away).

Button functions

You can access the following key functions from the SIP Multimedia Web Client home page.

Function	Description
	Displays your network-based personal address book, from which you can add, edit, or delete entries, or click on an entry to initiate a call or send an Instant Message. From the address book web page you can link to your Groups, Buddies, and Ban list. You can also click the All Users link to search for addresses.
	Allows you to set up and take part in multiparty conference calls.
	Provides point-and-click access to view or change your SIP Multimedia Web Client settings.
	Offers one-click access to online help.

Function	Description
	Opens a dialog box for you to send an Instant Message to one or more users, even users who are not actively on your multimedia call.
	Clears entries in the Address and Subject fields.
	Allows you to listen to your voicemails.
Volume control 	On the dialpad there are two buttons with a + and - sign. Either one allows you to adjust the speaker/ microphone volume level.

How to get help

There are two ways that you can access SIP Multimedia Web Client help:

- online help - from the SIP Multimedia Web Client home page, click the **Help** button to view a task-based help system. The online help provides:
  - — help pages containing forward and backward navigation icons
  - — procedures that help you use the SIP Multimedia Web Client
  - — links to all topics
  - — a table of contents with hypertext links
  - — an Index
- tool tip help - a small help description is available when you roll your mouse over a button on the SIP Multimedia Web Client interface.

Chapter 2
Managing your directories

This section discusses how you can set up your address book, including creating groups of addresses, a ban list and a list of Buddies that you would contact on a day to day basis.

Setting up your network-based address book

Your network-based address book is a dynamic list of information. The Address Book link that you access from the **Directories** button lists the nickname, first name, and last name of an entry in your address book. You can also define a group for an address book entry to belong to.

Anytime you add an entry or make a change to an entry, your address book is automatically updated.

Tip: The network-based address book is also available in the SIP Multimedia PC Client, SIP Personal Agent and on the i2004 Internet Telephone. Any additions or changes that you make to your address book is kept current and shared across these applications and devices.

Adding groups

Before you begin adding entries into your address book, you may want to create a group for the entry to belong to. By defining groups you can partition address book entries into convenient categories. For example you may define groups for customers, colleagues and personal entries. If you do not specify a group for an address book entry and you assign this person to your list of Buddies then their entry will appear in a group marked Unknown.

To set up a group

1 Click the **Directories** button. Your network-based address book displays.

2 Click the **Groups** link. A list of defined groups appears. If there are no groups defined then you will see an empty list.

3 To add a group, click **Add Group**.The Add Address Book Group page displays. Enter the name of the group. For example, you can group all the members of your family into a group called Family, or Home.

4 Click the **AddGroup** button.

5 Click the **AddGroup** link again to add more groups or click the **Address Book** link to add an entry and assign groups in your network-based address book.

Tip: Click the “X” on the upper right hand corner to close this Web browser page. Or leave the page open and continue with the next procedure.

Adding an address book entry

To add a new entry in your network-based address book

1 Click the **Directories** button on the SIP Multimedia Web Client home page. (Or, if the Directory page is already open, click the **Address Book** link). The address book page displays.

Tip: If this is the first time you are using the SIP Multimedia Web Client, your address book may be empty. However, if you have already set up your contact information using the SIP Personal Agent or the SIP Multimedia PC Client, the entries will appear on this page.

2 Click the **Add Entry** button. The Add Entry page displays a fill-in-the-blank form. Use your mouse to click in each field.

Tip: Do not press Enter or Tab, on your keyboard to move through the fields.

3 Enter a **Nickname** (required field), name and telephone numbers for the contact.

4 Enter the contact information (phone numbers, addresses) for the contact.

5 Select a group from the pull-down list. (This is the list of groups that you previously added.) If you do not choose a group from the available list then the entry will default to the first group in the list. If there are no groups defined then this entry will appear in your address book as Unknown.

6 In the Primary Contact field (required field), it is recommended that you enter the SIP address of the contact (a requirement for the list of Buddies). If you do not enter a SIP address then you will not receive any online Presence information for that user.

7 Select **Yes** or **No** from the drop down list next to the Add as Buddy field to have this contact added to your network-based list of Buddies.

8 Enter text in the Notes field about the contact. This information displays in the SIP Personal Agent and in the SIP Multimedia PC Client.

9 Click **Save**.The new entry appears in the address book list.

Modifying an address book entry

To modify an address book entry

1 Click the **Directories** button from the SIP Multimedia Web Client home page.

2 From your address book, click the nickname of the entry you wish to modify.

3 Enter your changes on the address book entry page.

4 Click **Save**. The address book changes are dynamically synchronized with the SIP Personal Agent, the SIP Multimedia PC Client, and the i2004 Internet Telephone.

Deleting an address book entry

To delete an entry from your network-based address book

1 Click the **Directories** button from the SIP Multimedia Web Client home page.

2 Click the **Delete** link for the address book entry you wish to delete. The entry is deleted from your network-based address book that is dynamically synchronized with the SIP Personal Agent, the SIP Multimedia PC Client, and the i2004 Internet Telephone.

Click to call and IMs

From the **Addrbook** tab on the SIP Multimedia Web Client home page you can right click on any entry to call or send an instant message.

Managing your list of Buddies

The SIP Multimedia Web Client allows you to add any address book entry to your list of Buddies. When an address book entry is marked as one of your Buddies, it shows up in the Buddies tree on the SIP Multimedia Web Client home page.

Figure 4 Buddies tree

The SIP Multimedia Web Client displays the presence information (the online status) of each of your Buddies. You define the address book entries that are added to the list of Buddies. For example, it may be useful to keep track of people that you regularly work with to see when they are available.

Tip: If this is the first time you are using the SIP Multimedia Web Client, your Buddies tree may be empty. However, if you have already set up your list of Buddies using the SIP Personal Agent or the SIP Multimedia PC Client, you will see those entries appear in your Buddies tree.

Creating and viewing a list of Buddies

To create and view a list of Buddies

1. Click the **Directories** button from the SIP Multimedia Web Client home page.
2. When adding or modifying an address book entry, leave the default **Yes** from the drop down list next to the Buddy field on the Add Entry page to add a buddy to your list. Select **No** if you do not want this person to appear in your list of Buddies.
3. Click the **Buddies** link from the address book page to view your list of Buddies.

Deleting a buddy

To delete a buddy from your network-based list of Buddies

1. Click the **Directories** button from the SIP Multimedia Web Client home page.
2. Click the entry in your address book and change the Buddies field to **No**.
3. Click the **Save** button. The entry is deleted from the list of Buddies. This information is dynamically synchronized with the SIP Personal Agent, the SIP Multimedia PC Client, and the i2004 Internet Telephone lists of Buddies.

Buddy actions

Each entry in the list of Buddies includes a link to call or send an instant message. From the SIP Multimedia Web Client home page, right click a Buddies entry and choose an action.

Using the ban list

There may be times when you want to ban someone from viewing your online presence status (Online, Busy, Show Offline, Be Right Back, Out to Lunch, Away). The person you add to the ban list can still call you, but they can no longer see your online status.

Adding a ban list entry

To add an entry to the ban list

1 Click the **Directories** button from the SIP Multimedia Web Client home page.

2 Click the **Ban List** link on the address book page. The Ban List page displays.

3 Enter the SIP address of the person you want to ban, or, if this person is in your address book you can enter their nickname.

4 Click **Ban Entry**. The entry appears on your ban list.

Deleting a ban list entry

To delete an entry from the ban list

1 Click the **Directories** button from the SIP Multimedia Web Client home page.

2 Click the **Ban List** link on the address book page. The Ban List page displays.

3 Click the **Delete** link next to the username you wish to remove from the ban list.The entry is deleted from your ban list.

4 Click a link to go to another Directories page or close the window.

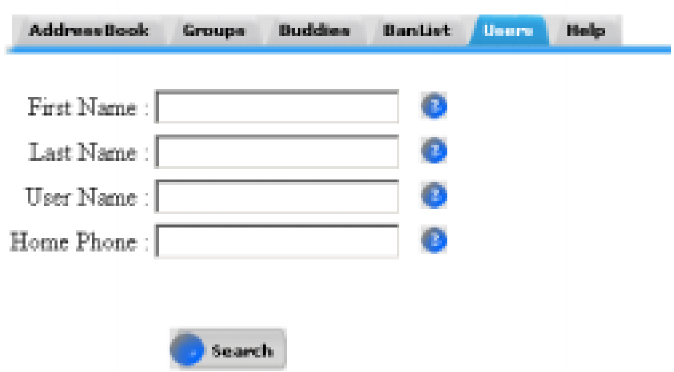
Searching for a contact

If you want to add a user to your address book but are unsure of their address you can do a search within your network group.

To search for a user in your network group

1 Click the **Directories** button from the SIP Multimedia Web Client home page.

2 Click the **Users** link on the address book page. The Search page displays.



3 Enter a search based on first, last, or username or the contact's phone number.

4 Click **Search**. A list of users, within the domain (network group) that meet the search criteria, displays. Write down the information you need.

Tip: If you leave the fields blank and click **Search**, then a list of all users in the domain displays.

Chapter 3 Multimedia Communication

This section provides instructions for using the many features of the SIP Multimedia Web Client that extend your communications ability. Topics include:

Making a call

To make a call

1. From the SIP Multimedia Web Client home page, enter an address (username, SIP address, or public network telephone number) in the **Address** field. You can also select a contact to call from the drop down list (a list of entries you have previously called, plus Buddies or Address book entries) in the Address field.

Tip: When dialing a public telephone network number, enter the numbers with no punctuation in the address field. (For example, 972555624). For outside calls from an office system, or for long-distance calls, be sure to include any necessary access codes. (For example: 1409551234).

2 Enter a subject for the call to be displayed to the calling party in the Subject field, or select a subject from the drop down list (optional).

3 Click the **Call** button or press **Enter** on your keyboard.

The SIP Multimedia Web Client starts dialing the call, and displays information about the called party and call attempt status in the Call Control window.

After making a call

The following table describes what happens after you make a call:

Table 2 After making a call

If...	**Then...**
The other party answers the call	1. connection is established and the call is live. 2. the SIP Multimedia Web Client opens a Call Control window for the call. 3. the Call Control window presents the following call options Hold - places the caller on hold Mute - mutes the call (voice) Instant Message - sends an Instant Message to the caller Transfer - transfers the caller to another party Web Push - pushes Web pages to the party you are calling Hangup - hangs up the call 4. the Call Control window remains open for the duration of the call.
The other party is busy	The SIP Multimedia Web Client home page displays a “User temporarily unavailable message” in a pop-up dialog box, and plays a busy tone.
The party you are calling declines the call	The SIP Multimedia Web Client plays a busy tone and the status window shows Declined. If the party sends a reason then the reason displays in the status area.
The party you are calling transfers the call to another destination	The call automatically transfers.

Receiving a call

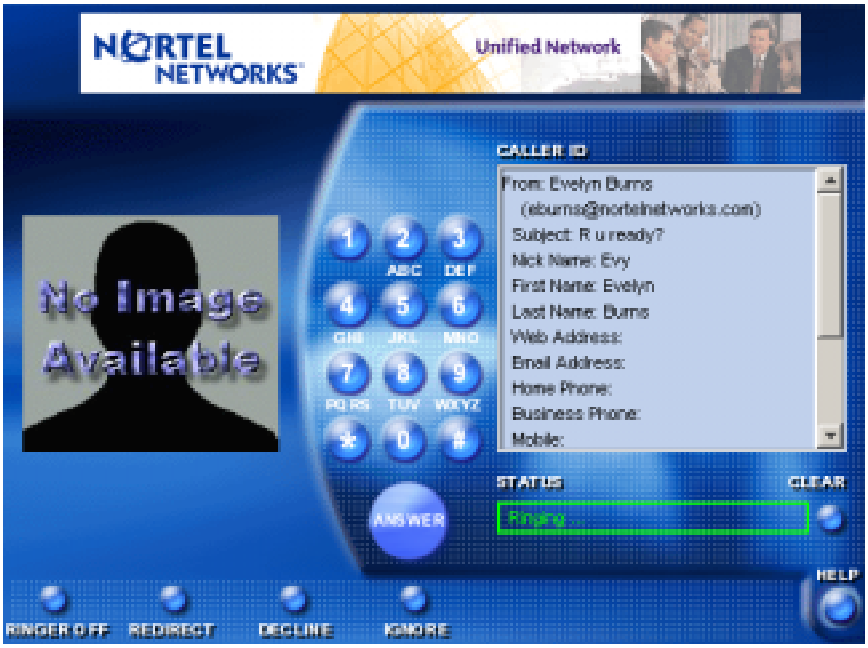
When the SIP Multimedia Web Client is active on your computer and receives an incoming call, it displays a Call Control window on your PC screen that shows:

- network username of the calling party
- name, phone, email, and business, if the information for this entry is in your address book
- the subject of the call (if the caller filled in this field when placing the call)
- photograph of the caller, if this is available from the network.

- call-handling options: buttons for Ringer Off, Redirect, Decline, Ignore and Answer

In the status area of the home page, the SIP Multimedia Web Client displays the caller's username and the message, “You have an incoming call.”

Figure 5 Incoming Call Control window



Tip: If you do not want to hear this incoming call ring then click **Ringer Off**. The Incoming Call Control window remains open and you can still answer the call but the audible ringing is muted.

Answering the call

To answer a call

1. Click the **Answer** button at the bottom of the Call Control window.
2. The SIP Multimedia Web Client connects the call and changes the buttons on the bottom of the Call Control window in the following manner

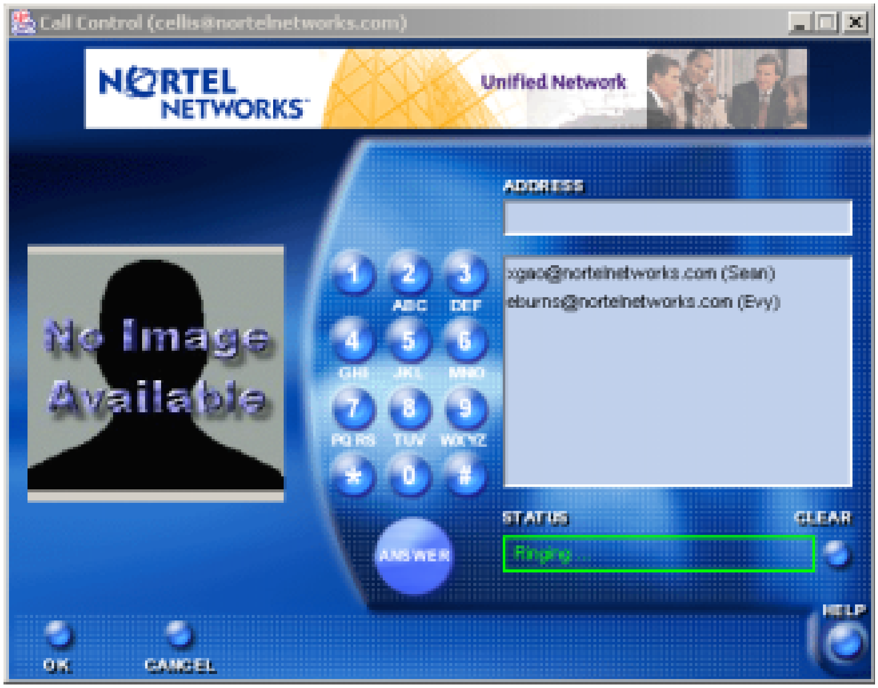
- Removes call initiation functions
- adds functions available during a call
- Provides the following options
  - Hold/CancelHold
  - Mute
  - Transfer
  - Web Push
  - Message

Redirecting the call

To redirect an incoming ringing call to another party

1. Click the **Redirect** button on the bottom of the Call Control window. The SIP Multimedia Web Client prompts you to enter the address where you want to redirect the call.
2. Enter a SIP address or select an available address from the list.
3. Click **Ok** to redirect the call.

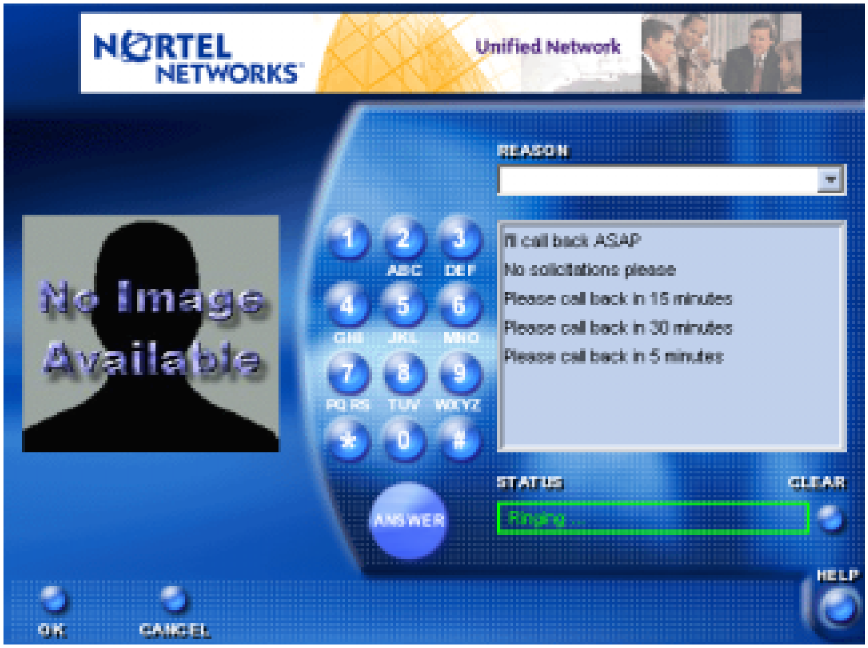
Figure 6 Redirecting an incoming ringing call



Declining the call

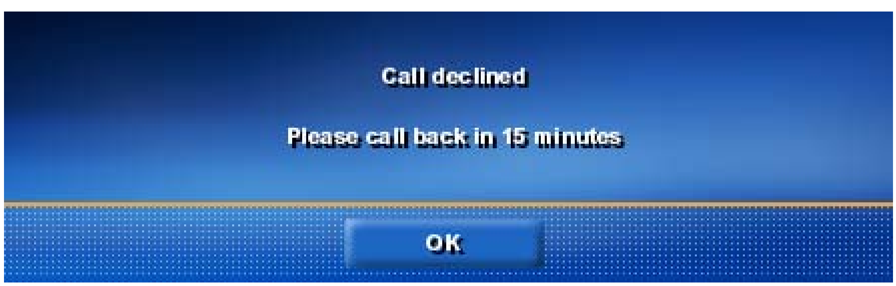
To decline answering a call:

1 Click the **Decline** button on the bottom of the Incoming Call Control window. The SIP Multimedia Web Client prompts you to select a reason.



2 Either select a reason from the list, or enter a text reason.

3 Click **Ok**.The SIP Multimedia Web Client declines the call while delivering the reason to the caller.



Ignoring a call

To pass on a call, click the **Ignore** button at the bottom of the Call Control window. The call is declined with a “temporarily unavailable” message to the caller, or passed on to the next contact number if you subscribe to a ring list using your SIP Personal Agent.

Tip: For more information about creating a ring list refer to the *SIP Personal Agent User Guide* or contact your system administrator.

Managing active call options

The SIP Multimedia Web Client provides a wealth of choices for managing the progression and interactions of an active multimedia call. During an active call, you can:

- place and retrieve a call on hold
- mute outgoing audio
- transfer the call to another destination
- push Web pages
- send Instant Messages

Figure 7 Active call options

Placing and retrieving a call on hold

To place a call on hold, click the **Hold** button at the bottom of the Call Control window. The SIP Multimedia Web Client places the active call on hold. The **CancelHold** button in the Call Control window flashes to remind you that a call is on hold. You can place another call, if you wish, without losing the first call.

To retrieve a call on hold, click the **CancelHold** button in the Call Control window for that call. The call is now active.

Muting outgoing audio

To mute outgoing audio on a call, click the **Mute** button at the bottom of the Call Control window. The **Mute** button changes to a flashing **CancelMute** button, and suppresses outgoing audio.

To restore (unmute) outgoing audio on a call, click the **CancelMute** button.The SIP Multimedia Web Client turns the Cancel Mute option back to **Mute**, and restores outgoing audio.

Transferring a call

You can transfer a call to another party directly without talking to the second party (Call Transfer) or by first consulting with them (Consult Transfer)

Call transfer

While on a call, you can transfer this call to another location without waiting for the new party to answer. The originating or terminating party can initiate the transfer, but is no longer involved in the call upon completing the “blind” transfer.

While on a call

1 Click the **Transfer** button at the bottom of the Call Control window. The Transfer dialog box displays.

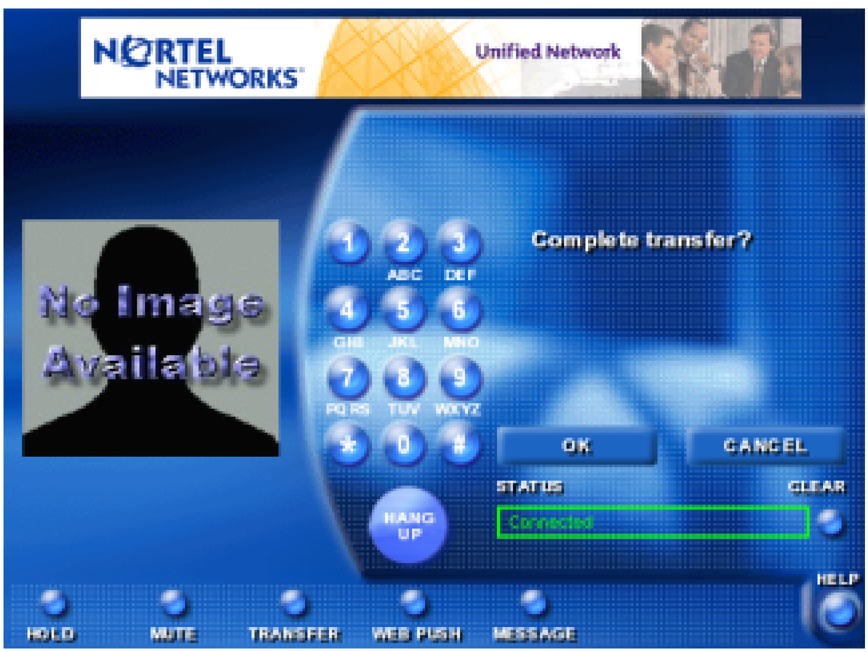
2 Enter the address of the party where want to transfer this call or, choose an entry from your address book. Click **Ok** to complete the transfer. If the transfer failed then a dialog box will pop up and prompt you to either **Retrieve** or **Hangup**.

Consult transfer

While on a call, you can call another address, speak with the person at the new address, and then transfer the original call to the new address. You must be the caller to the final destination in order to initiate a transfer with consult.

While on a call

1. Click the **Transfer** button at the bottom of the Call Control window. The Transfer dialog box displays.
2. Enter the address of the party where want to transfer this call or, choose an entry from your address book.
3. Select the **Talk to this person...** checkbox.
4. Click **Ok**. The current call is put on hold and the new call rings.
5. Consult with the party regarding the transfer of this call. A dialog box appears and prompts you to **Complete** or **Cancel** the call transfer.



6. Click **Ok** to transfer the call. If you click **Cancel** then the dialog box disappears and you can click **CancelHold** on the original Call Control window to re-connect with the call.

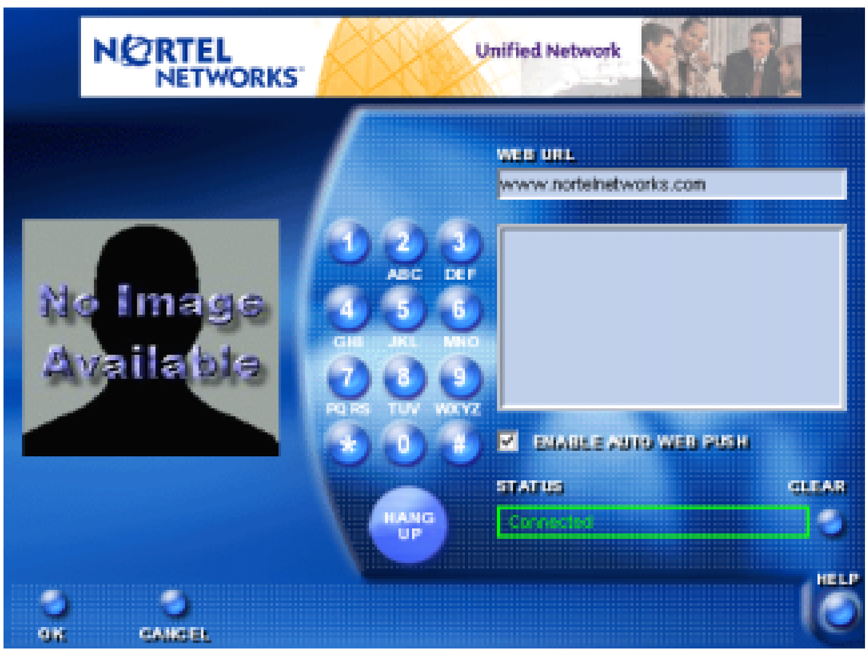
Sending Web pages

The SIP Multimedia Web Client allows you to push Web pages to display on another user's screen during an active call. You can also view Web pages pushed from another user.

Pushing a Web page

To push a Web page to another party on an active call

1. Click the **Web Push** button in the Call Control window. The SIP Multimedia Web Client displays a blank field for you to enter a Web page address.
2. Enter a URL or select one from the list.
3. Click **Ok**. The SIP Multimedia Web Client pushes the Web page to the far end computer.



Auto Web page push

To automatically push the Web pages you are browsing while on a call, click on the **Enable Auto Web Push** checkbox. The Web Push button in the Call Control window flashes to remind you that Auto Web Push is active. The other user's browser matches what you see on your browser.

Receiving a Web page push

No action is necessary to receive a Web page during an active call. The SIP Multimedia Web Client automatically launches your default Internet browser and displays the pushed Web page.

Tip: The Web push you receive should open a new browser window and not overwrite your SIP Multimedia Web Client home page (disconnecting the call). Refer to "Web browser configuration" on page 10 for the correct browser setting.

Sending an instant message

Instant messaging enables you to send and receive text notes among one or more recipients, whether or not they are on an active voice call.

You can use the Instant Message feature to send notification of issues needing urgent attention, confer with subject matter experts, respond to email inquiries, and more-even while you're tied up on a long conference call.

Tip: The SIP Multimedia Web Client maintains a running dialog of the messages, which can be copied and pasted into a text file to save.

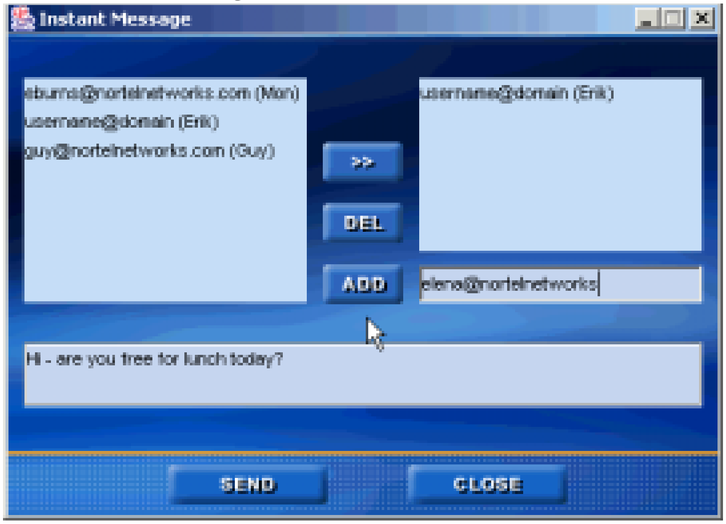
Whether you are on an active call or not you can send an instant message from the SIP Multimedia home page.

To send an instant message from the home page

1 Click the **Message** button.The SIP Multimedia Web Client displays a window for you to select recipient(s) and enter the message.

2 Highlight one or more names from the list on the left and select the arrow button or manually enter an address in the text box and click **Add**. Repeat this step for as many users as required.

3 Enter the message in the box at the bottom of the Instant Message window.



4 Click the **Send** button.

Sending an instant message to the person you are talking to

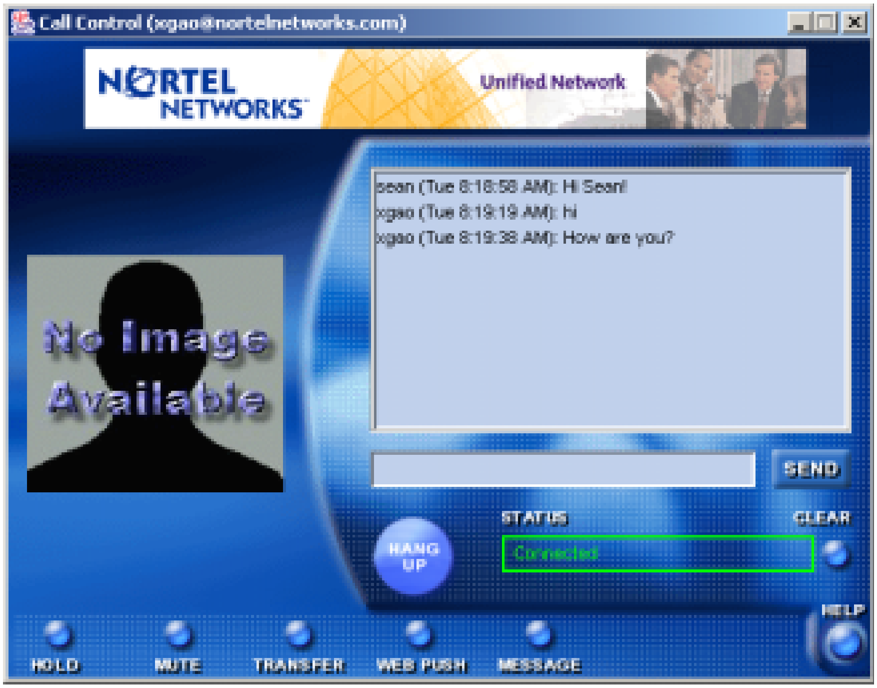
The following are examples when it would be helpful and productive to have a text chat with someone even though you are simultaneously talking on the phone:

- Your call might involve very processing-intensive activities, such as two-way live streaming video. Instant message can be a great way to converse effectively without asking your system for the additional demands of voice processing.
- Your conversation might include content that requires great specificity and accuracy that is easier and more efficiently conveyed in text rather than verbally, such as the precise spelling of proper names, or a list of figures.
- It might be beneficial to have a written record of the conversation for later reference, project tracking, or audit purposes.

To send an instant message to the person you are talking to

1 Click the **Message** button in the Call Control window.

2 Type your message in the field and click **Send**. Messages that are sent and received appear in the window.



Canceling an instant message

To cancel an instant message without sending it, click the **Close** button on the Instant Message window.

Replying to an instant message

To reply to an instant message, enter a reply in the blank box at the bottom of the Instant Message window and click **Send**. The SIP Multimedia Web Client automatically updates the dialog history window to show the full text of the instant message session.

To delete a recipient from the Send To list, highlight the name and click **Delete**.

Contacting your Buddies

From the SIP Multimedia Web Client home page

1 Click the Buddies folder in the Buddies tree. The window displays the presence information (online status) of your Buddies.

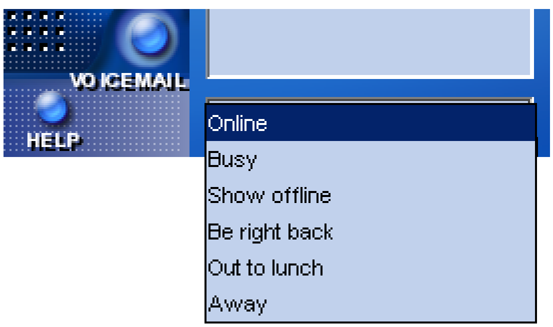
2 Right-click on an entry. The following actions are available

- make a call
- send a message

Setting your online presence status

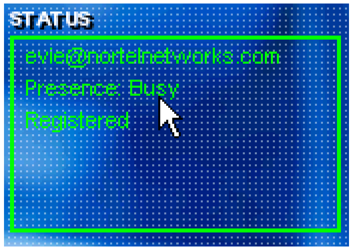
From the SIP Multimedia Web Client home page, select one of the following status indicators from the drop down list to manually set your presence information (online status):

Figure 8 Setting online presence



Your online presence displays in the status area of the SIP Multimedia Web Client home page.

Figure 9 Online status



Tip: You can set automatic presence indicators that will alert others when you are away from your PC or on the telephone. See Chapter 4, “Network settings,” on page 47 for more information.

Three-way calling

With the SIP Multimedia Web Client, you can combine two separate calls into one three-way call, with no need for a conference server.

Note: Video is not available during a three-way call.

Setting up a three-way call

To set up a three-way call

1. Make a call.
2. Place that call on hold.
3. Make another call to a different party.
4. Once the second party has answered the call (and without putting them on hold), retrieve the first call by clicking **CancelHold** from the Call Control window of the first call. You now have a three-way call.

Removing a person from a three-way call

You can choose from one of the following options to remove a person from a three-way call

- Select **Hang up** in the Call Control window of the person you wish to drop.
- Select **Hold** in the Call Control window to put the person on hold.
- Select **Transfer** in the Call Control window to transfer the person to another destination.

Conference calling

You can hold a conference with up to thirty-two participants (depending on your service package), using the resources of your network conference server.

Note: You must have network conferencing enabled in service package in order to use this feature. Contact your system administrator for more information.

Creating a conference

To set up a conference call

1. From the SIP Multimedia Web Client home page, complete a call to another party.
2. Place that party on hold.
3. Repeat this process up to the conference port limit defined in your service package. (Remember that, as the host, you occupy one port.)
4. Click the **Conference** button. The SIP Multimedia Web Client initiates a new call to the network conference server.
5. All held calls are transferred to the conference server and their Call Control windows close. A new call control window opens for this conference.

Leaving a conference

To leave a conference without disconnecting the call for others, click the **Hang Up** button in the Call Control window for the conference call.

Sending and receiving video

Tip: A video camera is required to send video. A high-bandwidth network connection and fast PC processor are recommended.

If you have a Web camera for your PC, you can easily transmit live video to your called party. If the party you are calling has a camera, you can receive their video transmission.

Refer to "Setting up video (optional)" on page 50 for video transfer. The default setup for video is disabled.

Tip: If you select **Mute** during an active audio-video call, the outgoing audio stops, but video is not affected. You will still have outgoing and incoming video.

If you select **Hold** during an active audio-video call, outgoing and incoming video is suspended and the video screen disappears. Click **CancelHold** for the video transmission to resume.

Preventing your computer from transferring video

To prevent your computer from both sending and receiving video

1. Click **Settings** from the SIP Multimedia Web Client home page.
2. Click the **Video** tab.
3. Select the **Video Disabled** radio button.

To prevent your computer from sending video (however, you will be able to receive video)

1 Click **Settings** from the SIP Multimedia Web Client home page.
2 Click the **Video** tab.
3 Select the **Receive Only** checkbox.
4 Click **Ok** to confirm your choice.

Note: Whether or not video settings are configurable is dependent on your service package.

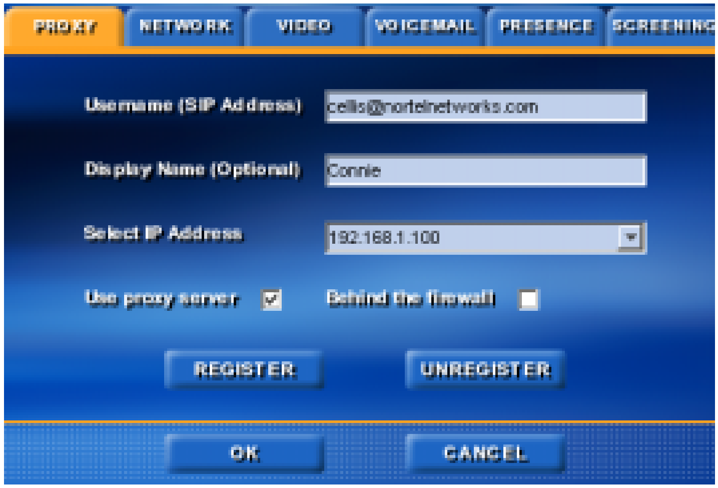
Chapter 4
Network settings

This section provides instructions for configuring settings between your SIP Multimedia Web Client and the communication network you are connected to. Topics include:

Configuring proxy and firewall settings

To configure the SIP Multimedia Web Client's proxy and firewall settings

1. Click **Settings** from the SIP Multimedia Web Client home page.
2. Click the **Proxy** tab.



3 If your SIP address is wrong then make corrections in the **Username** text box.

4 If you want to change the name that displays when you make a call then edit the **Display Name** text box.

5 Select the appropriate IP address from the **Select IP Address** pull-down menu if it is different from the default IP address the SIP Multimedia Web Client detected from the Windows networking subsystem. If public network and private network IP addresses are available, use the public IP address.

6 If your calls will be directed through a proxy (intermediary server), check **Use proxy server**.

7 If your calls will be made directly to another SIP Multimedia Web Client user through an IP address you provide, uncheck **Use proxy server**.(Optional.)

8 If your system is protected by a firewall, check **Behind the firewall**. (Optional. Contact your service provider if you are unsure of the setting.)

9 Click **OK** to confirm your choices.

Manually registering and unregistering

When you start or stop the SIP Multimedia Web Client you are automatically registered or unregistered.

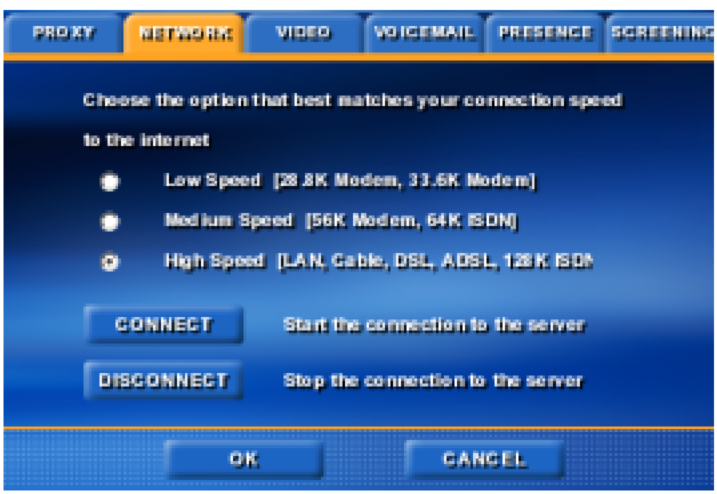
To manually unregister, or to register again without closing the SIP Multimedia Web Client home page

1. Click **Settings** from the SIP Multimedia Web Client home page.
2. Click the **Proxy** tab.
3. Click **Unregister** or **Register**.
4. Click **Ok**.

Setting the network connection speed

To specify your network connection speed to the SIP Multimedia Web Client

1. Click **Settings** from the SIP Multimedia Web Client home page.
2. Click the **Network** tab.



3. Click the speed of your computer's connection to the network: low speed, medium speed, or high speed. The SIP Multimedia Web Client selects the voice coder/decoder (codec) based on this connection speed.

Note: If you enter a connection speed higher than your actual connection, the application could overload your PC with incoming voice packets. Check with your system administrator if you're not sure what type of connection you have.

4 Click **OK** to confirm your choice.

Manually connecting and disconnecting

When you start or stop the SIP Multimedia Web Client you are automatically connected or disconnected.

To manually connect, or to disconnect the SIP Multimedia Web Client without closing the home page

1 Click **Settings** from the SIP Multimedia Web Client home page.

2 Click the **Network** tab.

3 Click **Connect** or **Disconnect**.

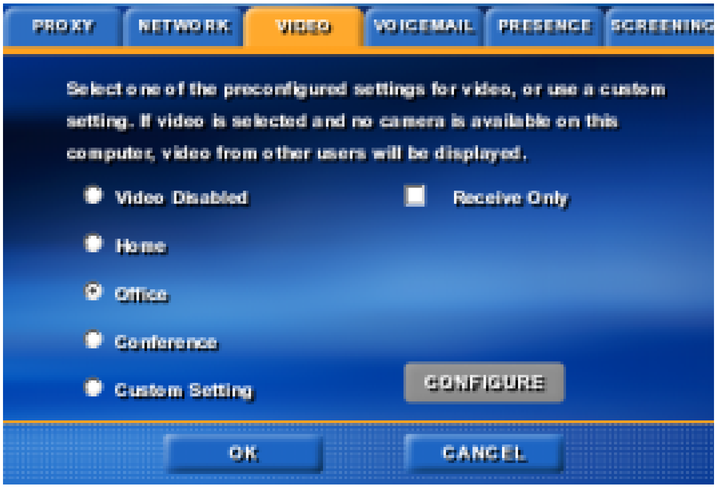
4 Click **Ok**.

Setting up video (optional)

If video is enabled in your service package

1 Click **Settings** from the SIP Multimedia Web Client home page.

2 Click the **Video** tab.



3 Select the video configuration that most closely meets your needs. The following table lists the video configuration settings.

Video configuration setting	Description
Home	enables video telephony over ISDN/DSL/Cable modem. This configuration produces a video bitrate around 60Kbps to 120Kbps.
Office	enables video telephony in an office using a 10/ 100Mbps LAN. This configuration produces a video bitrate around 150Kbps to 300Kbps.
Conference	enables video telephony in an office using a 10/ 100Mbps LAN. This configuration produces a video bitrate around 400 to 800Kbps. This video is suitable for overhead projection
Custom setting	fine tunes video performance (see below).

Note that the two SIP Multimedia Web Clients involved in a video call will negotiate between themselves and select a configuration that is acceptable to both. As such, the size of the video images may vary from call to call, depending on the configuration of the other party.

The first time you configure the video, the video camera is examined and verified for compatibility with the SIP Multimedia Web Client. If the camera is compatible, you will see a small window displaying video from the camera.

If the video camera is incompatible with the SIP Multimedia Web Client, follow the on-screen directions to configure the camera.

If no video camera is attached to the PC when you configure the video, a "No camera available" message displays the first time you select the **Home**, **Office**, or **Conference Room** setting. The message indicates that although video cannot be sent, you will receive video from other SIP Multimedia Web Clients that send it.

4 Click the **Ok** button to close the Settings window.

After the video camera has successfully passed compatibility testing with the client, video will be sent and received on all calls to other video-enabled SIP Multimedia Web Clients.

Disabling video transmission

To disable video transmission from your camera-equipped PC

1 Click **Settings** from the SIP Multimedia Web Client home page.

2 Click the **Video** tab.

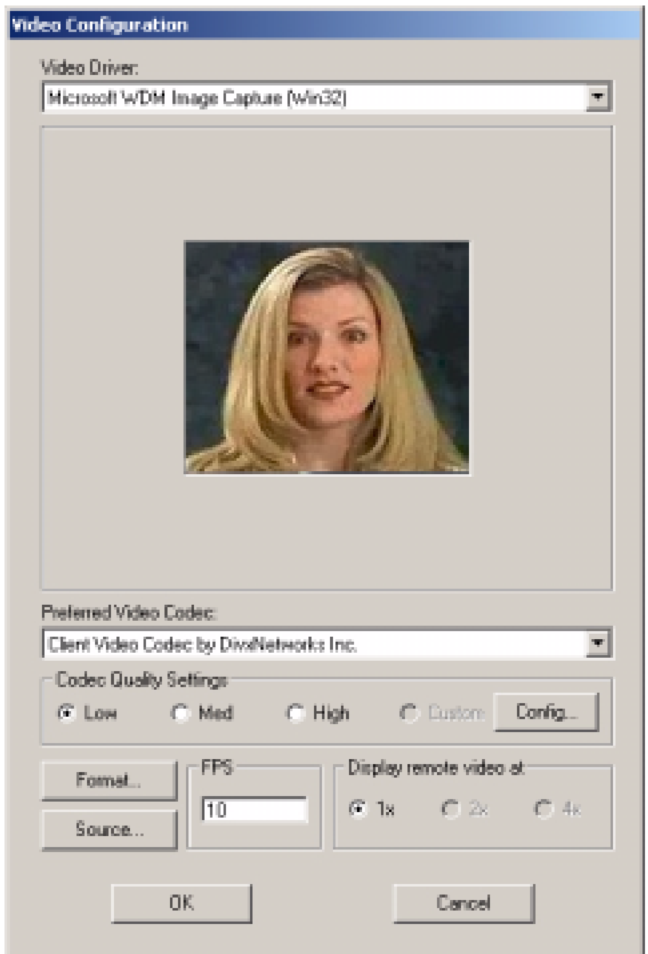
3 Select the **Receive Only** checkbox. This restricts the client to only receive video from other video-enabled clients.

If video has been enabled for all calls (either bi-directional or receive-only), video can be wholly disabled by selecting the Video Disabled configuration. Re-enable video by selecting the Home, Office, or Conference settings again.

Custom video configuration

1 Click the **Custom Setting** option on the Video tab.

2 Click the **Configure** button. The custom Video Configuration window displays to fine tune your video settings.



Note: Custom video configuration should be done with care. The combination of a high quality codec setting and high FPS produces video transmission rates over 1Mbps and erodes network and PC performance. You are encouraged to use the “Conference,” “Office,” or “Home” configurations for everyday use.

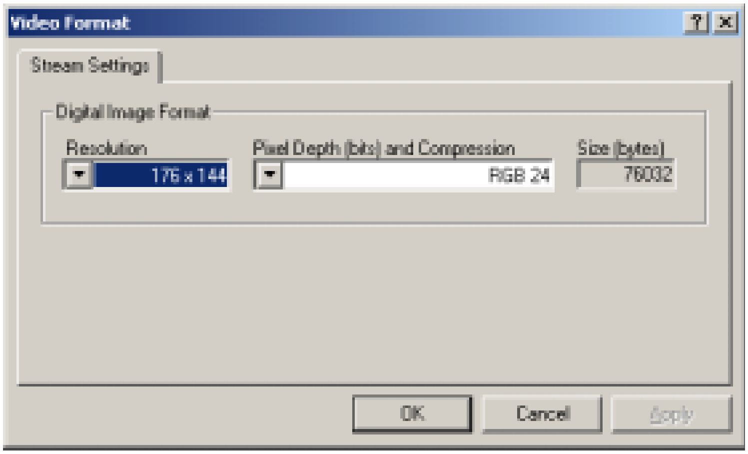
This custom Video Configuration window has the following controls:

- **Video Driver** drop down menu -- identifies the driver which controls the camera. This is usually “Microsoft WDM Image Capture,” but some cameras provide a different one. Other items on this menu allow you to:
  - disable video for all subsequent calls.
  - set video to receive-only operation on all subsequent calls
  - reset the video configuration to its never-been-set-up default values.
- **Preferred Video Codec** drop down menu -- should be set to “Client Video Codec by DivXNetworks, Inc.”
- **Low, Med**, and **High** quality settings -- allow you to specify the amount of detail in the transmitted video. High quality transmits the most detailed images, but at the expense of CPU and network bandwidth. The use of **Custom** quality settings is strongly discouraged unless you are highly knowledgeable. Note that the two SIP Multimedia Web Clients in a video call negotiate to a common video quality that is acceptable to both, so delivered image quality may vary from call to call.

Note: There is no tool or formula that allows you to compute the network bandwidth (in bits per second) for an arbitrary combination of video settings. However, network bandwidth increases with increasing quality.

- **Format** button -- produces a dialog box that specifies the size and internal organization of the video image. The dialog box typically appears as shown in the following figure (it will vary from camera to camera).

Figure 10 Video format dialog box



You can view, but not necessarily manipulate, the following controls:

— **Resolution**. This specifies the preferred size of the images transmitted during a video phone call, only 176x144 is supported.

— **Pixel Depth and Compression**. This specifies the organization of the video data captured by the camera. The SIP Multimedia Web Client supports two: **RGB 24** and **I420**, although others may work. If the Client Video Codec is unavailable under the **Preferred Video Codec** dropdown menu, you should try to adjust the Pixel Depth and Compression setting.

- **Source** button -- produces a dialog box that allows the you to:

  — Select a camera to use, if more than one “Microsoft WDM” camera is attached to the PC.

  — Adjust the camera’s color balance, brightness, contrast, and color saturation, among other settings.

  The **Video Source** dialog box usually varies greatly from camera to camera.

- **FPS** field -- allows you to specify the number of Frames Per Second that the client should transmit. Higher numbers increase the fluidity of motion, but at a cost of greater CPU and network bandwidth. A value of 15 produces a quite an effective sense of motion.
- **Display Remote Video at** buttons -- shows the 1x scaling factor on received video. Smaller screens disable these scaling factors as appropriate.

3 After updating video settings and returning to the **Video** tab, click **Ok** to save the changes.

Setting up voicemail access (optional)

If voice mail is enabled in your service package

1 Click **Settings** from the SIP Multimedia Web Client home page.

2 Click the **Voicemail** tab.

3 Enter the phone number of your voice mail system, without punctuation. Remember to enter any extra digits required by your organization's phone network.

4 Enter the Mailbox ID and the pound (#) sign, if required.

Tip: After auto-dialing your voicemail system number, by default, the SIP Multimedia Web Client waits three seconds before dialing your mailbox ID, then waits another two seconds before dialing your password. If you need to increase the delay to match the timing of your voice mail systems prompts, add one or more commas before the mailbox ID and/or password value. Each comma represents an additional one-second delay.

---

5 Enter the numeric function commands your voice mail uses to perform standard functions, such as playing and deleting messages and sending replies. The SIP Multimedia Web Client uses this information to support voicemail functions.

6 Optionally, define up to two custom voice mail functions and the keystrokes that activate them.

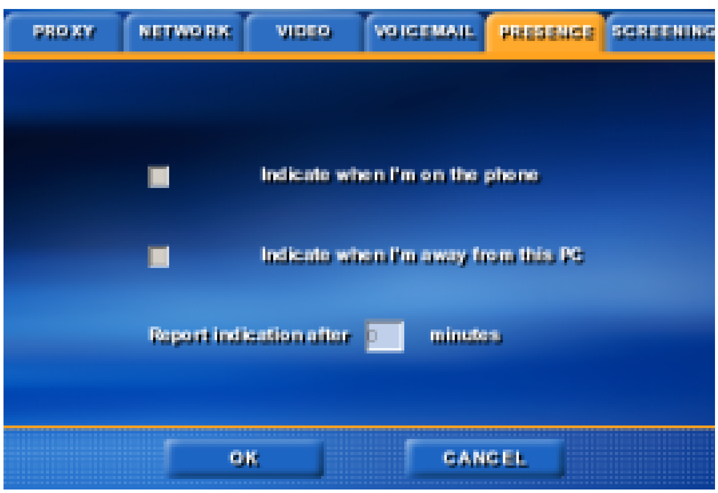
7 Click **Ok**.

Tip: Whether these PSTN voice mail settings are changeable depends on the type of service package assigned to you by your service provider.

Setting up presence (optional)

The SIP Multimedia Web Client provides a drop-down list, located under the Buddies/Address tree, for you to manually indicate your online status. Alternatively you can set your SIP Multimedia Web Client to automatically update your presence status to show when your PC is idle, or your telephone is occupied.

Figure 11 Presence settings



To set up your presence indicators

1 Click **Settings** from the SIP Multimedia Web Client home page.

2 Click the **Presence** tab.

3 Choose either or both presence options.

4 Enter the number of minutes that must elapse before your status is reported.

5 Click **Ok**.

Tip: Your system must support presence and you must be online for your presence to be detected by others.

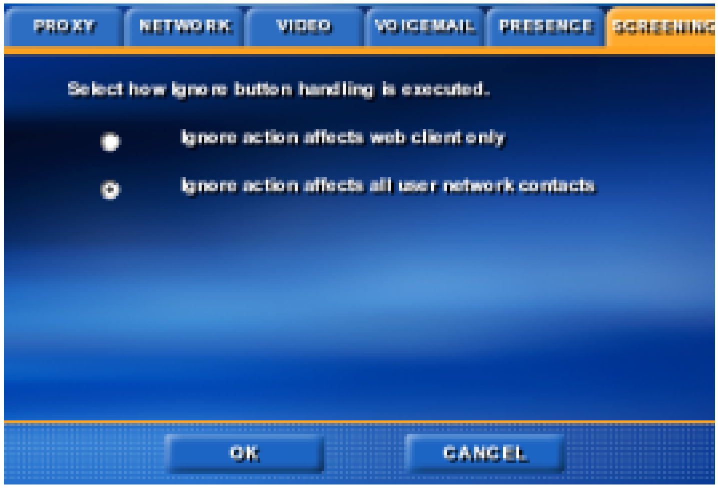
Setting up screening (optional)

You can choose how the **Ignore** softkey behaves for your incoming calls. The choices are as follows:

- **Ignore action affects web client only -** when you press the Ignore softkey in the incoming call control window then:
  - — the call stops ringing on the Web Client, however, if you have defined other devices in your first ring list then they will continue to ring (for example, you might configure both your Web Client and mobile telephone to ring simultaneously)
  - — the call control window disappears and you do not have the option to answer the call with the Web Client, although you can retrieve the call with other defined devices
  - — the call continues to ring (for the caller) for the specified number of rings before it passes to the next ring list entry
- **Ignore action affects all user network contacts -** when you press the Ignore softkey in the incoming call control window then:
  - — the ringing stops for all defined devices in your first ring list
  - — the call control window disappears and you do not have the option to answer the call with any device in your first ring list
  - — the call automatically passes to the next ring list entry

Tip: For more information about creating a ring list refer to the *SIP Personal Agent User Guide* or contact your system administrator.

Figure 12 Screening settings



To choose a screening option

1. Click **Settings** from the SIP Multimedia Web Client home page.
2. Click the **Screening** tab.
3. Choose one of the screening options.
4. Click **Ok**.

Appendix A
Hardware notes

Topics in this section

Compatible video cameras

The SIP Multimedia Web Client requires video cameras that capture video in RGB-24 or I420 video format, and the vast majority of USB 1.x web cameras meet these requirements.

Note: Nortel Networks can make no recommendation or statement of compatibility about cameras that will work with the SIP Multimedia Web Client on an individual user's PC. There are too many issues out of Nortel Networks' control and influence for any concrete recommendations to be made.

Issues which may influence the operation of a camera are:

- hardware revision of the CPU, CPU chipset, and motherboard
- software revision of CPU chipset and motherboard device drivers
- release and revision of the Windows operating system
- hardware revision of the camera
- software revision of the camera drivers
- the presence of other user-installed devices, USB or otherwise, which were previously installed on the user's PC. Other devices may cause issues regardless of whether they are still present or not.

- the installation of other software packages on the user's PC.

Compatibility with the SIP Multimedia Web Client

The responsibility of ensuring compatibility of the camera with the client application is critical. Compatibility is usually indicated by; successfully installing the camera, seeing the camera recognized by the SIP Multimedia Web Client application, and proper behavior of the SIP Multimedia Web Client application during and after several video telephony phone calls.

The following guidelines are recommended:

- Evaluate the camera in person before purchasing.
- If multiple PC's with different versions of the Windows operating system are going to be used with the camera, evaluate the camera on all operating systems before purchasing.
- If multiple PC's with different hardware configurations are going to be used with the camera, evaluate the camera on all hardware configurations before purchasing.
- Before installing a camera on a computer, always visit the camera vendor's web site for updated camera drivers and use the updated drivers if available.
- Do not purchase USB cameras for use on computers with the Windows NT operating system. Windows NT does not support USB devices.

Appendix B Troubleshooting

This section provides information about how to troubleshoot the following problems:

Start-up and configuration

The SIP Multimedia Web Client should automatically start and run if you have the correct system requirements. See the sections below for some troubleshooting tips.

Start-up or configuration problems

Check the following:

- Exit your browser (**File**>**Exit**) and restart.
- Check that your browser is correctly configured. see "Web browser configuration" on page 10.
- Check that you have the minimum hardware and operating system requirements. "Minimum hardware and operating system requirements" on page 9.
- Otherwise, contact your system administrator.

Connections speeds

The SIP Multimedia Web Client negotiates the best speed that can be supported by both users. It is not a requirement that connection speed settings match between two clients.

Audio

Audio settings may need to be fine-tuned depending on the type of equipment you have.

Echo

If you use a desktop microphone and speakers as your sound input/output devices, the sound from the speaker is often heard by your microphone and the person on the far end will hear an echo. This is why the use of a headset or handset with your PC is recommended. The use of headphones (without a microphone) along with your desktop microphone will also work.

Sometimes echo occurs even when using a headset. Usually, a quick adjustment of volume can fix this. Try lowering the speaker volume and microphone gain.

Some sound cards have input mixing capabilities. To see if your card supports this:

1 Launch the Windows volume control application via **Start** > **Programs** > **Accessories** > **Multimedia** > **Volume Control**.

Note: Your version of Windows may have a different path to the volume control. Check under the Windows Help menu to find it.

2 Select **Options** > **Properties**.

3 Select **Adjust Volume for Recording** and click **Ok**. If Mixed Input is checked, you can experience echo even when using a headset.

4 Uncheck the **Mixed Input** checkbox and see if the echo has disappeared. (Note that not all sound cards have this feature.) Muting the Wave on the recording settings may also help with this problem.

No voice

Make sure no other audio connections are running. If another application is using your sound card, the SIP Multimedia Web Client may not be able to access it.

- Check to make sure your volume settings are correct.

Note: Due to sound card conflicts, you cannot run the SIP Multimedia Web Client and the SIP Multimedia PC Client at the same time.

Chipmunk (distorted) sound on my calls

Your sound card drivers may not be completely compatible with the SIP Multimedia Web Client. The following table lists the recommended driver versions for the NeoMagic* card for the various Microsoft Operating Systems.

These cards are especially prevalent on Dell computers. These driver files can be downloaded from the www.dell.com website.

The Readme.txt file has instructions on how to install the drivers, or you can simply double-click on the **setup.exe** file and it should do everything for you. You'll be required to do a restart. Also, for a laptop, it is recommended to perform this upgrade while the computer is undocked.

Table 3 NeoMagic sound card drivers

Operating System	Recommended driver versions
`Windows 98`	`NMAUDIOD.DRV - 4.05.4204.0030Q` `NMAUDIOV.VXD - 4.05.4204.0030Q`
`Windows NT`	`NMXNT32.DLL - 4.03.00.2041` `NMXNT.SYS - 4.03.00.2041`

Table 3 NeoMagic sound card drivers

Windows 2000	NM5A2WDM.SYS - 5.00.2144.1 WDMAUD.DRV - 5.00.2184.1
Windows XP	NM6WDM.SYS - 5.1.2461.0 WDMAUD.DRV - 5.1.2481.0

I am using a headset and cannot speak when the other party is speaking

You may not have a full duplex sound card (some laptops default to half duplex mode).

Cannot hear or talk with headphones

Try plugging your headset directly into your laptop, instead of into the docking station.

Calling and messaging

Some of the enhanced features of the SIP Multimedia Web Client may not be supported on your network.

Cannot complete call

Connection to this number may not be supported by your network. Contact your support number for more information.

Web Push does not work

Make sure you are using Netscape version 4.77 or later, or Internet Explorer version 5.x.

No Picture Caller ID

The network may not have a picture stored for the caller.

Picture Caller ID not transmitting

The network may not have a picture stored for you. Use the SIP Personal Agent to check your network settings. Contact your system administrator for information on how to access your SIP Personal Agent (this is an optional program that may not be available.)

Video

Video cameras and video settings may require fine-tuning to optimize the quality of the transmission.

Blurry video

Most video cameras have a focus ring to adjust the image. In a call where you are transmitting video, click the video preview checkbox (or select the 1x button) to see your transmitted image. Turn the focus ring (it usually encircles the lens) until the image is sharper.

Poor color/contrast/brightness

Most video cameras allow the user to tune these settings. Terminate any active video call, then do the following:

1. Go to **Settings** and the **Video** tab, and click the **Configure...** button.
2. If video is not being displayed in the Video Configuration dialog, select your video camera from the **Video Driver** dropdown menu (it is probably “Microsoft WDM”).
3. Press the **Source...** button after video is displayed. This will usually produce a multi-tab dialog that has controls to tune video color and brightness. Operate the controls until you are satisfied with the image.

4 Click **Ok**.

Adding a new video camera

First, make sure that the SIP Multimedia Web Client has permission to perform video telephony. Go to **Tools**, **Preferences** and the **Video** tab. If all of the controls are grayed out and non-operational, you may not have logged in to the network (use the Login menu). If you have logged into the network (that is, if there is a green light on your Login menu), your service profile does not include video. Please contact your service provider or administrator.

If the video controls are enabled, make sure that the client has been configured to send/receive video by verifying that the Video Disabled selection is NOT selected. Just click on another setting to set up video if video was disabled.

“No Codecs” message

The camera may need to be manually configured

1. Click **Settings** and the **Video** tab, and press the **Configure...** button.
2. If video is not being displayed in the Video Configuration dialog, select your video camera from the **Video Driver** menu (it is probably “Microsoft WDM”).
3. Press the **Format...** button after video is displayed. The dialog which comes up varies from camera to camera, but look for a control for “Pixel Depth / Compression” or “Format”. Choose either RGB 24 or I420 in this control, and press **Ok**.

   If “RGB 24” or “I420” is not listed as a selection, the camera may not be usable by the client. However, try examining other programs that came with the camera in order to enable Video For Windows with this camera. Also, check with the camera vendor’s web site to obtain the most recent drivers.

Camera switches to receive-only

The SIP Multimedia Web Client will switch to receive-only video if it starts up and cannot locate a previously found camera. To transmit video again, go to **Settings** and the **Video** tab, disable video, then select your video configuration again.

Blue screen error

This is caused either by multiple cameras/video capture devices corrupting each other's installations or by buggy device drivers. Try to uninstall unneeded video devices and go to the vendor's web site and ensure that you are running the latest drivers for the video device. If you are unsure of how to troubleshoot device installation conflicts, seek knowledgeable assistance.

Index