



NetVanta Unified Communications

NetVanta Business Application Server

Administrator Guide

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Conventions

**NOTE**

Notes provide additional useful information.

**CAUTION**

Cautions signify information that could prevent service interruption or damage to equipment.

WARNING

Warnings provide information that could prevent injury or endangerment to human life.

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1 Introducing the Administrator Application

This chapter introduces the NetVanta UC Server Administrator application and presents some basic operations. This chapter covers the following topics:

- *Overview on page 9*
- *Conventions on page 9*
- *Product Features on page 10*
- *About the NetVanta UC Server on page 11*
- *Starting the Administrator Application on page 14*
- *Changing the Password on page 15*
- *Navigating the NetVanta UC Server on page 15*
- *Customizing the NetVanta UC Server on page 16*
- *Installing the NetVanta UC Client on page 18*

Overview

If your login authentication gives you access to a System profile (such as Administrator), you can configure the UC server according to the needs of your workplace. You can configure autoattendants and ports, add and configure SIP phones, and create user profiles for additional users on your network. Install the UC server software and manage the system using the Administrator tab.



For more information about the UC server, refer to the ADTRAN website at <http://kb.adtran.com>. Here you will find training, documentation, and technical notes.

Installing the UC server on a Windows operating system allows you to manage the business communications system within a Microsoft IT environment. Integration with Active Directory allows IT administrators to manage the communications system using the same user accounts and security policies that are used in the Windows environment. The automatic integration requires no extra programming or special integration.

Conventions

Text

Bold font is used to indicate important input data and keywords.

Shortcut Menus

Steps for procedures generally instruct readers to use shortcut menus, although commands can also be made using menus and toolbar buttons.

To obtain a shortcut menu, right-click the icon or folder you want to access then select the command you want.

Voice recording conventions

Computer audio card device



Telephone handset device



You can select the icon of the audio device when you play or record. The application saves the setting and automatically uses the last device that you selected.

Record button



Stop button



Play button



Product Features

The following are some of the many UC server features:

- Support for Windows Server 2008
- Support for Microsoft Exchange Server 2007 is available automatically through the Server Configuration Wizard
- Client software supports Microsoft Windows Vista and Microsoft Office 2007
- Support for connecting existing PBXs to the UC server through a SIP gateway (refer to [Generic PBX \(via Gateway\) on page 150](#))
- Ability to use a customized application and the OutDial executable to make outgoing calls and run services on connected calls (refer to [Using the Outbound Dialer on page 197](#))
- When importing users from text files you can specify the call answering mode, either Personal Assistant (PA), or Personal Business Assistant (PBA)

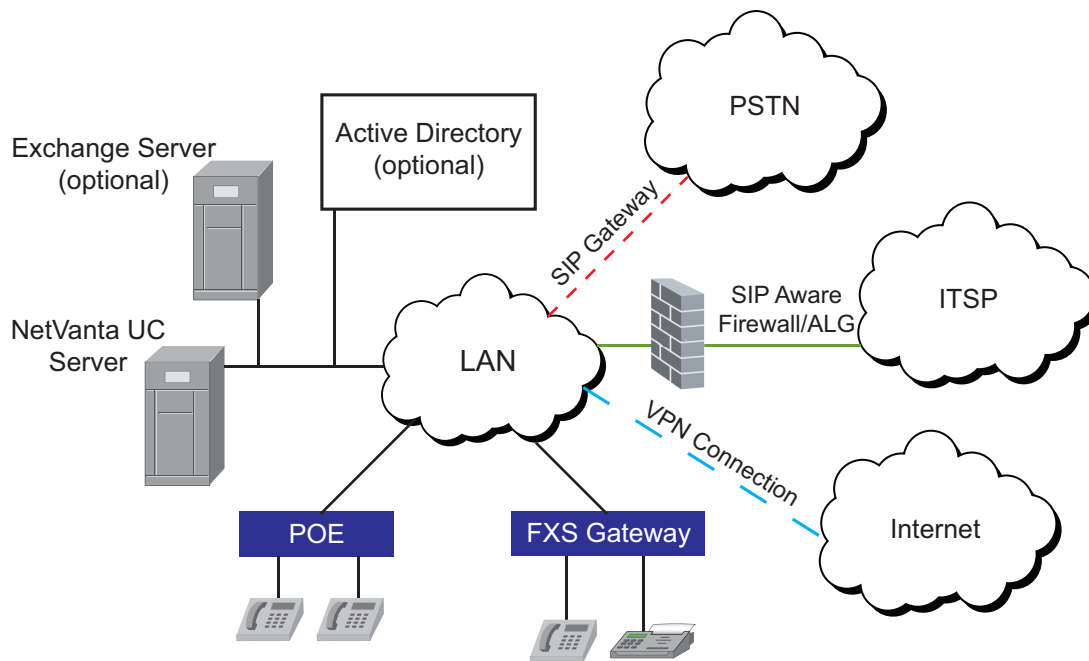
- Automatically populates the Active Directory field with the primary identity address (for example, an extension number)
- Active Directory management of a user's phone
- Interactive Voice Response (IVR) elements including *Move Current Row element on page 307* for selecting information from multi-record results from database searches, and *Prompt for Current Row element on page 308* to allow callers to select a result from a set through automated prompts
- Ability to sort database data from elements in a service (refer to *Sorting data rows on page 302*)
- The service code and service expiry date are available from the About box (refer to *Accessing Technical Support on page 327*)
- Support for *ITSP Gateways on page 132*
- Improved gateway management (refer to *Configuring Gateways on page 133*)
- Dial plan improvements (refer to *Routing calls using the dial plan on page 43*)
- Importing users from text files supports importing phone information, resulting in faster on site deployment (refer to the NetVanta UC Server *Configuration Guide, Users section*)
- Ability to restart or reboot phones from the Administrator user interface (UI) (refer to *Restarting phones and reloading phone configurations on page 127*)
- SIP phone user guides, available in the UC server help files, on the installation CD, and on the ADTRAN website at <http://kb.adtran.com>.
- Ability to change phones during a call using the connected call pickup feature (refer to *Changing phones during a call on page 164*)
- ucCompanion Console application provides phone, call, and availability status information at a glance. Refer to the ucCompanion documentation for more information.

About the NetVanta UC Server

The UC server is a combination of many applications:

- Session Initiation Protocol (SIP) communication system, including SIP registration and SIP proxy Back-to-back User Agent (B2BUA)
- Core application service—unified communications services (functioning as an attached voicemail and unified messaging server), that execute both system and user behaviors, as well as integrate with the messenger server. This includes support for Personal Assistant and Personal Business Assistant.
- Integration with traditional communication system
- Unified messaging plug-ins
- Administration managed through standard Microsoft tools

NetVanta UC Server integration



The UC server provides advanced business communications. A System profile allows you to configure all aspects of the business communications environment.

The SIP telephony environment allows you to integrate any hardware that conforms to the SIP standard, for example, SIP telephones, SIP gateways, and SIP trunking to internet telephony service providers (ITSPs). SIP telephones (hard phones and soft phones) allow users to make and receive telephone calls. SIP gateways connect to the phone company, also known as the public switched telephone network (PSTN), and allow users to make calls to people outside your company. Some examples of PSTN connections are FXO-loop start, PRI (primary rate interface for ISDN connections), or T1. In addition, SIP trunking can connect to ITSPs through a SIP aware firewall or application layer gateway (ALG), which then connects to the phone company (PSTN) over the Internet.

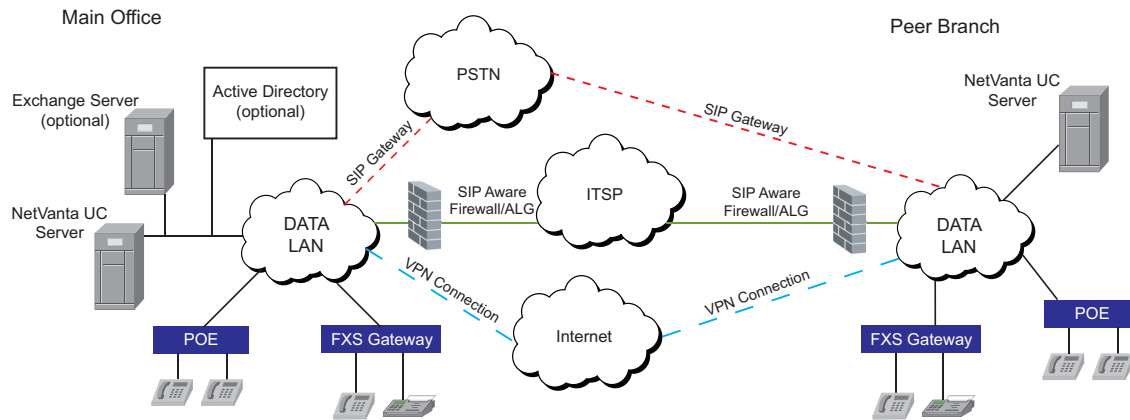
SIP telephones and gateways work within an existing network, including networks that have a Dynamic Host Configuration Protocol (DHCP) server for assigning IP addresses to SIP devices. You can configure specific options on the DHCP server to allow network devices to retrieve information, such as a File Transfer Protocol (FTP) server for downloading files.

You might require some other network services to support the automatic configuration of the devices. For example, FTP or Trivial File Transfer Protocol (TFTP). The UC server provides a SIP registrar and SIP B2BUA proxy functionality, which allow endpoints to register securely to facilitate routing to other SIP devices or advanced communications services, like unified communications, personal assistants, and voicemail access.

The UC server integrates with other applications within your corporate environment, such as email, in the case of Microsoft Exchange.

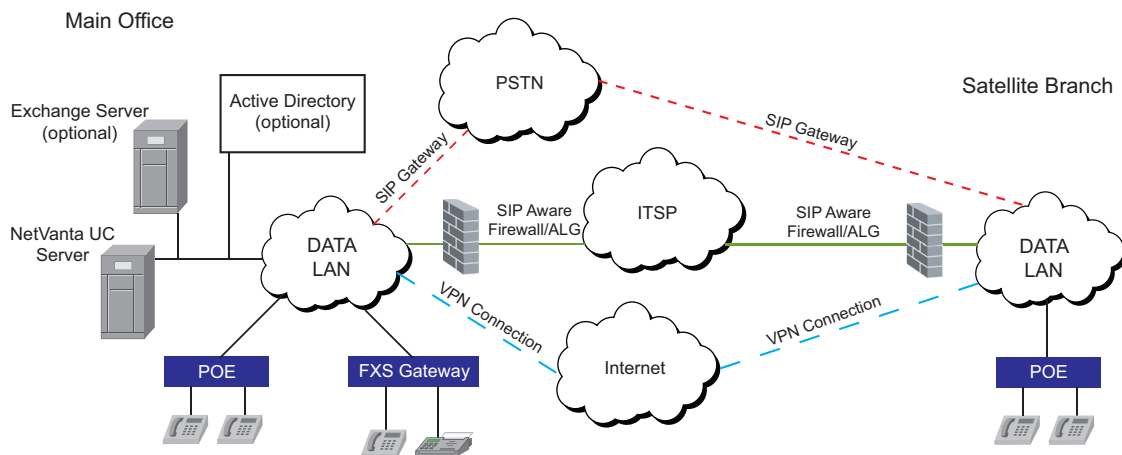
How your site is configured depends on your operational requirements, including the number of users, and the number of office locations. Some possible deployment scenarios are described below, and various options can be combined to accommodate your organizations needs.

Peer branch office deployment



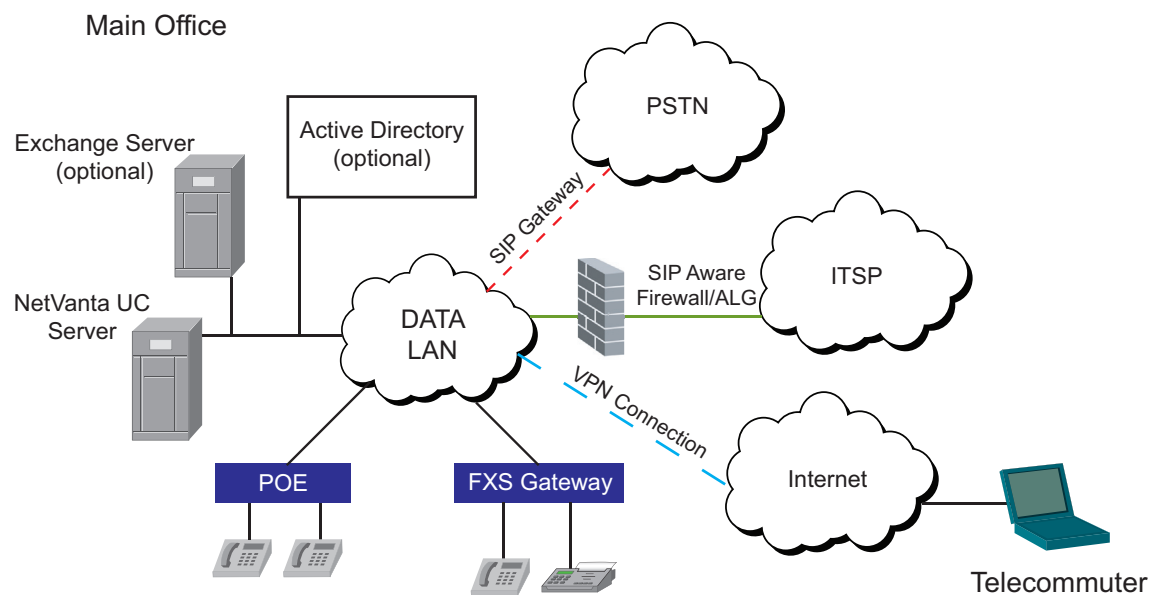
A peer-to-peer configuration might involve two UC servers, each configured to a different branch. Using the dial plan and call routing, the UC servers can connect and communicate with one another through the PSTN, an ITSP, or through a virtual private network (VPN).

Satellite branch office deployment



A satellite branch office implementation might involve one UC server at one location, and using the same UC server to connect to a different physical location at another branch.

Telecommuters deployment



The UC server can accommodate telecommuters using a variety of methods. For example, telecommuters can check their voice and email messages remotely using the telephone user interface (TUI) or through Web mail, using either a VPN connection or a SIP aware firewall over the Internet.

NetVanta UC Server integration with a legacy PBX system

UC servers can also connect to a traditional telephone system as a full service unified communications server, but with call routing managed by the existing telephone system.

Starting the Administrator Application



The UC server client must be installed. The client software must match the software version of the UC server.

To start the Administrator application

1. Select Windows **Start > Programs**.
2. Select **UC Server Client**.

3. In the Choose UC Server dialog box, type the name of a server to which to connect.



*If the server and the application are on the same computer, you can type **localhost** instead of the full name.*

4. Enable the Save server as default check box.
5. Select **OK**.

Changing the Password

If your user profile menu has access to multiple profiles, you can view those profiles as tabs across the top of the window. Make sure you have selected the correct profile for which you want to change the password.



*The default administrator login name is **admin**.*

To change the password

1. Select the tab of the profile for which you want to change the password.
2. Select **Tools > Change User Password**.
3. In the Change Password dialog box, do all of the following:
 - In the **Old password** box, type the old password.
 - In the **New password** box, type the new password.
 - In the **Confirm new password** box, retype the new password.The new password takes effect when you restart the application.

Navigating the NetVanta UC Server

In the Administration application, the **Admin** tab is available when you start the application. From the Administration navigation pane you can manage the following:

- **Users** (user profiles)
- **Phones**
- **Paging Groups**
- **Gateways**
- **Ports**
- **Authentications**
- **Identities** (typically extensions)
- **Data Sources** (if you are licensed for database integration, then data sources can be used in custom call

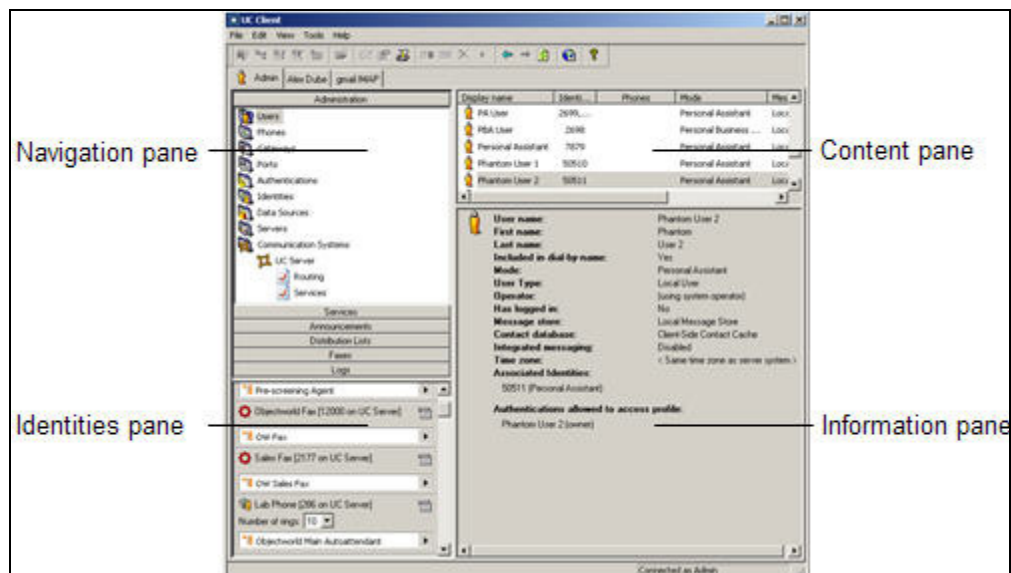
answering services)

- Servers
- Communication Systems

The administrator application is divided into the following four quadrants: the navigation pane, the content pane, the identities pane, and the information pane. Selecting an item in the navigation pane, for example **Users**, opens the information in the content pane, for example a list of users. Selecting an item in the content pane, for example a particular user, displays more information about the selection in the information pane.

The identities pane displays information about the identities associated with the administrator application. From here, you can change the default number of rings before a call is directed to voicemail, locate and edit information about services, and retrieve user login and password information for SIP authentication.

When you select the **Admin** tab, the following menu appears:



Customizing the NetVanta UC Server

Configuring system call options

You can configure system-wide call options, such as playing background music when a caller is on hold.

You can also configure what callers hear if an error occurs during a call. When the **Play warning prompts** mode is enabled, then a voice prompt plays to tell the caller that the system is unable to complete the call. When this mode is disabled, the caller hears a tone pattern when a call is not completed.

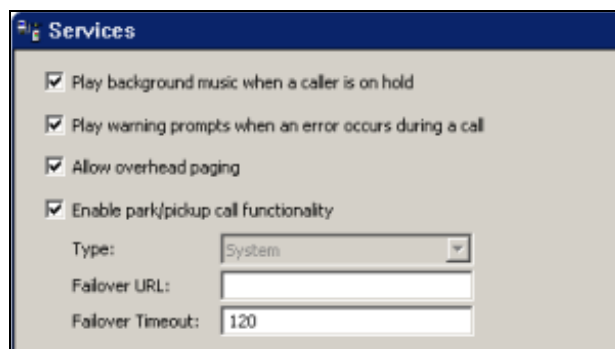
You can set the call **park** or **pickup** communication system feature, which allows a user to park a caller on their phone and allows another user to pick up that caller on their phone.

The two modes for this feature are **System** and **Extension**.

System The **System** mode applies the services options to a attendant service. When a call is parked (*99) with this mode selected, every call receives a unique park identifier. The park ID is announced to the user parking the call at the time the call is parked (for example, extension 800 parks a call, and to pick up, the user dials *98 <unique identifier>). **System** is the default mode.

Extension The **Extension** mode applies the services options to an extension service. When a call is parked (*99) with this mode selected, the call is labeled with the extension parking the call (for example, extension 800 parks a call, and to pick up, the user dials *98 800).

- Choose **System** park if you expect users to want to park more than one call at a time.
- Choose **Extension** park if there is only one call parked at a single extension at any time.



To configure call options

1. In the Administration navigation pane, double-click **Communication Systems** to open the communication system options.
2. Double-click the communication system for which you want to configure services options.
3. Select **Services**.
4. Enable any of the following check boxes:
 - **Play background music when a caller is on hold**
 - **Play warning prompts when an error occurs during a call**
 - **Allow overhead paging**
 - **Enable park/pickup call functionality**



*If a caller places a call on hold and then parks the call, the other party hears background music, regardless of whether **Play background music when a caller is on hold** is enabled.*

5. From the **Type** list, select one of the following:
 - **System**—users are able to park more than one call at a time
 - **Extension**—users can park only one call at an extension at any time

6. Type a URL in the **Failover URL** box. The call rings back at this extension after the failover timeout period elapses. If this box is blank, the call rings back to the extension that originally parked the call.
7. Type a time out in the **Failover Timeout** box. The default time out is **120** seconds.

Installing the NetVanta UC Client

Installing the UC server client application on a user's computer, provides the following capabilities:

- Fax printer driver enables sending faxes from a user's desk
- Greeting management
- One time messages
- Assistant options (for example, transferring callers and notification services)
- Personal settings for message delivery, mailbox options, client time zone settings, and so forth
- For the Personal Business Assistant (PBA), the ability to manage services and announcements

To install the application

1. Insert the NetVanta UC Server CD-ROM into the CD-ROM drive of the user's computer or browse the shared directory where it is located.
2. On the desktop, open **My Computer** and double-click the CD-ROM drive icon. The auto-run utility starts.
3. Select **Install UC Client**.
4. Double-click **Setup** to obtain the **Welcome** menu.
5. Select **Next**.
6. Select **Yes** if you accept the Software License Agreement. The **Choose Destination Location** menu appears. The default destination folder is:
C:\...\NetVanta UC client
Select **Browse** to select a different destination folder.
7. Select **Next** and the **Setup Type** menu opens.
The default setup is **Typical**.
8. Select **Next** and the **Select Program Folder** menu opens.
The default program folder is ADTRAN UC server.

2 Setting System-Wide Options

The UC server Administrator options allow the administrator to configure the behavior of the system. This chapter explains how to set system-wide options. Many of the system-wide options can be set by selecting the **Tools > Options** menus, and selecting one of the tabs available. Options fall under the following categories:

- [Setting User Options on page 19](#)
- [Setting Auto attendant Options on page 29](#)
- [Setting Outbound Calls and Dialing Options on page 32](#)
- [Setting Administrative, Maintenance, and Diagnostic Options on page 36](#)

To access NetVanta UC Server Administrator options

1. From the UC server Administrator application select **Tools > Options**.
2. Select from the tabs available to change system-wide options.

Setting User Options

There are a number of system-wide user options to be set to define the default user behavior.

- Message delivery options
- Message waiting light options
- Message playback order
- Voice mail options
- Assisted transfer options
- Fax message delivery options
- Fax-on-demand options
- Message options

Setting message delivery options

Message delivery options apply to the Active Message Delivery and pager notification features that are available to Personal Assistant (PA) users.

When Active Message Delivery is in use, you can specify the maximum number of concurrent calls, and the number of seconds to wait after connecting before prompting the caller. The default value for the maximum number of concurrent calls is 1. Increasing this value increases the number of simultaneous calls that can be made. However, setting the value too high reduces the number of ports available for other activities such as auto attendants, calls forwarded to voicemail, outgoing Active Message Delivery, and on some PBX integrations, message waiting indicators. You must also consider the number of circuits or trunks when setting this value, to ensure that you have enough PSTN gateways/trunks that can be used at any one time.

For the Pager Notification feature, you can specify the number of attempts to deliver messages. You can also specify the interval between successive delivery attempts.

For the Sending Email field, you can select the email service that is used to send notifications by email.

You can set the **Message Sensitivity** mode for the message that you want to deliver. This mode lets you mark a message as one of the following:

- **Normal**—this is the default mode
- **Private**—for personal or business-related messages that you do not want others to see
- **Personal**—for personal but not business-related messages
- **Confidential**—for business-related messages that you want to be treated according to the confidentiality policy of your company

You can set this mode for both voice and fax messages.

To specify the maximum number of concurrent calls

1. Select **Tools > Options**.
2. Select the **Message Delivery** tab.
3. Select the maximum number of concurrent calls.
4. Select **OK**.

To specify the number of seconds to wait after connecting before prompting a caller

1. Select **Tools > Options**.
2. Select the **Message Delivery** tab.
3. In the pause box, select the number of seconds to wait after connecting before prompting the caller for a password.
4. Select **OK**.

To specify the number of delivery attempts using the pager notification feature

1. Select **Tools > Options**.
2. Select the **Message Delivery** tab.
3. In the **Try** box, select the number of attempts for delivery to the pager system.
4. In the **Delay** list, select the number of minutes to wait between notification delivery attempts.
5. Select **OK**.

To specify the email service that is used to send email

1. Select **Tools > Options**.
2. Select the **Message Delivery** tab.

3. From the Send Email using list, select the email service that you want to use for the Send email feature. The Send email using list consists of servers that have already been configured on the system.
4. Select **OK**.

To specify the message sensitivity mode

1. Select **Tools > Options**.
2. Select the **Message Delivery** tab.
3. In the **Message Sensitivity** section, do one of the following:
 - From the **Voice Messages** list, choose the sensitivity level that you want for a voice message.
 - From the **Fax Messages** list, choose the sensitivity level that you want for a fax message.
4. Select **OK**.



Messages that have different sensitivity levels are managed by individual message stores. You can check the email server's documentation for the specific capabilities that are available when using this mode.

Configuring message waiting indicator options

You can specify how the message waiting indicator operates after a user receives a message. The instructions below describe how to set the message waiting indicator behavior system wide, however you can also set it individually for specific users. Users can also set their own message waiting indicator behavior. If a user selects Use system defaults, then the message waiting indicator behavior defaults to the message waiting indicator behavior set system wide.

Because message waiting indicator behavior can be set by both the administrator and the user, and the administrator can set it both system wide and individually, sometimes these settings may differ. The message waiting indicator behavior in effect corresponds to the last mode set. Thus, the administrator can set the system wide message waiting indicator behavior when a new system is installed or upgraded. The user can then set their own message waiting indicator behavior, which may differ from the system setting. However, if the administrator then changes the system wide setting, this setting overrides the user's setting, and they will have to reset it if they want a different setting.



Phantom mailboxes do not have message waiting indicators.

Disable the mode if you do not want to manage the message waiting activity. If you enable the mode, the message waiting indicator turns on when:

- A caller leaves a message in the user's mailbox.
- The user receives a message sent using the manage mailbox feature.

- A forwarded voice or fax message arrives in the user's mailbox.



A voice or fax message that is forwarded by email does not turn on the recipient's message waiting indicator.

You can specify the message waiting indicator to turn off when the user listens to the first new message, or the last new message in the mailbox.

If the system is busy and is using voicemail ports to manage the message waiting indicator activity, you can increase the number of simultaneous operations. Doing so can reduce the time it takes to deliver changes to message waiting indicators. Set the number at about 1/8 the total number of ports in the system. There is no advantage to setting the number higher than the number of ports.

When you are configuring message waiting indicators there are some options that should rarely be changed. The simultaneous event monitors value controls the number of threads used to process mailbox events for the purpose of updating message waiting indicators. The default value is **1**, and this mode rarely requires changing.



Only change the value of the simultaneous event monitors under the direction of the ADTRAN Technical Support team.



For message waiting indicators to update when messages are accessed in Microsoft Outlook Web Access or on a mobile device, you can configure Microsoft Exchange Server to monitor for changes in users' mailboxes, and for the UC server to adjust the waiting light accordingly. For more information, refer to [Configuring Exchange Server on page 138](#).



For troubleshooting information, refer to [Troubleshooting Message Waiting Indicators on page 317](#).

To disable the message waiting indicator mode

1. Select **Tools > Options**.
2. Select the **message waiting indicator** tab.
3. Enable the **Never turn on or off** button.
4. Select **OK**.

To specify message waiting indicator options

1. Select **Tools > Options**.
2. Select the **message waiting indicator** tab.

3. Enable the **Turn on when new message arrives** button.
4. Enable either of the following:
 - **Turn off when first message is accessed in mailbox**—the light turns off when the user *accesses* the first new message, regardless of whether the user disconnects before listening to the message
 - **Turn off when no more new messages in mailbox**—the light turns off when the user accesses the last new message
5. Select **OK**.



The changes that you make to the message waiting indicator options come into effect for new messages only.



For information on setting the message waiting indicator options for individual users, refer to [Configuring a user's message waiting indicator on page 99](#).

To set the number of simultaneous message waiting indicator operations

1. Select **Tools > Options**.
2. Select the **message waiting indicator** tab.
3. In the Operational section, type a number in the **Maximum number of simultaneous** box.



You can set the number anywhere between 1 and 16 inclusively. The default is 1.

4. Select **OK**.

To set the number of simultaneous event monitors

1. Select **Tools > Options**.
2. Select the **message waiting indicator** tab.
3. Under Operational parameters, **Maximum number of simultaneous monitors**, select a number.



You can set the number anywhere between 1 and 16 inclusively. The default is 1.

4. Select **OK**.

Setting the message playback order

You can configure system wide the order in which new (unheard) and old (previously heard) messages are played over the telephone user interface (TUI).

- If **Enable sorting** is checked, then the default values are for new messages to be played in chronological order (oldest messages first), while old messages are played in reverse chronological order (newest messages first). **Enable sorting** is the default value.
- If **Enable sorting** is unchecked, then messages will be played in an order that is dependent on the server. That is, the user cannot rely on any specific order for message playback. If a specific order of message playback is required, then **Enable sorting** should be checked.

Users do not have the same options for setting message playback sort order, and instead have the following options available to them:

- Listen to voice and fax messages only.
- Listen to voice and fax messages first then email messages.
- Listen to all messages in the order they were received.



*If **Enable sorting** is checked, and a user has selected **Listen to voice and fax messages only**, or **Listen to voice and fax messages first then email messages**, then all voice, fax, and email messages (if selected), are sorted together. Previously, voice messages were played first, followed by fax messages, and then email messages (if selected).*

To set message playback order for new and old messages

1. In the Administrator application, select **Tools > Options**.
2. Select the **Mailbox Management TUI** tab.
3. Check the **Enable sorting** checkbox. Play new (unheard) messages defaults to playing in chronological order, while old (heard) messages defaults to playing in reverse chronological order.
4. Select the radio buttons to play new (unheard) messages and old (heard) messages in either chronological or reverse chronological order, and select **OK**.

Setting voicemail options

These options define how voicemail elements work when you include them in a service that you activate on a trunk or when a user with Personal Business Assistant mode inserts the voicemail element in a service. The options also apply system-wide to Personal Business Assistant mode.

You can allow callers to transfer out of a user's voicemail when they press a specified key. This is useful if, for example, a caller wants to talk to any salesperson. While the caller listens to a salesperson's voicemail, the caller can press a key to transfer to another salesperson. Alternatively, callers can leave a message for one salesperson, and transfer to another after they finish recording their message, without having to hang up and call back into the system.



For information on voicemail elements, refer to [Voicemail element on page 236](#).

To specify the NetVanta UC Server access key

1. Select **Tools > Options**.
2. Select the **Voicemail** tab.
3. From the **Enter mailbox management** list box, select a key to be used to start mailbox management. This key is the UC server access key.



The UC server access key cannot be the same as the operator access key (refer to [To specify the dial by name mode on page 31](#)).

To enable a beep to signal the start of recording

1. Select **Tools > Options**.
2. Select the **Voicemail** tab.
3. Enable the **Beep before recording** check box.
4. Select **OK**.

To set the maximum length of a recording

1. Select **Tools > Options**.
2. Select the **Voicemail** tab.
3. In the **Record for** text box, enter in seconds the maximum length of a recording. The length can range from **5** to **900** seconds. The default maximum length of a recording is **240** seconds.
4. Select **OK**.

To enable silence detection

1. Select **Tools > Options**.
2. Select the **Voicemail** tab.
3. Enable the **Stop recording after** check box.
4. Specify in seconds the period of silence to elapse before the call is terminated. Recording stops after the number of seconds that you specify. The default period is **8** seconds.
5. Select **OK**.

To enable the termination of a recording when a caller presses a key

1. Select **Tools > Options**.
2. Select the **Voicemail** tab.
3. Enable the **Stop recording when key pressed** check box.
The recording stops when a caller presses a key.
4. Select **OK**.

To allow transfer from voicemail when a caller presses a key

1. Select **Tools > Options**.
2. Select the **Voicemail** tab.
3. Select a key from the **Enter skip-out of voicemail when** list box.
4. Enable the **Allow transfer from greeting** check box if you want callers to be able to transfer out of a user's greeting before leaving a message.
5. Enable the **Allow transfer after record** check box if you want callers to be able to access another user's voicemail after they record a message.
6. Select **OK**.

Setting assisted transfer options

The Assisted Transfer element is used in call answering services. You can specify the maximum number of seconds that the UC server waits for an answer before recalling an assisted transfer attempt. The default is **20** seconds.



For information on the Assisted Transfer element , refer to [Assisted Transfer element on page 239](#).

To change the assisted transfer timer mode

1. In the Administrator application, select **Tools > Options**.
2. From the **General** tab, enter or select the number of seconds to wait before recalling an assisted transfer attempt.

Setting fax message delivery options

Fax specifications relate to outgoing fax transmissions from your workplace and to Fax-on-demand elements that you add to services.

You can specify the number of times that the UC server attempts to transmit a fax. You can also specify the interval between successive attempts.

Fax transmission options allow you to specify what is included with a fax transmission report. Fax transmission reports go to a user's Sent Items folder when a fax is sent successfully. If the fax is unsuccessful, an email stating the failure appears in the user's inbox.



For more information on faxing, refer to [Faxing on page 167](#).

To set the number of transmission attempts for outgoing faxes

1. Select **Tools > Options**.
2. Select the **Fax Messages** tab.
3. In the **Try** list, select the number of transmission attempts for outgoing faxes.
4. Select **OK**.

To set the delay between transmission attempts for outgoing faxes

1. Select **Tools > Options**.
2. Select the **Fax Messages** tab.
3. In the **Delay** list, select the number of minutes between transmission attempts.
4. Select **OK**.

To set the number of simultaneous outgoing faxes

1. Select **Tools > Options**.
2. Select the **Fax Messages** tab.
3. From the **Send up to** list, select the number of simultaneous faxes that you want to send.
4. Select **OK**.

To specify fax transmission options

1. Select **Tools > Options**.
2. Select the **Fax Messages** tab.
3. In the **Fax transmission report options** section, select one of the following from the list:
 - **Entire fax only**—delivers the entire fax to the user's Sent Items folder, without including the fax transmission report.
 - **Entire fax with fax transmission report**—delivers the entire fax including the fax transmission report to the user's Sent Items folder.
 - **Fax transmission report only**—delivers only the fax transmission report to the user's Sent Items folder. If you select this mode, the following warning appears: "Selecting "Fax Transmission Report only" will not include the original fax contents when delivering to individual mailboxes or the system defined mailbox for Message Archiving. Ensure that delivering "Fax Transmission Report only" complies with your internal document retention policy and adherence to your industry legislative guidelines."



Changes to the fax transmission options take effect immediately.

4. Select **OK**.

Setting fax-on-demand options

For a Fax-on-demand element activated on a service, you can specify the minimum and maximum number of digits that a fax number must have to initiate transmission. You can also enable or disable the service for long distance callers. Advanced Fax-on-demand options involve specifying the keys that callers press to indicate that they have finished entering their fax number or to cancel the operation.

You can specify the number of times callers are allowed to attempt to enter their fax number, as well as the interval between attempts. You can also specify the number of simultaneous faxes that the user can send.



For more information on faxing, refer to [Faxing on page 167](#).

To set the minimum and maximum number of digits for local fax-on-demand services

1. Select **Tools > Options**.
2. Select the **Fax-on-demand** tab.
3. In the **Local phone number** boxes, select the minimum and maximum digits for fax-on-demand services.
4. Select **OK**.

To make fax-on-demand services available to long distance callers

1. Select **Tools > Options**.
2. Select the **Fax-on-demand** tab.
3. Check the **Enable long distance phone numbers** check box.
4. Select the minimum and maximum number of digits for Fax-on-demand services for long distance phone numbers.
5. Select **OK**.

To specify keys to signal the end of caller input and to cancel a fax-on-demand request

1. Select **Tools > Options**.
2. Select the **Fax-on-demand** tab.
3. Select **Advanced...** to open the Advanced Fax-on-demand Options dialog box.
4. From the **End of input** list, select the key that callers can press to signal that they have finished entering their fax number.
5. From the **Cancel** list, select the key that callers can press to cancel their fax-on-demand request.
6. Select **OK**.

To set the maximum number of times a caller can attempt to enter a fax number and the timeout period

1. Select **Tools > Options**.
2. Select the **Fax-on-demand** tab.
3. Select **Advanced...** to open the Advanced Fax-on-demand Options dialog box.
4. In the **Maximum** box, type the maximum number of times callers can attempt to enter their fax number. When callers reach the maximum number of attempts, the operation terminates.
5. In the **Timeout** box, type the number of seconds to wait for callers to enter a fax number before the operation terminates.
6. Select **OK**.
7. Select **OK**.

Message archiving options

You can configure message archiving options for each user on the system.

To specify the message archiving mode

1. Select **Tools > Options**.
2. Select the **Message Archiving** tab.
3. Enable any of the following check boxes:
 - **Incoming fax messages**
 - **Outgoing fax messages**
 - **Incoming voice messages**
4. From the list of user mailboxes, select the mailbox of the user that you want to apply message archiving to.
5. Select **OK**.

To specify the user's mailbox for duplicate messages

1. Select **Tools > Options**.
2. Select the **Message Archiving** tab.
3. Select a user's mailbox from the **Select the user's mailbox that all duplicate messages will be recorded to** list.
4. Select **OK**.

Setting Auto attendant Options

You can set the following auto attendant options:

- Setting operator options
- Selecting a male or female voice for system announcements

- Selecting the order in which callers enter a name using the Dial by Name option
- Configuring non-user extensions and identities



For more information about auto attendants, refer to [Auto attendant Identities on page 112](#).

Setting operator options

You can set system-wide options for the operator, typically the company receptionist.

To set operator options

1. In the Administrator application, select **Tools > Options**.
2. Select the **Operator Definition** tab.
3. Check the **Enable operator key** check box if you want to allow callers to enter the operator access key and be transferred to the operator extension.
4. From the **Operator access key** list, select a key for operator access. This is the number that callers will press to be transferred to a company receptionist. Typically the number 0 is used.



The operator access key cannot be the same as the UC server access key (defined in the next section, [Setting voicemail options on page 24](#)).

5. In the **Operator Identity** box, type the operator extension to which callers are transferred when they press the operator access key (as defined in step 4, above).
6. From the **Default mailbox** list, select an Exchange mailbox. This is the mailbox where voice and fax messages are left for the operator. This includes incoming faxes that are sent directly to an attendant service using the **Play Announcement** element, or where a service element does not route them to a user extension. The configured mailbox can then redistribute the fax to the appropriate person.
7. Select **OK**.

Selecting a male or female voice for system announcements

auto attendant options allow you to specify male or female for the system announcements that are used for prerecorded announcements, and voicemail behavior for voicemail elements and for Personal Business Assistant mode.

To specify a male or female voice for system announcements

1. Select **Tools > Options**.
2. Select the **General** tab.
3. In the **System announcement gender** section, select one of the following:
 - **Female**
 - **Male**

4. Select **OK**.

Selecting the order in which callers enter a name using the dial by name mode

Using the dial by name mode, you can specify the order in which callers can enter a name (either the first or last name) using their telephone keypad, and then be connected to a corresponding extension.

To specify the dial by name mode

1. Select **Tools > Options**.
2. Select the **General** tab.
3. Select one of the following:
 - **First name, last name**
 - **Last name, first name**
4. Select **OK**.



For information on the Dial by Name element, refer to [Dial by Name element on page 235](#).

Configuring non-user extensions and identities

Identities typically refer to extensions. The UC server validates a transfer destination identity. You can transfer to identities that do not have a user associated with the extension number, such as conference rooms, lobby phones, etc.



For more information about identities, refer to [Managing User Extensions and Identities on page 109](#).

The identities that you enter allow the **Dial by Extension** and **Select Extension** elements, used in custom call answering services, to transfer callers to these identities.



For more information about services, refer to [Creating Custom Call Answering Services on page 213](#). For more information about the Dial by Extension and Select Extension elements, refer to [Dial by Extension element on page 235](#) and [Select Extension element on page 249](#).

In order for inbound callers to transfer to the UC server conference bridge, the conference bridge number

must be added to the list of valid entries. The default number is 7050.



For more information about the UC server conference bridge, refer to [Using the NetVanta UC Server conference bridge on page 157](#).

To define a list of accessible identities

1. In the Administrator application, select **Tools > Options**.
2. Select the **Identity** tab.
3. Select the Insert identity icon.
4. Type the identity.
5. Select **OK**.

Setting Outbound Calls and Dialing Options

Dialing rules allow you to restrict outgoing calls, specify dialing properties, and specify the keys that users must press to access an outside line for local and long distance calls. Dialing options are used to set restrictions, properties, and rules related to outgoing calls, and are used for services and many UC server features including:

- Active Message Delivery
- Pager Notification
- Transfer Call
- Fax-on-demand element
- Fax sending capabilities

Active Message Delivery, Pager Notification, and Transfer Call are options available to Personal Assistant (PA) users. The Fax-on-demand element is available for use in custom call answering services, which are available to Personal Business Assistant (PBA) users.



*The dialing options are not global dialing rules that apply to regular outgoing calls from phones. These dialing options work in conjunction with the dialing restrictions set on the communication system. The dialing restrictions set on the communication system override those set from the **Tools > Options > Dialing** tab. To set restrictions and rules for incoming and outgoing calls, refer to [Routing Calls on page 43](#), specifically the sections [Routing calls using the dial plan on page 43](#) and [Allowing and Restricting Long Distance and Other Calls on page 46](#).*

For more information on faxing, refer to [Faxing on page 167](#).

You can set restrictions for outgoing calls, for example to prevent users from transferring calls to long distance numbers. You can also set exceptions to the restrictions. For example, you might restrict long distance calls in general, but allow long distance calls to a specific area code where another office or a

customer is located. In addition, you can define when dialing restrictions are applied for PA, PBA, and Attendant services. For example, a restriction set to prevent long distance calls denies calls starting with 1. However, this restriction also prevents calls to extensions starting with 1. If internal extensions were a maximum of four digits, then enabling Enforce only for numbers longer than 4 digits, prevents long distance calls while allowing calls to extensions starting with 1.

Rules relate to the number that users must enter on their telephone keypad to access an outside line for local or long distance telephone calls.

To restrict outgoing calls

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. In the **Exclude** box, type the restrictions that you want to apply to outgoing calls. Use a semi-colon (;) to separate each restriction.
4. In the **But allow** box, type any exceptions that are not restricted.
5. In the **Enforce only for numbers longer than x digits**, enable the check box and select the number of digits to apply the restrictions only to numbers above the specified number of digits.
6. Select **OK**.

To set dialing properties

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. From the **Country/region** box, select the country or region that you want to dial from.
4. Type the area code in the **Area code** box.
5. Select **OK**.

To define the keys that are used to obtain an outside line for local and long distance calls

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. Type a number in the **To access an outside line for local calls, dial** box.
4. Type a number in the **To access an outside line for long distance calls, dial** box.



The number in each box represents the key that users press to access an outside line for local and long distance calls. That key depends on the configuration of the PBX. The default key on most PBXs is 9 for both types of access.

5. Select **OK**.

Advanced dialing rules

Advanced dialing rules allow you to specify the conditions under which you can dial telephone numbers

from your office. You can apply dialing rules to a specific area code or office code of outgoing telephone numbers. The office code consists of the first three digits of a regular seven-digit telephone number. You can also specify local 10-digit dialing.

You can change the prefix that is applied to specified outgoing telephone numbers.

To specify advanced dialing rules for outgoing calls

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. Select the **Advanced** button.
4. Select **New** to specify a set of advanced dialing rules.
5. Type an area code in the **Area Code** box.
6. Choose one of the following:
 - **All office codes**—the advanced dialing rules apply to all office codes
 - **Specific office codes**—the advanced dialing rules apply to the office codes that you specify
7. If you enable the **Specific office codes** button, type the office code in the office code box, and then select **OK**.
8. Select **Add** to enter an additional specific office code.
9. Select **OK**.

To specify the prefix

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. Select the **Advanced** button to open the Advanced Dialing Properties box.
4. In the **Before the number, dial** section, choose one of the following:
 - **Local prefix**
 - **Long distance prefix**
 - **Custom**
5. If you choose Custom, in the **Custom** box type the prefix that you want.
6. If you want to dial the area code, enable the **Include the area code** check box.



Check the **Preview the dialed number** box to preview the telephone number elements to dial. These elements are based on the advanced dialing properties that you specify.

7. Select **OK**.

Setting account codes

You can set a default account code that applies to any service, or you can specify the account code dialing

rules so the user can use any account code that follows the rules that you specify. An account code is a specified set of digits that the caller must enter after completing the dialing pattern. Account codes represent an accounting category, for example, expenditures, revenues, or salaries. They are also used to ensure security because a caller who does not have an account code cannot access specified information.

The account code dialing rules let you specify the following:

- **Account code dialing position**—determines the position where the account code is inserted in the dialing string. You can insert the account code before the PBX external dialing prefix, between the PBX external dialing prefix and the telephone number, or after the telephone number.
- **Account code dialing information**—specifies special digits or pauses that must be inserted before or after the account code. A single comma (,) represents a half-second pause. You can specify special PBX dialing sequences or feature access codes.



You can also specify multiple commas using the following syntax: \xx, where xx is the number of commas. e.g.: 9\05,5551234 -> 9,,,,,5551234

- **Account code dialing format**—specifies the minimum and maximum length of account codes that a user can enter when dialing out. If users enter an account code that is too long or too short, an error message states that they must enter an account that is within the specified range.

You can use account codes to allow long distance dialing, in which the telephone company applies the long distance charges to your account code.

To set the default account code

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. In the Account code section, enable the **Enable account codes** check box.
4. Type the account code in the **Account code** box.
5. Select **OK**.

To set the account code dialing position

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. In the Account Code section, enable the **Enable account codes** check box.
4. Select the **Details** button to open the Account Code Dialing Rule box.
5. In the Account code dialing position section, specify where you want to insert the account code in the dialing string.
6. Select **OK**.

To set the account code dialing format

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. In the Account Code section, enable the **Enable account codes** check box.
4. Select the **Details** button to open the Account Code Dialing Rule box.
5. In the first Account code dialing format section, specify pauses (,) or dual-tone multi-frequency (DTMF) characters that you want to be inserted before or after the account code.



A comma inserts a half-second pause before or after the account code.



You can also specify multiple commas using the following syntax: \xx, where xx is the number of commas. e.g.: 9\05,5551234 -> 9,,,,5551234

6. Select **OK**.

To set the account code dialing format length

1. Select **Tools > Options**.
2. Select the **Dialing** tab.
3. In the Account Code section, enable the **Enable account codes** check box.
4. Select the **Details** button to open the Account Code Dialing Rule box.
5. In the second Account code dialing format section, specify the minimum and maximum length of account codes that a user can define.
6. Select **OK**.

Setting Administrative, Maintenance, and Diagnostic Options

The following administrative, maintenance, and diagnostic options can be set system-wide.

- Automatically archive logs
- Configuring diagnostic system options
- Setting mail server message buffering
- Specifying active domains
- Configuring system call options

Automatically archive logs

You can enable automatic log archiving, in order for the UC server to periodically create log archives based on a set of criteria that you configure, allowing you to manage log information more effectively.

You can set the maximum size of an individual log file. Once the size of a log file exceeds that threshold, it is automatically archived.

The UC server ensures that the log files do not fill the hard drive of the UC server. The log system does not store logs to the UC server hard drive if the hard drive is lower than the limit configured for the disk threshold limit.



For more information on logs and log archives, refer to [Managing logs on page 324](#).

To automatically archive logs

1. In the Administrator application, select **Tools > Options**.
2. On the **General** tab, check **Enable automatic archiving**.
3. Set the log file size limit.
4. Set the disk storage limit.
5. Select OK.

Configuring diagnostics system options

You can configure Diagnostics System Options to notify ADTRAN of any problems with the UC server. This information is then used to make improvements to the UC server. You can also configure the exception reports and diagnostic notifications to be sent to yourself or your reseller if the UC server is not functioning properly, in order that the appropriate action may be taken. Multiple email addresses can be entered to receive the reports and notifications.



ADTRAN does not take action in response to the emails received, but gathers the data for statistical purposes only to aid in improving the UC server. ADTRAN will not take action unless a support incident is raised, following standard support procedures. For information on contacting ADTRAN Technical Support, refer to the ADTRAN website at www.adtran.com.



For information, refer to [Enabling Diagnostic Information on page 327](#).

The diagnostics system options allow you to specify the following:

Exception Reports	Generates and sends a report to ADTRAN whenever the application is not operating properly.
Diagnostic Notifications	Sends diagnostic notifications to specified email addresses.
Port Auditing	Specifies how frequently you want to test idle ports. If a port is faulty, it is reset and returned to idle status.

To configure the NetVanta UC Server to send exception reports

1. Select **Tools > Options**.
2. Select the **Diagnostics System Options** tab.
3. In the **Exception reports** section, check the **Enable** check box.
4. Type an email address in the email address box, where you want to send the exception reports. The default email address is: exceptionreports@adtran.com. You can enter multiple email addresses separated by semi-colons.
5. Select **OK**.



An exception report does not contain confidential company information.

To configure the NetVanta UC Server to send diagnostic notifications

1. Select **Tools > Options**.
2. Select the **Diagnostics System Options** tab.
3. In the Diagnostic notifications section, check the **Enable** check box.
4. Type an email address in the Email address box, where you want to send the diagnostic notifications. You can enter multiple email addresses separated by semi-colons.
5. Select **OK**.

To specify the port auditing interval

1. Select **Tools > Options**.
2. Select the **Diagnostics System Options** tab.
3. In the Port auditing section, check the **Enable** check box.
4. In the Audit interval box, type the number of minutes that you want to elapse between port audits.



The port auditing interval cannot be less than one minute or more than 60 minutes. The default is 5 minutes.

5. Select **OK**.

Setting mail server message buffering

The servers temporarily cache voice and fax messages when the mail server is not available—possibly because of a power outage or network problems. You can specify the maximum number of voice and fax messages that are cached on the mail server, the maximum period that is attempted to deliver the messages to the email server (after that period, the administrator might restart the server to continue delivering the messages), and the time between successive attempts at delivering the voice and fax messages to the user's mailbox. After the mail server communication link has been re-established with the UC server, you can specify the message throughput to the mail server.



For more information on mail servers, refer to [Configuring Servers on page 137](#).

To configure mail server message buffering

1. Select **Tools > Options**.
2. Select the **Message Buffering** tab.
3. Enter a value in each of the following boxes:
 - **Maximum days to attempt delivery**—specifies the number of days the UC server attempts to deliver cached messages to the mail server (the default is 7 days).
 - **Message re-send time interval (min)**—specifies the period that the UC server waits between concurrent attempts to deliver messages to the mail server (the default is 5 minutes).
 - **Message delivery rate**—specifies the number of messages per minute that are delivered to the mail server (the default is 60 messages per minute).
 - **Maximum messages to queue**—specifies the number of messages that are temporarily stored on the UC server in the event that communication with the mail server has stopped (the default is 10,000 messages).
4. Select **OK**.

Specifying active domains

Active domains are domains for which the server supports the creation of Windows users and authentications based on profiles on those domains. The server also supports single sign-on only for profiles in active domains. The server's local domain is always an active domain.

Active domains must have a trust relationship with the server's local domain. A trust relationship is a logical relationship established between domains to allow authentication and authorization to shared resources. The authentication process verifies the identity of the user, and the authorization process

determines what the user is allowed to do on a computer system or network.



Active domains are only relevant to systems integrated with Active Directory. For more information on Active Directory, refer to [Integrating with Active Directory on page 69](#).

To specify active domains

1. Select **Tools > Options**.
2. Select the **Active Domains** tab.
3. From the **Trusted Domains** list, select the trusted domain that you want to specify as an active domain.
4. Select the **Select>>** button. The trusted domain appears in the Active Domains list.
5. Repeat steps 3 and 4 until all the trusted domains that you want appear in the Active Domains list.
6. Select **OK**.

To remove an active domain from the list

1. Select **Tools > Options**.
2. Select the **Active Domains** tab.
3. From the **Active Domains** list, select the active domain that you want to remove.
4. Select the **<<Select** button.
5. Repeat steps 3 and 4 until all the active domains that you want are removed.
6. Select **OK**.

Configuring system call options

You can configure system-wide call options, such as playing background music when a caller is on hold, and enabling overhead paging for sound card paging.



For more information on sound card paging, refer to [Configuring Sound Card Paging on page 63](#).

You can also configure what callers hear if an error occurs during a call. When the **Play warning prompts** mode is enabled, then a voice prompt plays to tell the caller that the system is unable to complete the call. When this mode is disabled, the caller hears a tone pattern when a call is not completed.

You can enable the call **park/pickup** Communication System feature, which allows a user to park a caller on their phone and allows another user to pick up that caller on their phone. This feature is also useful if

you want to change phones during a call.



For more information on using the call park/pickup feature, refer to [Parking calls and picking up parked calls on page 160](#).

In addition, you can enable automatic logging out of agents after missing calls in a call queue, and you can specify the number of calls missed before the agent is logged out.



For more information about queuing calls, refer to [Routing Calls to Queues on page 52](#).

To configure system call options

1. In the Administration navigation pane, double-click **Communication Systems** to open the Communication Systems options.
2. Select **Services**.
3. Enable any of the following check boxes:
 - **Play background music when a caller is on hold**
 - **Play warning prompts when an error occurs during a call**
 - **Allow overhead paging**

This mode is only applicable to sound card paging. It does not apply to SIP paging or gateway paging.

4. Enable park/pickup call functionality if you wish to enable call park/pickup.
 - Type a URL in the Failover URL box.
 - Type a timeout in the Failover Timeout box. The time can range from 15 to 7200 seconds (two hours).



*If a caller places a call on hold and then parks the call, the other party hears background music, regardless of whether **play background music when a caller is on hold** is enabled.*

5. Enable auto logout agent if you wish to automatically log out agents after missing a call in a call queue. You must specify the number of missed calls, from 1 to 99, before an agent is automatically logged out.
6. Specify the amount of time in seconds that a call is presented to available agents, until the overflow timeout expires, and the call is redirected to the overflow service. This time can range from 15 to 7200 seconds (two hours). The default is five minutes.
7. Specify the duration of wrap up time that an agent is given after finishing a call. This number can range from 1 to 7200 seconds (two hours).

3 Routing and Restricting Calls

This chapter details how to route calls, restrict long distance and other calls, and route calls to queues. It covers the following topics:

- [Routing Calls on page 43](#)
- [Allowing and Restricting Long Distance and Other Calls on page 46](#)
- [Routing Calls to Queues on page 52](#)



For information on setting dialing options, refer to [Setting Outbound Calls and Dialing Options on page 32](#).

Routing Calls

Calls are routed using the integrated SIP communications system, which routes to registered identities or telephones first. The communication system routes calls according to dial plan entries by order of priority. If multiple dial plan entries match and the priorities are equal, the communication system determines the route.

Routing and dial plans allow phones on the UC server to use:

- **Outgoing call access**—accesses a gateway to dial calls through the PSTN
- **Feature codes**—to access system resources. The dial plan works in conjunction with gateways, telephones, and system features that are accessed by a feature code.

Routing calls and restricting calls both use standard regular expressions.

Routing calls using the dial plan

The dial plan is used to instruct the call processing agent about how to route calls.

You route calls by assigning dial plans to a gateway, or to a host although it is not recommended. Adding digits to the dial plan allows you to specify a range of numbers, attendant line, or other set of digits.

You can add rules to the dial plan table that change system behavior. You can add special dialing requirements, such as prefix or suffix digits, or system-wide speed-dial numbers. The rules are evaluated based on their priority.

The dial plan entries include UC server feature access codes to enable various UC server features.



You can edit existing dial plan entries, for example, changing a feature access code number if there is a conflict, or replacing the current feature access code with a number that users used in a previous system. However, changing a dial plan entry incorrectly can cause the feature to no longer function. Ensure that you follow the instructions given, and only change those fields that are necessary, in order to maintain feature functionality.

The Dial Plan table uses the following information, from the **Dial Plan Entry** dialog.

To add a new dial plan

1. In the Administration navigation pane, under **Communication Systems**, select the communication system to which you want to add a new dial plan.
2. Select **Routing**.
3. From the Dial Plan table, select the **plus (+)** button to open the **Dial Plan Entry** dialog.
4. In the **Dial Plan Entry** dialog, enter any of the following information that you want to include in the dial plan entry:
 - **Original Digits**
 - **Description**
 - **Priority**
 - **Gateway (or Host)**
 - **Call next member after X seconds**
 - **Digits to Skip**
 - **Prefix to Add**
 - **Suffix to Add**
 - **Transport**
 - **Source Pattern**
5. Select **OK**.

To edit an existing dial plan

1. In the Administration navigation pane, under **Communication Systems**, select the communication system for which you want to edit an existing dial plan.
2. Select **Routing**.
3. Double-click the dial plan entry that you want to edit, to open the **Dial Plan Entry** dialog.
4. In the **Dial Plan Entry** dialog, edit the dial plan information that you want to change.
5. Select **OK**.

To delete a dial plan entry

1. In the Administration navigation pane, under **Communication Systems**, select the communication system from which you want to delete a dial plan entry.
2. Select **Routing**.

3. Select the dial plan entry that you want to delete.
4. Select the trash can icon.

Using regular expressions to route and restrict calls

Dial plan and toll restriction digits are managed using regular expressions. Regular expressions are a flexible way of delivering patterns that are a match. For example, if you specify the regular expression `[0-9]{7,}`, the UC server recognizes any digits from zero to nine, repeated a maximum of seven times, in other words, a standard local telephone number.

Routing incoming DID numbers to extensions

Incoming direct inward dialing (DID) services offer a range of telephone numbers provided by a telephone company, that are typically mapped to attendants, user extensions, fax machines, or PBX services. Through the dial plan, you can map any incoming or outgoing numbers to either a block of telephone numbers or to an individual number.

Routing a block of incoming DID numbers to a block of extension numbers

Using regular expression codes, you can use one rule to map all numbers, rather than mapping numbers individually.

Example:

- A company has purchased a block of 100 DID numbers.
- Using 10-digit dialing, the numbers include 613-555-11xx, where xx ranges from 00 to 99.
- The company extension numbers are 3-digit numbers beginning with the number 2, or 2xx.

You can map all incoming numbers, 613-555-11xx to extension numbers 2xx, by entering the following information in the Dial Plan Entry dialog.

Routing a specific incoming DID number to a specific extension number

You can also route a particular DID number to an individual extension number.

Example:

To map the incoming DID number 613-555-1100 to extension 271, you would enter the following information in the dial plan.

Assign the priority number to a higher priority than the general rule.

Other applications include:

- 1-800 numbers
- Conference calls (for the Host, enter 7050)
- ENUM ITSP prefix
- Call queuing

Allowing and Restricting Long Distance and Other Calls

Use toll restrictions to restrict calls, and to allow only certain types of calls. For example, you can restrict some users from making long distance calls, or restrict all users from calling 1-900 numbers.

There are two types of toll restrictions that you can manage. The toll restrictions that you manage in the Routing configuration category apply to telephones and unified communications servers. The UC server also includes toll restrictions that apply only to Personal Business Assistant and Personal Assistant.

Toll restrictions allow you to define rules that control access to particular ranges of dialed numbers, such as long distance or 900 number services. Toll restrictions are evaluated and executed from top to bottom, as they appear in the table. If the Toll Restrictions table is empty, the system allows all calls.

The Toll Restrictions table uses the following information:

- **Calling Number**—indicates individual extensions, or ranges of extensions, to which you want to control access (uses regular expressions).
- **Called Number**—indicates individual extensions, or ranges of extensions or devices, to which you want to restrict or allow access (uses regular expressions).
- **Allow**—specifies if a call is blocked or allowed. A **0** blocks calls, a **1** allows calls.
- **Priority**—allows you to assign priority to the rule that you want. The default value for priority is 1. The lower the number, the higher the priority.
- **Description**—allows you to describe the toll restriction.

Calling Number	Called Number	Allow	Priority	Description
private	audiocodes	1	10	To Pstn
private	mediatrix	1	10	
private	multivoip	1	10	To multivoip
private	vega	1	10	To Pstn
pstngateway1	pstngateway1	0	20	deny all calls from PSTN to PSTN
public	pstngateway1	0	20	deny calls from public internet to PSTN gateway

Use the Toll Restrictions table to define rules for individual phones or groups of phones. The table also uses regular expressions to specify calling and called numbers. Configuring the priority determines the position of a rule within the table. By changing the priority, you can specify exceptions to the general rules.



For information on setting dialing options, refer to [Setting Outbound Calls and Dialing Options on page 32](#).

Allowing local and toll free calls

You can give users access to local and toll free numbers.

First line allows any registered SIP phone to make local or toll free calls on the PSTN gateway. Dialed numbers similar to the following are allowed by this rule:

- 9-123-4567
- 123-2345
- 1-800-234-4567
- 1-866-543-5678

- 1-877-657-7689
- 1-888-768-3546

Second line prevents any calls not matched by the previous rule from accessing the PSTN gateway trunks, which blocks other users from making toll calls over the PSTN trunks

Restricting long distance calls

You can control user access to long distance numbers. To do this, enter a rule that gives unrestricted access to one extension, and then all the other extensions have only local and toll free access.

First line enables extension 1112 unrestricted access to the PSTN gateway.

Second line allows any unregistered SIP phone to make local or toll free calls on the PSTN gateway. Dialed numbers similar to following are allowed by this rule:

- 9-123-4567
- 123-2345
- 1-800-234-4567
- 1-866-546-5678
- 1-877-657-7689
- 1-888-768-3546

Third line prevents any call not matching the previous rule from accessing the PSTN gateway trunks.

To add a new toll restriction

1. In the Administration navigation pane, double-click **Communication Systems** to open the communication system options.
2. Double-click the communication system to which you want to add a new toll restriction.
3. Select **Routing**.
4. Right-click anywhere in the **Toll Restrictions** content pane.
5. Select **Add...**
6. Enable the **Allow these types of calls** check box.
7. Type the information that you want in any of the following boxes:
 - **Calling number**
 - **Called number**
 - **Priority**
 - **Description**
8. Select **OK**.

To edit an existing toll restriction

1. In the Administration navigation pane, double-click **Communication Systems** to open the communication system options.
2. Double-click the communication system for which you want to edit an existing toll restriction.
3. Select **Routing**.
4. Right-click the toll restriction that you want to edit.
5. Select **Edit...**
6. Change the toll restriction information that you want.
7. Select **OK**.

To delete a toll restriction

1. In the Administration navigation pane, double-click **Communication Systems** to open the communication system options.
2. Double-click the communication system for which you want to delete a toll restriction.
3. Select **Routing**.
4. Right-click the toll restriction that you want delete.
5. Select **Delete**.

Allowing and restricting access to paging

Access to paging is disabled by default for all users. However, you can modify toll restrictions using regular expressions to enable selected users to send pages. For more information on toll restrictions, refer to [Allowing and Restricting Long Distance and Other Calls on page 46](#). For more information on using regular expressions, refer to [Using regular expressions to route and restrict calls on page 45](#).

The first example in the table below shows how to allow paging for all users. The second example shows how to restrict selected users from accessing paging. The third example shows how to allow paging for selected users.

Calling Number	Called Number	Allow	Priority	Description
*	*@pagingservices	1	49	Allows all users to access paging services
1000 1001 1002	*@pagingservices	0	50	Prevents selected users from accessing the paging server
1000 1001 1002	*@pagingservices	1	50	Allows selected users access to the paging server to enable sending pages



The calling numbers outlined above are default examples only, and can be replaced with valid user identities to enable specific users the ability to send pages.



Do not alter the default entry (where Calling Number = `.*@ucservices`) that allows UC services to access the paging service, as this may cause the paging feature to no longer function.



For information on configuring paging, refer to [Configuring Paging on page 61](#). For instructions on sending pages, refer to [Paging on page 159](#).

Restricting picking up connected calls

Using connected call pickup, you can seamlessly transfer calls from one device to another, such as from a cell phone to your desk phone. This feature is most commonly used with mobile twinning, where calls to one phone number can ring simultaneously at multiple devices, such as an internal extension number and a cell phone number. For more information on connected call pickup, refer to [Changing phones during a call on page 164](#). For more information on mobile twinning, refer to [Configuring mobile twinning using a ring group identity on page 113](#).

Connected call pickup is enabled by default, so if your company policy restricts this behavior, then follow the instructions below to disable connected call pickup, using a dial plan entry. This disables connected call pickup for all users, unless you enable the feature for specific users. For more information on dial plans, refer to [Routing calls using the dial plan on page 43](#).

You can disable connected call pickup for most users, while enabling it for some users. To do so, follow the instructions below to disable connected call pickup. Then, follow the instructions to enable connected call pickup for specific users, using individual dial plan entries. Specified users can then pickup calls from specific phone numbers, such as their cell phones, using particular phones, such as their internal extension number.

To disable connected call pickup

1. In the Administration navigation pane, under **Communication Systems**, select the communication system, and select **Routing**.
2. From the Dial Plan table, select the plus (+) button to open the Dial Plan Entry dialog.
3. Enter the information in the dial plan entry, as described below.

Routing Rule

Original Digits	[*]27[0-9]{3,} This regular expression refers to any calls starting with *27 followed by any three or more numbers.
Description	Disable connected call pickup for all users.
Priority	20

Destination

Gateway

Host

prompt-403@pbxservices

This results in an error message if a user dials *27 plus any three or more numbers.

Call next member after X seconds 0

Digit Manipulation

Digits to skip 0

Prefix to add

Suffix to add

Options

Transport udp

Source pattern *

Any user dialing *27 plus three or more numbers now hears an error message, and is unable to pickup a connected call.

To enable connected call pickup for specific users

1. In the Administration navigation pane, under Communication Systems, select the communication system, and select **Routing**.
2. From the Dial Plan table, select the plus (+) button to open the Dial Plan Entry dialog.
3. Enter the information in the dial plan entry, following the example described below.

Routing Rule

Original Digits

[*]27

Because you are specifying the called number below (Suffix to add), in this case Jane's cell phone number, Jane is not required to enter the called number to answer a call from her cell phone.

Description

Enable Jane to answer her cell phone from her extension

Priority

25**Destination**

Gateway

Host

pbxservices

Call next member after X seconds 0

Digit Manipulation

Digits to skip	3 The dialed digits *27 are skipped.
Prefix to add	connectpickup- Accesses the connected call pickup feature.
Suffix to add	6135551111 Adds Jane's cell phone number automatically as the called number.

Options

Transport	udp
Source Pattern	2602@.+ Jane's extension number, followed by "@.+" refers to any calls originating from Jane's extension, 2602.

The dial plan entry above allows Jane to pickup calls to her cell phone number, 613-555-1111, from her extension, 2602. Jane can pickup calls in this manner, even though connected call pickup has been disabled for all other users using the previous dial plan example.

Routing Calls to Queues

Call queuing allows you to route incoming calls to a specific queue. The queued calls are then answered by a person who belongs to a certain group. For example, a call to the sales department can be routed to a sales queue and then answered by any sales agent.

The feature is useful for companies that handle a lot of incoming phone calls, and where the caller is not trying to reach a specific individual, but wants to talk to a member of an agent group, such as a salesperson, as quickly as possible. Call queuing allows callers to remain in a queue while they wait for their call to be answered by the next available agent in the group.



The total number of simultaneous calls that can be queued in the system is 64. Contact ADTRAN if you require your system to queue more than 64 calls.

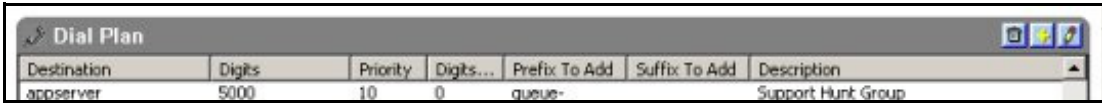
The agent is typically associated with a particular area of expertise, for example, sales or support. A support agent who is ready to answer calls logs into the agent group and receives support calls in the order that callers enter the queue.

To create a queue, create a dial plan entry that directs calls to an attendant identity number (for example, 5000). Add the prefix **queue-** to the identity number in the routing plan. After you create the queue, an agent can log into the agent group for the queue with that identity number, to answer queued calls. For information about using the call queuing feature, refer to [Working with call queues on page 161](#).

You must also configure an attendant identity and a service associated with that identity to manage overflow calls. How a service manages overflow calls is called the overflow behavior. If an agent does not answer a queued call within a specified time period (the default is five minutes), the call is routed to the service that is associated with the attendant identity that has the number that is used to access the queue in the dial plan. The service determines the overflow path that the calls follow, for example, to the support voicemail box.

For information about creating and managing services and activating them on attendants, refer to [Managing Services on page 214](#). For information about building services, refer to [Working with Elements to Build a Service on page 219](#).

Routing calls to queues using dial plan entries



Destination	Digits	Priority	Digits...	Prefix To Add	Suffix To Add	Description
appserver	5000	10	0	queue-		Support Hunt Group

Use a simple dial plan entry to direct calls into a queue, by adding a prefix of **queue-** to the called number in the dial plan, such as in the example above.

In this example, when the call is placed to number 5000, the dial plan entry routes the call to **queue-5000**. The dial plan entry does this by adding the prefix **queue-** to the original called number (**5000**). If the call is not answered, the call is forwarded to the overflow attendant. For more information about overflow behavior, refer to [Configuring overflow calls to queues on page 56](#).

To create a call queuing dial plan entry

1. In the Administration navigation pane, select **Communication Systems** to open the communication system options.
2. Select the communication system to which you want to add a new dial plan.
3. Select **Routing**.
4. In the Dial Plan table, select the plus (+) button.
5. In the Dial Plan Entry dialog, type any of the following information that you want to include in the dial plan entry:

Routing Rule	
Original Digits:	This corresponds to the digits used to dial the queue, for example, 2170 or 5000 .
Description:	Type a description that identifies the call queuing entry.
Priority:	Enter a priority of 10 . Priority determines the matching priority between different or conflicting dial plan entries. The priority must be set below 30.
Destination	
Host:	Enter pbxservices

Call next member after X seconds:	0
Digit Manipulation	
Digits to Skip:	If you want to strip any digits out of the called number, for example, the called number is 8006132170, but you want to map it to 2170, you can skip the first 6 digits by entering 6 in Digits to Skip.
Prefix to Add:	Enter queue- so the dial plan recognizes that the entry is routing calls to a queue.
Suffix to Add:	Leave blank.
Options	
Transport:	UDP
Source Pattern:	.*

6. Select **OK**.

Planning call queuing

You can use services to determine overflow behavior, and to direct calls to a queue.

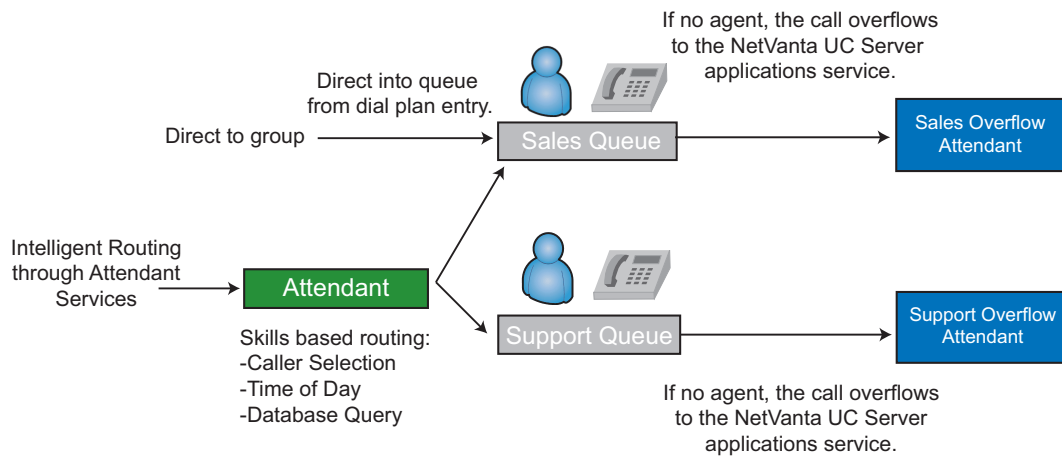
Using a dial plan entry, you can route incoming calls from specified numbers directly to a queue. This is useful when a company does not need to filter types of callers into specific queues. For example, if you have dedicated numbers for each of sales and support, your dial plan entries can simply direct callers to the appropriate queue based on the number they dialed.

Use attendant services to route incoming calls to a queue based on information provided by callers, which then is gathered by a service. This is useful when you want to route callers to a queue based on information the caller provides. For example, the service can ask the caller to provide a customer number and, based on that number, route the caller to the most appropriate queue, particularly when there is a single called number, but several queues are available on the system.

Calls are distributed to the most idle agent. When at least one agent is logged into the agent group, the caller hears an introductory queue greeting. When no agents are logged in, the caller hears that agents are currently not available. The call waits in the queue for a specified period of time (the default is five minutes), for an available agent, and, if an agent does not answer the call, the call takes the specified overflow behavior. Calls to the queue cannot be recorded or monitored.



The maximum number of agents that can log in to a queue at the same time is 64.



Planning considerations when queuing calls

Gather some information about your system and about how you want the call queuing feature to function on your system.

Routing called numbers (DID numbers)

Note the called number (DID number) that is sent by the phone company. If your company uses analog trunking, you can have only one dialed number, which is configured in the gateway.

If your company uses digital trunking, you can negotiate how many dialed numbers you purchase, for example, all the numbers between 613-555-2170 and 613-555-2199. You can also negotiate what numbers the telephone company sends through the digital facilities. You can, for example, request that the phone company deliver 4 digits to you. In the example of 613-555-2170, you can have them send any number of digits up to the length of the telephone number. In the same example, you can have them send the entire telephone number 6135552170.

You must know the called number in order to correctly configure a dial plan entry that routes callers into a queue.

Routing multiple incoming numbers

Note whether multiple incoming numbers must be routed to the same queue. Rather than telling callers which numbers to dial to enter the queue, you can create dial plan entries that automatically route several called numbers into the same queue. For example, the called number 2170 is always directed to the support queue. The company decides to add a 1-800 support number, which also must go to the same queue. You can create dial plan entries that route calls from either number into the same queue.

Routing incoming calls to queues (using a dial plan entry or an attendant service)

Decide whether you need an attendant service to route incoming calls to the appropriate queue, or if a dial plan entry is sufficient.

Routing overflow calls

Whether you use an attendant service and a dial plan entry to route calls to a queue, or only a dial plan entry, you must determine where the call goes when no queue agent is available. If there is no service to direct overflow behavior, the caller receives an error message that the number entered is invalid. Create an attendant identity for overflow calls, and then assign a service to it.

Configuring overflow calls to queues

To configure the overflow behavior, you must create an attendant identity having the same number as the dial plan used to create the queue, and a service for the identity to determine the call flow actions to take. For example, this service can allow the caller to leave a message in a voicemail box, or provide more complex actions such as using caller information or collecting more information from the caller to use database lookups to determine a more complex behavior. For example, such complex behavior could send some callers to a voicemail box, but escalate others to another agent queue, or send email notifications to agent mailing lists. You can associate the attendant service with the administrator profile, or any other user profile.

The service for overflow behavior can be as simple as a single Voice mail element.

To create an attendant identity for overflow calls

1. In the Administration navigation pane, select **Identities**.
2. Right-click anywhere in the Administrator Identities content pane.
3. Select **Create Identity** to display the New Identity Wizard welcome menu.
4. On the Select Identity Type menu, select the communication system from the **Select a communications system** list that corresponds to the communication system on which you created the dial plan entry.
5. In the **Select the class of identity** section, select **Attendant**.
6. In the **Name** box, type a name for the attendant identity.
7. In the **Address** box, type an address for the attendant identity (for example, 5000). This must correspond to the dial plan entry Digits field, which is used to dial the queue (with the prefix **queue-**removed).
8. Enable the **Run Service** button.
9. From the Run Service list, select the service you want to assign as overflow behavior for the queue.
10. Select **Finish**.

Routing calls using attendant services

You can use attendant services to intelligently route calls. An attendant service can route calls based on information from the caller, for example, Caller ID or time of day. Use, for example, a Transfer Call element to transfer calls to a queue from an attendant or Personal Business Assistant service.

Use UC server database integration to route calls. For example, if customers routinely call to find out the status of an order, you can have different agent groups with agents who are trained to respond to questions regarding each status. The service prompts the customer to enter an account number, checks the database

to determine the status of the order, and routes the call to the agent group that has agents capable of answering the customer's specific questions.

A dial plan entry adds the prefix **queue-** to the called number and sends the call to the queue specified in the Digits field of the dial plan. The call is placed in the queue and answered by the first available agent logged into the agent group.

If the call is not answered, an attendant service handles overflow behavior. The service for overflow behavior can be as simple as a Voice mail element.

You can associate the attendant service with the administrator profile, or any other user profile.

To route calls using attendant services

1. Create an attendant service to route calls to a queue based on criteria you specify (see the example below).
2. Create the dial plan entry by following all the steps in the [To create a call queuing dial plan entry on page 53](#).
3. Create an identity for overflow calls by following all the steps in the [Configuring overflow calls to queues on page 56](#).

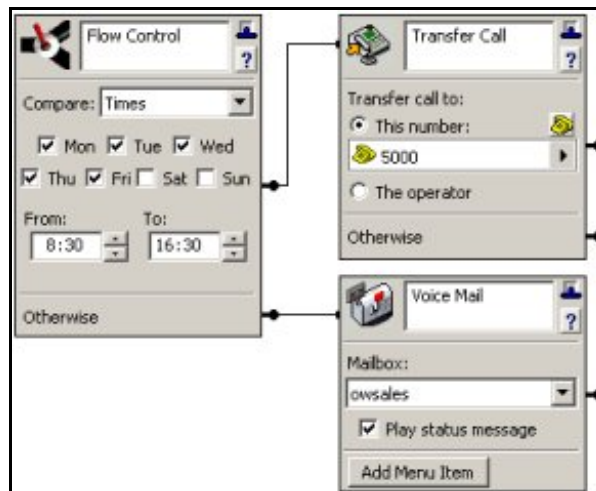
Example—Routing calls to a queue using the Flow Control element

If you want to route calls to an appropriate queue during business hours, and to voicemail after hours, you can insert the Flow Control element in a service to intelligently route the calls. Refer to the following procedure as an example:

To route calls to a queue using the Flow Control element

1. Create a service by following all the steps in [To create a service on page 214](#).
2. Double-click the new service to start the Service Editor.
3. Drag a **Flow Control** element to the canvas.
4. In the **Compare** box, select **Times**.
5. Enable the check boxes for the days in which you want calls to the attendant routed to the queue.
6. In the **From** and **To** boxes, enter the times each day when you want calls to the attendant routed to the queue.
7. Drag a **Transfer Call** element to the Compare Times connector pin.
8. On the Transfer Call element, enter the queue number to which you want calls routed during the times that you specified.
9. Drag a **Voicemail** element to the Otherwise connector pin.

10. In the Mailbox list, select the voicemail box to which you want to send callers outside business hours.



For information about building services to create a call flow, refer to [Creating Custom Call Answering Services on page 213](#).

Example—Routing calls using database integration

Use a service with database elements to route calls to the appropriate agent group based on dynamic database information about the caller.

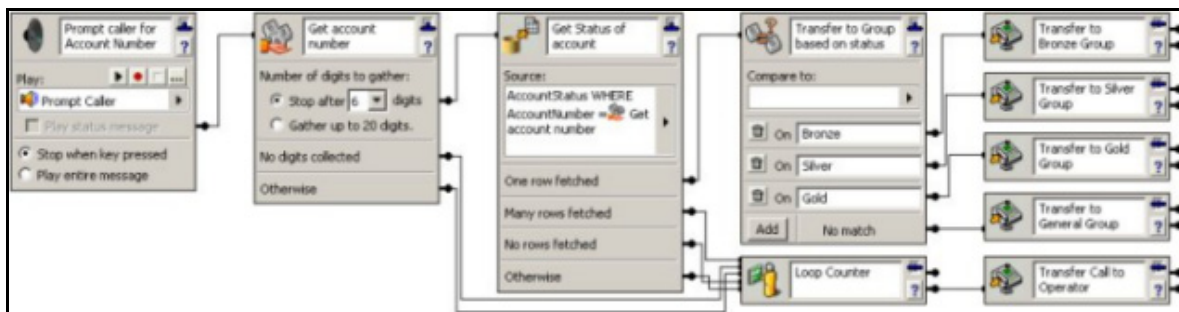
- **Play Announcement** element—informs callers they must have their account number handy to proceed
- **Gather Digits** element—gathers the account number.
- **Fetch Data** element—finds a match in the Account Status database.
- **Compare Digits** element—compares the gathered digits to the digits fetched from the database.
- **Transfer Caller** elements—route the calls to the appropriate queue.

Refer to the following for instructions on how to route calls using database integration.

To route calls using database integration

1. Create a service by following all the steps in [To create a service on page 214](#).
2. Double-click the new service to start the Service Editor.
3. Drag a **Play Announcement** element from the Standard tab to the canvas.
4. Record or select a greeting that informs the caller to be prepared to enter a customer account number.
5. Drag a **Gather Digits** element from the Advanced tab to the connector pin of the Play Announcement element. In the example, the service gathers 6 digits because all account numbers must be 6 digits long.
6. Drag a **Fetch Data** element from the Database tab to the top connector pin of the Gather Digits element.
7. On the Fetch Data element, specify the Data Source, Filter Field, and Equals (if required) information.

8. Drag a **Compare Digits** element to the **One row fetched** connector pin of the Fetch Data element.
9. In the **Compare** section, select the value to which you want to compare the gathered digits (in this example, the customer account number). The service compares the customer account number that the caller entered to the customer account number that it fetched from the database. In this example, the agent groups that correspond to the different customer status types have been named as follows: Bronze, Silver, or Gold. Each group corresponds to a customer status type.
10. Drag a **Transfer Call** element from the Standard tab to the connector pin of each of the status groups to which customer calls can be routed.



For information about using database elements in a service, refer to [Using Database Elements](#) on page 300

Configuring call queues

You can configure the following call queuing options:

- Enable automatic logging out of agents after missing a specified number of calls.
- Set the amount of time that a call remains in the call queue unanswered before being directed to the overflow behavior.
- Set the amount of time that agents are given to wrap up a call after a call has ended, for example, to complete necessary paper work.

You can enable automatic logging out of agents after missing calls in a call queue, and you can specify the number of calls missed, from 1 to 99, before the agent is logged out. When an agent is logged out, either automatically or manually, calls are no longer presented to that agent, until the agent manually logs in again. If auto logout is disabled, then agents are not automatically logged out, regardless of the number of missed calls, so calls continue to be presented to those agents.

As long as an unanswered call remains in the queue, then it is dispatched to the next available, logged in agent until the overflow timeout expires. Therefore, the same call can be presented to an agent multiple times in a round-robin sequence, until it is either answered or the overflow timeout expires, and the call is directed to the overflow behavior. You can set the duration of time that a call remains in the queue unanswered before being redirected to the queue's overflow service. The overflow timeout can range from 15 to 7200 seconds, or two hours. The default value is five minutes. If no agents are logged in at a given time, then calls are automatically redirected to the overflow service.

You can also set a wrap up time period, which is the duration of time that an agent is given to finish up their work after finishing a call, for example to complete relevant documentation or input data. This time period can range from 1 to 7200 seconds, or two hours.

To configure call queues

1. In the Administration navigation pane, double-click **Communication Systems** to open the Communications Systems options.
2. Select Services.
3. Under Global Advanced Call Queue (ACQ), check Enable auto logout agent after X missed calls if you wish to automatically log out agents after missing a call in a call queue. Specify the number of missed calls, from 1 to 99, before an agent is automatically logged out.
4. Specify the amount of time in seconds that the call is presented to available agents, until the Overflow timeout expires, and the call is redirected to the overflow service. This time can range from 15 to 7200 seconds (two hours). The default is five minutes.
5. Specify the duration of wrap up time that an agent is given after finishing a call. This number can range from 1 to 7200 seconds (two hours).

4 Configuring Paging

The UC server provides paging to overhead paging devices and desktop telephones. You can send live pages over the phone, that specified users can receive over their phones or through overhead speakers. You can create and select paging groups to determine who receives a page. Live and prerecorded pages can also be sent from within a call answering service.

Various paging devices are supported, and which paging device is chosen depends on a number of factors including your particular system configuration, the paging feature required, existing paging equipment, and the overall cost.



For information on using paging, refer to [Paging on page 159](#).

For more information, refer to the following sections:

- [Configuring a Paging Device on page 61](#)
- [Creating a Paging Group on page 65](#)
- [Enabling Users to Access Paging on page 68](#)

Configuring a Paging Device

The UC server supports multiple paging devices including:

- SIP telephones
- VoIP speakers
- VoIP zone paging controllers
- Sound card
- Gateway paging to an existing PBX
- Gateway paging to a paging controller



*For information on paging devices, refer to the *NetVanta UC Server Planning and Deployment Guide*, available on the ADTRAN website at <http://kb.adtran.com>.*

Configuring SIP Telephones and VoIP Paging Devices

SIP and VoIP paging includes the following paging features:

- Pages can be received both over the phone and through overhead speakers.
- Does not require additional equipment for receiving pages over the phone.

- Supports multicast paging using telephone handsets or speakers for overhead speakers, that support multicasting.
- Allows callers to select from multiple paging groups or zones, by supporting dual tone multi-frequency (DTMF, also known as touchtone dialing) tones.
- Supports sending live pages.
- Supports sending live and recorded pages from call answering services.

Paging device requirements

Ensure that the paging devices you are using support SIP paging.

- Ensure that the phones you are using support SIP paging.
- If you are using multicast paging (optional), then ensure that the phones and speakers you are using support multicasting.



*For a list of devices that support SIP paging and multicast paging, refer to the technical note NetVanta UC Server **Interoperable SIP Device Features and Comparisons**, available from the ADTRAN website at <http://kb.adtran.com>, or the installation media.*

Configuring multicast paging

Multicast paging can be used for both telephone handset and overhead paging, and it is typically used for large paging deployments. You can send live pages over the phone to a selected group of users, who receive the page either over their phone, through overhead speakers, or both. You can also send live or recorded pages from call answering services. Multicast support also enables sending pages to multiple paging groups or zones.

Multicasting is a method of distributing single streams of audio or video broadcasts to thousands of endpoints simultaneously. Multicasting is used in paging applications to conserve bandwidth while distributing audio pages to several telephony endpoints. For a large paging deployment with several endpoints, multicasting is the recommended method, as it reduces performance issues for the PBX.



*Multicast paging is currently only supported using speakers for overhead paging and telephones for handset paging that support multicasting, and requires some manual configuration. For information on devices that support multicasting, refer to the technical note NetVanta UC Server **Interoperable SIP Device Features and Comparisons**, available from the ADTRAN website at <http://kb.adtran.com>.*

Paging across multiple subnets—network requirements

You must configure the routers correctly in order to forward multicast requests or packets across subnets. Refer to the manufacturer's documentation for your particular router for instructions on how to do so.

Creating paging groups using multicast IP addresses

Before sending a page you must create a paging group to indicate who receives the page, as outlined in [To create a paging group on page 65](#). You can use multicast IP addresses to be members of a paging group. Multicast uses a reserved range of addresses for network broadcasts.

You must also set the port for the multicast IP address to match the paging equipment. Multicast paging devices can configure the multicast IP addresses and ports.

Multicast IP address range 224.1.0.0 to 239.255.255.255

Paging all users of snom phones

If you are using multicast paging with handset paging on snom phones, then use the following multicast IP address and port number for an *everyone* page to users using snom phones. This multicast address and port number have been added to the configuration files for snom phones.

IP address 224.6.9.2

Port 2000

Configuring Sound Card Paging

Sound card or integrated paging uses analog speakers, and a sound card installed in the server, to connect with an external paging system. Using this function, you can send a live page over overhead speakers.



While this method of paging is supported, the recommended paging method is using SIP telephones and VoIP paging devices. For instructions on paging, refer to [Paging on page 159](#).

Sound card paging includes the following features:

- Supports sending pages to overhead speakers
- Supports sending live and recorded pages

Paging requirements for sound card paging

Ensure that your system has met the necessary requirements for sound card paging outlined below.

- A sound card must be installed on the machine where the UC server is installed.
- Analog speakers are required to be used as the output device.
- You must have a paging amplifier that accepts 4 or 8 ohm terminations.
- You must have specialized overhead speakers and amplifiers.
- The speakers or amplifier must be connected to the output jack of the sound card, typically the green colored jack.
- To allow overhead paging it must be enabled for the communication system. To enable overhead paging,

refer to [Configuring system call options on page 40](#).



To properly detect the audio device that you will use for sound card paging, you must remove or disable any extra audio devices. This must be done before deployment of the UC server.



Sound card paging does not support DTMF tones. Therefore, if you require additional paging zones or groups, then you need to use SIP telephone and VoIP paging devices, or an FXO or FXS gateway paging system, so that callers can select the paging zone or group to send the page to. For more information, refer to [Configuring SIP Telephones and VoIP Paging Devices on page 61](#) or [Configuring Gateway Paging on page 64](#).

Ensuring audio quality

To prevent sound effects from the host server being played on the speakers, you must turn off device mapping for Windows sounds.

To turn off device mapping

1. Select **Start > Settings > Control Panel**.
The path may vary slightly depending on the operating system installed.
2. Double-click on **Sounds and Audio Devices**, and select the **Hardware** tab.
3. Select your sound card from the list available, and double-click on it.
4. Select the **Properties** tab, expand **Audio Devices** and select your audio device.
5. Select the **Properties** button.
6. Check **Do not map through this device**.

Configuring Gateway Paging

If your paging system does not support using SIP telephones and VoIP paging devices, or sound card paging, then you can use gateway paging as an alternative method of overhead paging. Gateway paging systems connect to an FXO or FXS gateway. The FXO/FXS gateway provides a combination of paging access and audio path to the paging system.

Gateway paging includes the following features:

- Supports sending live pages
- Allows callers to select from multiple paging groups or zones, by supporting DTMF tones
- Pages are received through overhead speakers

FXO/FXS Gateway Paging Requirements

Before configuring gateway paging, ensure that you have met the requirements outlined below.

- The FXO gateway port must be able to maintain a connection to the FXO paging interface in the absence

of dial tone.

- The FXO gateway port must be able to maintain an audio path in the absence of answer supervision.
- The FXO gateway must support an interface voltage of -48 volts (DC).
- The FXO gateway should support an interface voltage of -24 volts (DC).
- The FXO/FXS interface must be able to pass DTMF tones that are entered by the caller initiating the page request.

To configure gateway paging

1. Connect the paging system to the gateway.
2. Configure the dial plan on the gateway so that calls to the gateway are directed to the FXO or FXS port.
3. For FXS gateways:
 - Create a new identity. On the gateway, configure registration for the new identity. For more information, refer to *Using the New Identity Wizard on page 114*.
4. For FXO gateways:
 - Configure a dial plan on the UC server. For more information, refer to *Routing calls using the dial plan on page 43*.



For information on configuring gateways, refer to the appropriate technical note for your gateway. For the latest technical notes, refer to the ADTRAN website at <http://kb.adtran.com>. Technical notes are also included on the UC server installation media.

Creating a Paging Group

Before sending a page you must create a paging group to indicate who receives the page. You can select users to be members of a paging group by selecting identities, or by selecting specific phones. You can also use multicast IP addresses to be members of a paging group, however this is currently only supported using telephones and speakers that support multicasting, and requires some manual configuration.

Once you have created your paging groups, you can add or remove users from the paging groups as needed.



For information on using paging, refer to [Paging on page 159](#). For information on sending live or prerecorded pages using services, refer to [Page Live element on page 263](#) and [Page Recorded element on page 264](#).

To create a paging group

1. In the Administration navigation pane, double-click on **Paging Groups** and select **New Paging Group**.
2. Select **Next** at the Welcome menu.
3. Enter a name for the paging group (for example, “Everyone”).

4. Enter a description of the paging group (for example, “All employees, contractors, and service phones”).
5. Enter a paging number.
This is the number that will be used to uniquely identify this paging group.
6. The default Maximum Wait time is 2000 ms. You can change this number if necessary.
This is the amount of time that will elapse after the paging device answers the call, before starting the page.
7. If you want an audio prompt sent to paging group members prior to sending a page, then enable the check box. If you enable this check box, then users will hear a beep over their phone or over an overhead speaker, prior to hearing a page.
8. Select **Next**.
9. Enable all user identities that you want to be included in the paging group, and select **Next**.
10. If you do not want to select users by their identities, then you can select specific phones to be included as members of the paging group, and select **Next**. If you have already selected the identity, then you do not need to select the corresponding phone.
11. Enable any multicast IP addresses that you want to be members of the paging group, enter the corresponding port number, and select **Next**.
Multicast paging is supported only using telephones and speakers that support multicasting. Refer to the section [Configuring multicast paging on page 62](#) for more information.
12. Review the paging group summary. If there are any changes you wish to make then select the **Back** button and make the changes. When you are satisfied with the paging group summary then select **Submit**.

Adding a Paging Device to a Paging Group

SIP telephones and VoIP devices are automatically configured to enable live paging using the dial plan. That is, users that have paging enabled, can follow the instructions to send a live page using the feature access code *72. However, in order for users to follow the same instructions using sound card or gateway paging devices, you will need to add the appropriate feature codes as dialable numbers to the paging group. Paging devices cannot be added when creating new paging groups, however once a paging group has been created, you can edit the paging group and add paging devices as dialable numbers.

Adding a feature access code for gateway paging

If you are adding gateway paging to an existing PBX or to a paging controller, then you must also add an appropriate feature access code to the dial plan, which is not already in use by the UC server, or by the SIP phone models in use at your organization. Once the feature access code has been added to the dial plan, then add the feature access code as a dialable number, as a member of the paging group.



For information on adding feature access codes to the dial plan, refer to [Routing calls using the dial plan on page 43](#).

Using sound card paging feature access codes

The following feature access codes already appear in the dial plan by default. To enable users to send pages using a sound card, add the following numbers as dialable numbers, as members of a paging group. Users can then follow the default instructions for sending a live page, using feature access code *72, while the page is sent using the sound card device.

Sending a live page	*75
Sending a recorded page	*77

Adding a Member to a Paging Group

You can add new members to an existing paging group including:

- Identities
- Phones
- Multicast IP addresses
- Dialable numbers

To add a member to a paging group

1. In the Administration navigation pane, double-click on **Paging Groups**.
2. Select a paging group, and in the content pane on the right, right-click on the paging group and select **Add members**. The Paging Group Wizard opens.
3. Select **Next** at the Welcome menu.
4. Select the Member Types to Add:
 - Select **Identities** if you want to select specific user identities to add to the paging group.
 - Select **Phones** if you want to add specific phones to the paging group. If you have already added a user's identity, then you do not need to add their corresponding phone as well.
 - Select **Multicast IP addresses** if you want to enter multicast IP addresses to add to the paging group. In order to use multicast paging, users must be able to receive the page over a telephone handset or speaker that supports multicasting. For more information, refer to [Configuring multicast paging on page 62](#).
 - Select **Dialable numbers** if you want to add a paging device. If you are adding a sound card paging device, then add the specified feature access codes for sound card paging devices. If you are adding gateway paging to an existing PBX or to a paging controller, then ensure that you have already added an appropriate feature access code to the dial plan, before adding the feature access code as a dialable number.
 - Select **Next**.
5. Enable any user identities that you want to be included in the paging group, and select **Next**.
6. Enable any phones that you want to be included as members of the paging group, and select **Next**.
7. Select **Add** if you want to add a multicast IP address, enter the multicast IP addresses and the corresponding port numbers, select **OK**, and select **Next**.

8. Select **Add** if you want to add a paging device, enter the dialable number for the paging device, select **OK**, and select **Next**.
9. Review the paging group summary. If there are any changes you wish to make then select the **Back** button and make the changes. When you are satisfied with the paging group summary then select **Submit**.
10. Select **Next**.
11. Select **Finish**.

To remove a user from a paging group

1. In the Administration navigation pane, double-click on **Paging Groups**.
2. Select the paging group where you want to remove a member.
3. In the content pane on the right, right-click on the paging group member(s) that you want to remove and select **Remove members**.
4. Select **Yes** to confirm. The selected user is removed from the paging group.

Enabling Users to Access Paging

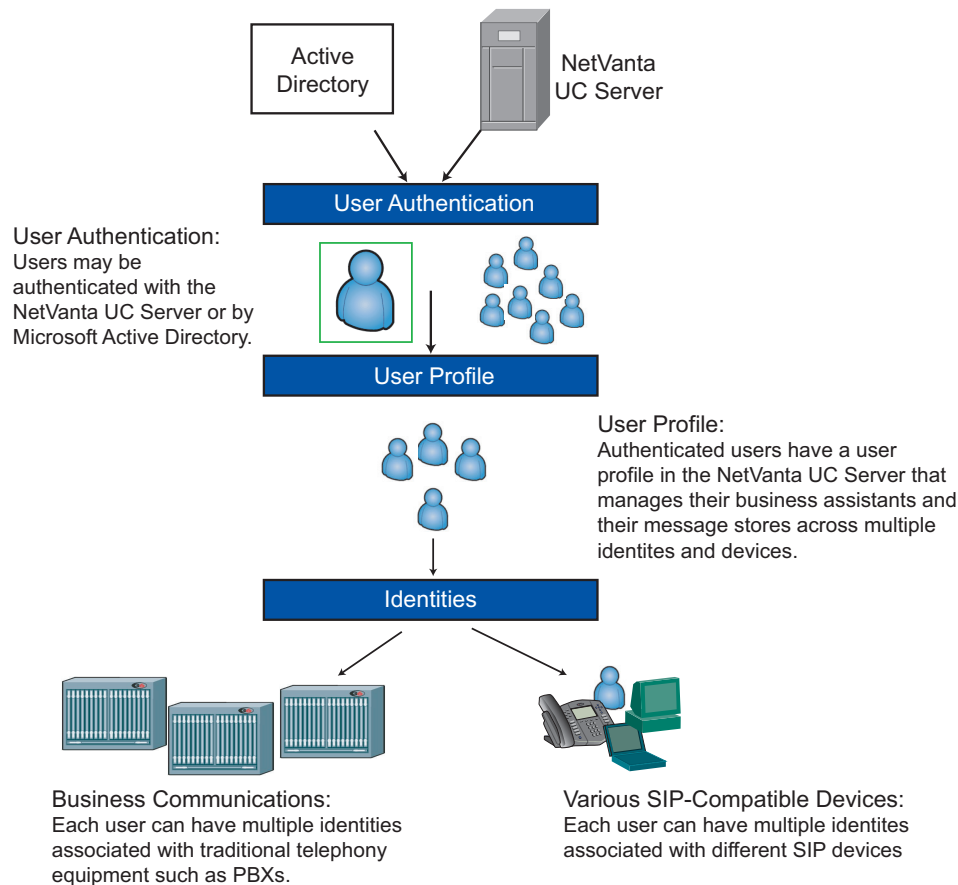
Access to paging can be disabled by default for all users. To enable selected users to send pages, you must modify the toll restrictions. For more information, refer to [Allowing and restricting access to paging on page 49](#).

5 Integrating with Active Directory

Overview

All users on the system are characterized by user profiles, identities, and authentications. Provided a user authentication has access to a system profile, a user might manage user profiles, identities, authentications.

For management purposes, authentications can give a user access to the profiles of other users to configure personal profile settings, such as announcements and notification services.



Users can have any number of identities on the system, such as an extension number. Through the user profile, users can manage the identities that are associated with their profile. The identities can also exist on different communications systems.

UC server is integrated with Active Directory. If your network includes Active Directory for managing networks and shared resources, UC server can take advantage of the Microsoft credentials for granting

access to UC server resources.



For information on selecting an active domain, refer to [Specifying active domains on page 39](#).



For information on changing users to local or Windows users, refer to [Changing Users to Local or Windows Users on page 85](#).

Understanding Active Directory

Microsoft networks include Active Directory, which contains information about all computer accounts, users, and network components, such as printers, file shares, and permissions. Active Directory organizes information about network elements in a tree structure, similar to the way that Windows uses folders and files to organize information on a computer.

When you use the UC server New User Wizard to create a new user, you must choose the user type that you want to create, either a **Windows user** or **Local user**.

Some users and authentications might be defined entirely within UC server. These are called local authentications and users.

Authentications might also be defined based on some Windows user. This Windows authentication is logically linked to exactly one Windows user defined in a domain that has a two-way trust relationship with the domain within which the server is located. Windows authentications are used to enable single sign-on.

Windows users are linked to Active Directory. Certain attributes of a user are linked to Active Directory as well, including the display name, first and last name, Exchange mailbox, and Contacts Database IDs. The last two are used only if the user is configured to use Exchange Server for messaging.

Using Active Directory, the UC server is able to identify and authenticate Windows user logon, using Microsoft credentials, and allow Windows users specific privileges and permissions based on their system-defined roles. Active Directory provides:

- **Single sign-on**—allows users to access network resources without logging on every time.
- **Single management**—provides a single, consistent point of management for users, applications and devices.

Single sign-on strengthens system security and provides administrators with consistent tools to manage security services for local users and remote users. Active Directory improves security by centralizing management and enforcing role-based security, which means that users have access to aspects of the system based on roles that are defined by the UC server administrator.

Single management of resources allows, for example, one place for managing users, including attributes related to the UC server, Windows, and Exchange.

You can, however, create a **local user** that does not have access to Active Directory. Local users must enter a password each time they log on to the system. For example, creating a local user is useful for the UC server and mailbox accounts that do not have Windows accounts, or service accounts.

The New User Wizard prompts you to specify a password when you create a local user. For information about using the New User Wizard to create a user, refer to [Using the New User Wizard on page 87](#).

Using the Active Directory Users and Computers Console



You cannot use the UC server plug-in for Active Directory Users and Computers if the client operating system is Windows 2000.

The UC server is integrated with Active Directory Users and Computers Console, which unifies and simplifies day-to-day system management tasks. It allows administrators to create special tools to delegate specific tasks to users or groups. Administrators can create unique consoles for workers who report to them or for workgroup managers. They can assign a tool with a system policy, deliver the file by email, or post the file to a shared location on the network. Access to Active Directory Users and Computers is automatic with the installation of the UC client. Administrators can change, and delete UC server user profiles from the UC server or from Active Directory.



For information on how to create, change, and delete user profiles from the UC server, refer to [Using the New User Wizard on page 87](#).



For information on changing users to local or Windows users, refer to [Changing Users to Local or Windows Users on page 85](#).

To open the Active Directory Users and Computers Console

1. On the Windows desktop, double-click the shortcut **Active Directory Users and Computers**.



*If there is no desktop shortcut, select **Start > Programs > Administrative Tools > Active Directory Users and Computers**. If this shortcut doesn't exist, select **Start > Settings > Control Panel > Administrative Tools > Active Directory Users and Computers**.*

2. Once Active Directory Users and Computers Console is open, navigate to the Users folder. From here, there are two ways to configure users for the UC server:
 - Server User Tasks Wizard
 - UC server MMC snap-in

Server User Tasks Wizard

The Server User Tasks Wizard allows you to enable or disable users in the UC server, and to change user attributes. Follow the steps in the Server User Tasks Wizard to perform any or all of the following tasks:

- Enable a UC server user, including:
 - Identity and fax address
 - PIN
 - Authorization role
 - Location of the user's mailbox and contact information
 - Phone selection
 - Call answering mode and operator properties
- Modify UC server user-specific attributes
 - Change PIN (for TUI access)
- Disable a UC server user

To start the Server User Tasks Wizard

1. From the user list, right-click the user that you want to modify.
2. Select **All Tasks > UC Server Tasks**.

NetVanta UC Server MMC snap-in

Similar to the Server User Tasks Wizard, the MMC snap-in allows you to enable or disable users in the UC server, and to change user attributes. Unlike the Server User Tasks Wizard, the snap-in provides the ability to configure additional user properties on an attribute-by-attribute basis. For example, the snap-in allows you to configure additional identities for a user, and allow access to the profile from other authentications, and provision additional phones.

To start the NetVanta UC Server MMC snap-in

1. From the user list, right-click the user that you want to modify.
2. Select **Properties**.
3. Select the **UC Server** tab.

Enabling user profiles from Active Directory



*To create a new user it is recommended that you use the **New User Wizard** in the UC server. When using the New User Wizard, you create an identity for that user at the same time, and then assign a temporary PIN to that user. If you enable a user profile in Active Directory, rather than through the New User Wizard, then an identity is not created for the user, and therefore a PIN number cannot be assigned, and this could create security issues. For information on creating UC server user profiles, refer to [Creating User Profiles on page 85](#).*

When you enable a UC server user profile in Active Directory, then the UC server is able to identify and authenticate a Windows user logon using Microsoft credentials, and allow Windows users specific privileges and permissions based on their system-defined roles.

User profile information is available from the **UC Server** tab. For more information on user profiles, select the **Advanced** button to open the User Details dialog.

To enable NetVanta UC Server user profiles from Active Directory

1. From the drop-down list, select the UC server for which you want to configure this user.
2. Check **Enable Unified Communications Server for this user** to link the UC server user to their Windows Active Directory profile. The user profile information is now available.

Changing user profiles from Active Directory

You can change the following types of user profile information, depending on the type of user you have configured for the system:

- General user information
- User messaging information
- User's phone and identities
- User authentications

Changing general user information using Active Directory

You can change general user information from Active Directory, including resetting a user's PIN. Users enter a PIN when they access their mailbox using the telephone. You can reset a user's PIN if the user forgets it, or for added security.

UC server user profile information includes the first, last, and display name of the user, as well as the option to include the user in the dial by name directory.

You can specify the call answering mode, either Personal Assistant (PA) or Personal Business Assistant (PBA).

- The PA user is the default call answering mode, and includes access to all of the standard voicemail features.
- The PBA user has access to more advanced features, and is for users who want to create personalized

services and announcements.

You can also specify the operator for the user, either System or Personal. Callers are directed to an operator by pressing a button in a voicemail greeting, typically 0.

- A System operator is the general operator, and the call is typically transferred to the extension of the company receptionist.
- A Personal operator is an alternative to the System operator, and callers are directed to a different extension, for example, to a personal receptionist, or to an individual in a particular department or location.

Each user profile has a time zone setting that indicates in which time zone the user is located. The time zone setting affects the times and dates that appear on voice, fax, and email messages when users access their mailbox using the telephone user interface. For workers who travel, or are relocating to another site for a period of time, you can change each user's time zone settings.

To reset a user's PIN

1. In the Users content pane, right-click the user name for which you want to reset the PIN, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable UC Server for this user** is checked.
5. Select **PIN**.
6. Type a new number in the **New PIN** box.
7. Retype the new number in the **Confirm new PIN** box.
8. Select **OK**.



The new PIN takes effect immediately.

To change general user information

1. In the Users content pane, right-click the user name that you want to modify, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Advanced** button to open the User Details dialog.
6. On the **General** tab, in the User Information section, change any of the following options:
 - **First name**
 - **Last name**
 - **Display name**

7. Enable the **Include in dial by name directory** check box if you want the user to be accessible by the dial by name mode.
8. Select **OK**.

To change the call answering mode for a user

1. In the Users content pane, right-click the user name that you want to modify, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Advanced** button to open the User Details dialog.
6. On the **General** tab, in the Call Answering section, select one of the following answering modes for the user:
 - **Personal Assistant (PA)**—allows this user access to all the standard voicemail features. This is the default call answering mode.
 - **Personal Business Assistant (PBA)**—allows this user access to more advanced features than the PA user. This call answering mode is for users who want to create personalized services and announcements.
7. Select **OK**.

To change the operator for a user

1. In the Users content pane, right-click the user name that you want to modify, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable UC Server for this user** is checked.
5. Select the **Advanced** button to open the User Details dialog.
6. On the **General** tab, in the Call Answering section, select one of the following operators for the user:
 - **System**—the general operator, typically a company receptionist, where calls are directed when a caller presses 0 in a voicemail greeting.
 - **Personal operator**—a specific operator, for example a personal receptionist or an individual user in a particular department or location, where calls are directed when a caller presses 0 in a voicemail greeting.
7. If you selected **Personal operator**, type the operator's extension number in the box.
8. Select **OK**.

To change the time zone setting for a user

1. In the Users content pane, right-click the user name whose time zone setting you want to change, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.

4. Ensure that **Enable UC Server for this user** is checked.
5. Select the **Advanced** button to open the User Details dialog.
6. On the **General** tab, in the User Information section, select a **Time zone** from the drop-down list.
7. Select **OK**.



You cannot change the time zone setting for the system profile. The system profile time zone setting is always the same as the time zone of the server.

Changing user messaging information using Active Directory

For each user profile, you can specify the email storage type, mailbox, and server, which determines where voice, fax, and email messages are stored.

Select a server to handle contact information. This functionality includes contact matching for messages and call routing based on contact. It also applies to managing mailbox functionality, including contact search for manage mailbox, forwarding messages to contacts, and reply or callback to contacts.



*The Messaging tab is linked to the user profile mailbox selection. The Messaging tab is not visible for Active Directory users where **Use mailbox defined by Active Directory** is enabled. The Messaging tab is also not visible for Microsoft Exchange Server mailboxes. For more information, refer to [Configuring a user's message storage on page 96](#).*

You can enable users to have access to integrated messaging. Enabling integrated messaging adds a Mailboxes navigation bar to a user's UC client. The user is then able to manage their voice and fax messages visually through the Inbox and Sent Items folders in the UC client.

To change messaging information for a user

1. In the Users content pane, right-click the user name that you want to modify, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable UC Server for this user** is checked.
5. Select the **Messaging store** button to open the **Messaging** tab.
6. Select an email storage type from the **Email Storage** list.
7. Check **Enable integrated messaging client**, if you want the user to have access to integrated messaging.
8. Select a server from the drop-down list.
9. Select **OK**.

Configuring user identities using Active Directory

An identity is associated with a specific extension or SIP identity for a hard or soft phone. A user profile

can have zero or more identities, for example, a user might have several telephones. However, identities can only be associated to one user profile.

You can start the New Identities Wizard to create an identity for a user using the Users content pane in Active Directory.

When you create an identity, SIP devices require both a SIP authorization and password to successfully register with UC server SIP registration. The SIP authorization and password are associated with a user identity. Some phones, for example, Polycom phones, support automatic configuration. For more information, refer to [To get the SIP authorization and password on page 124](#).



When configuring an identity on a SIP telephone that does not support automatic configuration, consult ADTRAN technical notes and the phone manufacturer documentation for the specific configuration for your environment.

To add an identity to a user using Active Directory

1. In the Users content pane, right-click the user name to which you want to add an identity, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Identities** button to open the **Identities** tab.
6. Select **Add** to add an identity to the user profile. The New Identity Wizard begins. For information on using the New Identity Wizard, refer to [Using the New Identity Wizard on page 114](#).

To delete an user's identity using Active Directory

1. In the Users content pane, right-click the user name from which you want to delete an identity, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable UC Server for this user** is checked.
5. Select the **Identities** button to open the **Identities** tab.
6. Select the identity that you want to delete.
7. Select the **trash** icon.
8. Select **OK**.

Changing a user's phone and identities using Active Directory

You cannot assign a user directly to a phone, but you can assign an identity to a user, and then assign that identity to a phone. For example, Jane Smith has a phone on her desk with two line buttons. Line one has identity 201, which is Jane's personal phone line. Line two has identity 301, which is a customer support line that is also assigned to Jane Smith, and to other users.

You can assign or remove identities to phones from Active Directory.



When you assign an identity to a phone, the phone immediately reboots.

To assign an identity to a new phone

1. In the Users content pane, right-click the user to whom you want to assign a phone, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Advanced** button.
6. Select the **Phones** tab.
7. Select **Add**.
8. In the Property Page dialog, enter the information for the new phone, and select **Next**.
9. From the Select Identities dialog, select the identity that you want to assign to the new phone.
10. Select **Finish**. The identity that you assign to the phone appears in the Identities section of the Phone tab.



If the phone type has several buttons, the identities are assigned to the buttons in numerical order. You can assign as many identities as there are buttons on the phone.

11. Repeat steps 7 to 10 for all the identities that you want to assign to the new phone.
12. Select **OK**.



You can also assign an identity to a phone using the Identities navigation pane in UC server. For more information, refer to [To assign a user identity to a new phone on page 119](#), or [To assign a user identity to an existing phone on page 120](#).

To assign an identity to an existing phone

1. In the Users content pane, right-click the user to whom you want to assign an identity, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Advanced** button.
6. Select the **Phones** tab.

7. Select **Add**.
8. In the Property Page dialog, select **Assign to an existing phone**. All existing phones with available line buttons are displayed. To show all phones, enable the **Show all phones** check box. All phones are displayed.
9. Select a phone and select **Next**.
10. Select the identity that you want to add, and select **Finish**.

To change information associated with a phone

1. In the Users content pane, right-click the user whose phone you want to change, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Advanced** button.
6. Select the **Phones** tab.
7. Select a phone and select **Open**.
8. Change the phone's information.



You can only change some of the information associated with a phone, for example, IP address or description. If you want to reconfigure the phone, remove it from the network and re-install it using [To add a phone manually on page 125](#).

9. Select **OK**.

To change the button number for identities on a phone

1. In the Users content pane, right-click the user whose button number you want to change, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Advanced** button.
6. Select the **Phones** tab.
7. Select the identity whose button number you want to change.
8. Select the up or down arrow to change the button number for the identity. The button number refers to the line buttons on the phone.
9. Select **OK**.

To delete an identity on a phone

1. In the Users content pane, right-click the user whose identity you want to delete from a phone, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Identities** button.
6. Select the identity that you want to delete.
7. Select the trash icon.
8. Select **OK**.

Changing user authentications using Active Directory

An authentication is a set of credentials that allow or deny users access to the system, for example, using a user name or password. User profiles can have many authentications, and vice versa. Therefore, when an authentication is used to access the UC client, the user might have access to several user profiles, each with several identities. For more information, refer to [Enabling Access to User Profiles using Authentications on page 105](#).

Authentications are created automatically when a user is added using the New User Wizard. The primary authentication is the “main” authentication that owns the user profile.



Although it is possible to remove the primary authentication, ADTRAN does not advise doing so.

When you add an authentication to a user profile, the authentication then has access to another user’s UC server client application. The profiles that a user has authorization to access are displayed in the Authentications field, on the UC Server tab.

To add an authentication to a user

1. In the Users content pane, right-click the user name to which you want to add an authentication, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Authentications** button to open the **Authentications** tab.
6. Select **Add** to add an authentication.
7. Select a user name from the Select Authentication list.
8. Select **Select**.

9. Repeat steps 6 to 8 until you have added all the users that you want.
The users that you select appear in the Authentications list.
10. Select **OK**.

To delete an authentication from a user

1. In the Users content pane, right-click the user name from which you want to remove an authentication, and select **Properties**.
2. Select the **UC Server** tab.
3. Select the server for which you want to configure this user.
4. Ensure that **Enable Unified Communications Server for this user** is checked.
5. Select the **Authentications** button to open the **Authentications** tab.
6. Select the authentication that you want to delete.
7. Select the trash icon.
8. Select **OK**.

Deleting user profiles from Active Directory

When you delete a user profile, all of that user's data is lost. For example, all greetings the user recorded, all announcements, and all services the user created are deleted.



If the local message store user also has messages stored on UC server, the messages are deleted.



The user's email account is not affected when you delete the UC server user profile.

To delete a user profile

1. From the Users folder in Active Directory, right-click a user name.
 2. Select **Delete**.
 3. Select **Yes** to confirm the deletion.
- You have deleted the UC server user profile.

6 Managing User Profiles

This chapter outlines the types of user profiles that you can create and add to the system using the New User Wizard, and how to manage users in the UC server. It includes information about Active Directory, local users, and Windows users so that you can decide the types of users that you want. The chapter covers the following topics:

- [Overview on page 83](#)
- [Creating User Profiles on page 85](#)
- [Changing Users to Local or Windows Users on page 85](#)
- [Using the New User Wizard on page 87](#)
- [Configuring User Profiles on page 93](#)
- [Enabling Access to User Profiles using Authentications on page 105](#)
- [Managing User Profiles from EBS on page 107](#)



For information on managing user profiles from Active Directory, refer to [Using the Active Directory Users and Computers Console on page 71](#).

Overview

Accounts for all users on your system might have been imported from the Microsoft Exchange Server when the UC server was configured at your work site. If accounts were successfully imported using the Server Configuration wizard, you do not have to set up user accounts, or profiles, unless a user is new to the system. If user profiles were not imported, you need to set up a user profile for each user in your office.

Each UC server user needs a profile to access unified communications services. A user profile is associated not only with the general information about a user, such as first and last name, but also with authentications and identities. An authentication determines whether a user can access the system. For example, an authentication might correspond to a login name and password, or a Windows log on. An identity is the address of a user on some device, for example, an extension number. It might be an extension on a communication system or a SIP address that can be used on a SIP hard phone or soft phone.

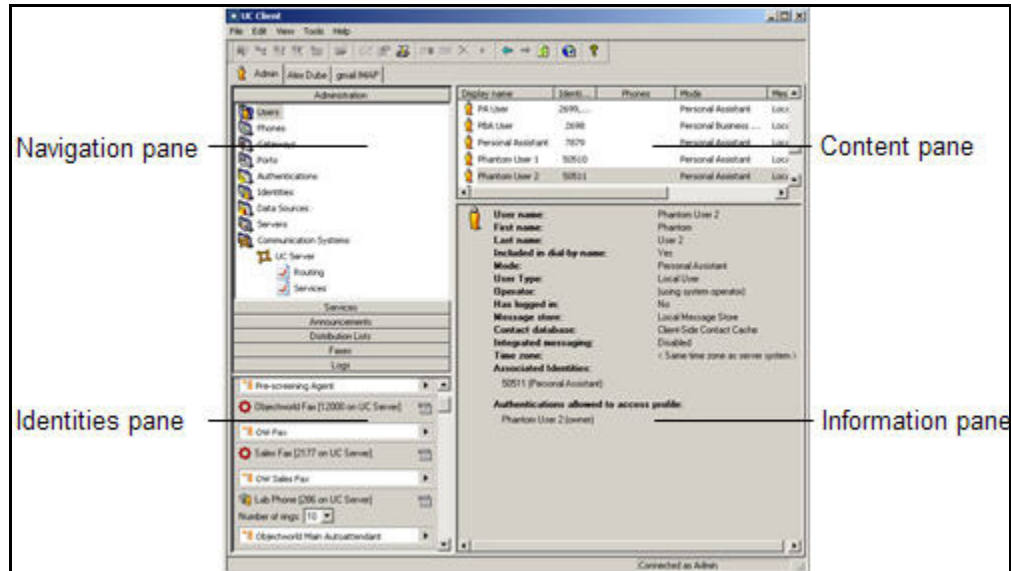
Each user profile consists of one mailbox and at least one identity, or possibly several identities. You can also have phantom mailboxes that are not associated with a physical identity, such as an extension number. UC server routes incoming calls for each user to the user's extension number so that messages can be received and then stored in the user's mailbox for later retrieval. Each identity can have a separate set of behaviors, but messages are saved in the message store that is associated with the user profile.



For more information on identities, refer to [Managing User Extensions and Identities on page 109](#).

To access user profiles

In the Administration navigation pane, select **Users**. User names are displayed in the content pane.



The Users content pane lists the following information for each user on the system:

- Display name** Displays the name of the user or mailbox.
- Identities** Displays the identity of the user or mailbox, for example, an extension number.
- Phones** Displays the phones that are associated with the user.
- Mode** Displays the call answering mode that is configured for the user or mailbox, for example, Personal Assistant or Personal Business Assistant.
- Message Store** Displays the name of the message store type.

The New User Wizard helps you set up each new user profile. You can create either an internal (or local) user or a user based on a Windows user.

Local User

Local users require a password each time they log into the application. The message store for local users can be Exchange.

Windows User

Passwords for Windows users are maintained by Active Directory, so that Windows users do not have to enter a password each time they log on. This functionality is called single sign-on. Windows users can automatically choose to configure an Exchange mailbox from Active Directory.

Creating User Profiles

Select the **Admin** tab to obtain the Administration navigation pane.

After you connect the SIP telephones to the network, the UC server automatically recognizes the phones and allows you to assign user identities to the phones using the New User Wizard. For information about installing and configuring SIP telephones, refer to [Installing and Configuring SIP Telephones on page 123](#).

You can also edit user profiles, identities, or phones after you create user profiles. For information about editing users, identities, and phones, refer to [Configuring the NetVanta UC Server on page 131](#).

Before you create a user profile

1. Connect the new user's telephone to the manufacturer's PBX according to the manufacturer's instructions.
2. If you want to create a unified communications mailbox, then you must first create an email account for the new user, for example, an Exchange mailbox. If you do not want to create a unified communications mailbox, then you can create a local mailbox for users.

After you connect the new user's telephone to the PBX and computer to the server, you can create a UC server user profile.

There are three types of message store for which you can create user profiles: local message store, Exchange, and IMAP. IMAP is the preferred configuration for Lotus Notes.



The New User Wizard is dynamic and prompts you for different information depending on the choices that you make when configuring the new user. For example, Active Directory authentication, extension numbers, message store (Exchange, Integrated Messaging, Voicemail, or IMAP4). The procedures below are only a few of the possible wizard menus that you might view when creating a user profile.

The type of message store that you choose depends on where you want to store the new user's messages. Choosing Exchange means that the user's messages and personal contacts are all stored on the Exchange server. Choosing local message store means that the user's messages and Outlook contacts are stored locally, on the UC server. Choosing IMAP message store means that messages are stored on the IMAP server.

Changing Users to Local or Windows Users

Using the NetVanta User Migration Wizard you can quickly change the location and status of groups of users. You can migrate local users to Microsoft Windows users, linked to Microsoft Active Directory, including converting the users to using Microsoft Exchange Server as a message store.

If a system has been configured off-site, prior to customer installation, users might have been created or imported as local users. Once at the site, you can migrate all or some of those users to Active Directory by running the User Migration Wizard. Link users to Active Directory to enable single sign-on. You can also choose to automatically link users to an Exchange Server.

You can migrate Windows users to local users, which are controlled exclusively by the UC server. Local users have local authentications and you can also choose to use Local Message Store.

To change local users to Windows users

1. Right-click anywhere in the Users content pane, and select **Migrate Users**.
2. Select **Next** to begin the wizard.
3. Enable **Local users to Windows users and authentications**.
4. From the list, select the domain where Active Directory users will be matched to local users.
5. Select one or both of the following:
 - **Use Active Directory for messaging configuration**—switches users to the Exchange Server message store
 - **Delete existing Local authentications that own each user**—causes the authentication to be deleted if the local user is owned by a local authentication
6. Select the local users that you want to migrate to Windows users.



*You can select each user by enabling the check box next to the user's name. Select **Select All** to select all users, or select **Deselect All** to clear all check boxes.*

7. Review the information on the summary menu and select **Submit** if the information is correct.



You must have already performed network integration on the system, using the Server Configuration Wizard, prior to attempting to migrate users.



Voice and fax messages do not migrate from one system to another.

To change Windows users to local users

1. Right-click anywhere in the Users content pane, and select **Migrate Users**.
2. Select **Next** to begin the wizard.
3. Enable **Windows users to Local users and authentications**.
4. Enable any or all of the following check boxes, and select **Next**:
 - **Migrate to Local Message Store**—migrated users are switched (if necessary) to using Local Message Store
 - **Create default login password**—the login password is initialized to the first identity for the user
 - **Delete existing owning Windows authentication**—if the Windows user is owned by a Windows authentication, the authentication is deleted after the users migrates and the new Local authentication is created

5. Select the Windows users that you want to migrate to local users.



*You can select each user by enabling the check box next to the user's name. Select **Select All** to select all users, or select **Deselect All** to clear all check boxes.*

6. Review the information on the summary menu and select **Submit** if the information is correct.



You must have already performed network integration on the system, using the Server Configuration Wizard, prior to attempting to migrate users.



Voice and fax messages do not migrate from one system to another.

Using the New User Wizard

After you finish entering information on a menu, select the **Next** button to move to the next menu. If you are missing information that the UC server needs to create a user, a warning displays with instructions about the type of information that is missing. You can move **Back** through the menus at any time to change the information that you entered. You can also select **Cancel**, which closes the wizard and deletes all the information that you entered. Restart the wizard to begin again.

If you want the new user to be linked to Active Directory, select **Windows User** on the User Type Selection menu. This mode allows user information and messaging options to be linked with Active Directory. For information about Active Directory, refer to [Understanding Active Directory on page 70](#).

If you want the new user to be exclusive to the UC server, select **Local User** on the User Type Selection menu. This mode allows user information and messaging options to be exclusively controlled by the UC server.

If the user is a SIP user, you can assign a phone. If you do not assign a phone to a new user, you can use the Phones pane to do so later. If you assign a new user to a phone that does not have buttons available, the last user that was assigned to the phone is no longer associated with it. To avoid this problem, do not assign new users to telephones that do not have at least one available button.

With the New User Wizard you can do the following tasks:

- Create a new user.
- Create initial passwords for accessing the new user profile, either over a telephone or from a computer.
- Associate the user to a message store for voice or fax messages.
- Associate the user with an identifying number, for example, an extension number.
- Define the call answering behavior.

To start the New User Wizard

1. In the Administration navigation pane, select **Users**.
2. Right-click anywhere in the Users content pane.
3. Select **New**.
4. Select **Next** to begin.

To create a Windows user

1. Follow all the steps in *To start the New User Wizard on page 88*.
2. On the User Type Selection menu, enable the **Windows User** button, and select **Next**.
3. On the User Information (Active Directory) menu, select **Browse**, and type or select the following new user name information:
 - **First name**
 - **Last name**
 - **Display name**



*You can also select a user by selecting the **Browse** button to open the Active Directory Select User box.*

4. Enable the **Include in dial by name directory** check box, if you want the user included in the dial by name directory.
5. Type an extension number in the **Identity #** box.
This extension number is also known as an **Identity**.
6. From the server list, select a server for the identity.
7. If you want the identity to belong to a phantom mailbox, enable the corresponding **Phantom** check box.



Phantom extensions can also be created for extensions with no physical phones attached.

8. Type a fax number in the **Fax #** box.
9. From the server list, select a server for the fax number, and select **Next**.
10. On the Authentication (Active Directory) menu, type a numerical telephone access PIN in the **PIN** box.



The PIN number is used to access a mailbox using the telephone user interface (TUI), either locally or remotely.

11. Retype the PIN in the **Confirm** box, and select **Next**.

12. Select Exchange, IMAP, or Local message store.
13. On the Phone Selection menu, enable one of the following:
 - Skip assigning a phone to this user
 - Assign to a new phone
 - Assign to an existing phone
14. If you selected **Assign to a new phone**, type the MAC Address and choose a phone type from the Phone Type list.

The MAC address is located on a sticker on the phone.
15. If you selected **Assign to an existing phone**, select the phone to which you want to assign the new user.



*Only existing phones that have buttons available appear in the list. To show all phones in the list, enable the **Show all phones** check box. If you select a phone that does not have buttons available, then the Replace Existing Identity dialog appears when you select Next. From the drop list, select an existing identity to replace with the current identity.*

16. Select **Next**.
17. On the Call Answering menu, select one of the following call answering modes:
 - **Personal Assistant**—provides standard voicemail features and is the default call answering mode
 - **Personal Business Assistant**—provides more advanced features for the user who wants to create personalized services and announcements
18. Select the operator for the new user:
 - **System**—the general operator, typically a company receptionist, where calls are directed to when a caller presses “0” in a voicemail greeting
 - **Personal operator**—a specific operator, for example a personal receptionist or an individual user in a particular department or location, where calls are directed to when a caller presses “0” in a voicemail greeting
19. If you selected Personal operator, type the operator’s extension number in the box.
20. Select **Next**.
21. Review the Summary of the information for the new user.



*If there is information in the Summary box that is incorrect, you can select **Back** to the appropriate menu and change it.*

22. Select **Next** to create the new user profile.



You cannot use the same name, extension, or mailbox number as another user. The extension number cannot be the same as a port extension. The UC server displays a warning if any of the above conditions exist, and the New User Wizard restarts.

To create a local user

1. Follow all the steps in [To start the New User Wizard on page 88](#).
2. Enable the **Local User** button, and select **Next**.
3. On the User Information (Local) menu, type the following new user information:
 - **First name**
 - **Last name**
 - **Display name**
4. Enable the **Include in dial by name directory** check box, if you want the user included in the dial by name directory. The **Include in dial by name directory** check box is enabled by default.
5. Type an extension number in the **Identity #** box.
6. From the server list, select a server for the identity.
7. If you want the identity to belong to a phantom mailbox, enable the corresponding **Phantom** check box.
8. Type a fax number in the **Fax #** box.
9. From the server list, select a server for the fax number, and select **Next**.
10. On the Authentication (Local) menu, in the **Password** box, type a login password for the user.
11. In the **Confirm** box, retype the login password.



If you do not type a password, the UC server assigns a 4-digit password based on the extension number. For example, extension number 300 is assigned the password 0300.

12. In the **PIN** box, type a numerical telephone access PIN for the user.
13. In the **Confirm** box, retype the telephone access PIN, and select **Next**.
14. On the Messaging menu, select an email storage type from the list.
15. If you selected Local Message Store, go to [To create a local message store user profile on page 90](#).
16. If you selected IMAP, go to [To create an IMAP user profile on page 92](#).

To create a local message store user profile

1. Follow all the steps in [To start the New User Wizard on page 88](#).
2. On the Messaging menu, select **Local Message Store** from the **Email Storage** list.
3. Enable the **Enable integrated messaging client** check box, if you want to enable message integration for the user. Integrated messaging adds a Mailboxes navigation bar to a user's UC client. The user can then manage their voice and fax messages visually through the Inbox and Sent Items folders in the UC client.
4. From the **Select the server that handles contact information** list, select a server, and select **Next**.

5. On the Phone Selection menu, enable one of the following:
 - **Skip assigning a phone to this user**
 - **Assign to a new phone**
 - **Assign to an existing phone**
6. If you selected Assign to a new phone, type the **MAC Address** and choose a phone type from the **Phone Type** list.

The MAC address is located on a sticker on the telephone.
7. If you selected Assign to an existing phone, select the phone to which you want to assign the new user.



*Only existing phones that have buttons available appear in the list. You can show all phones in the list by enabling the **Show all phones** check box. If you select a phone that does not have buttons available, then the *Replace Existing Identity* dialog appears. From the drop list, select an existing identity to replace with the current identity.*

8. Select **Next**.
9. From the Call Answering menu, select one of the following call answering modes:
 - **Personal Assistant**—provides standard voicemail features and is the default call answering mode
 - **Personal Business Assistant**—provides more advanced features for the user who wants to create personalized services and announcements
10. Select one of the following operators:
 - **System**—the general operator, typically a company receptionist, where calls are directed to when a caller presses “0” in a voicemail greeting
 - **Personal operator**—a specific operator, for example a personal receptionist or an individual user in a particular department or location, where calls are directed to when a caller presses “0” in a voicemail greeting
11. If you selected **Personal operator**, type the operator’s extension number in the box.
12. Select **Next**.



Even after you connect the user’s telephone to the PBX, the UC server cannot answer a new user extension unless you also program the PBX. It is important to program the PBX so that it directs incoming calls for the new user extension to a port that is connected to the UC server. Otherwise, the user does not have access to the unified communication service.

13. Review the Summary of the information for the new user.



*If there is information in the Summary box that is incorrect, you can select **Back** to the appropriate menu and change it.*

14. Select **Next** to create the new user profile.



You cannot use the same name, extension, or mailbox number as another user. The extension number cannot be the same as a port extension. The UC server displays a warning if any of the above conditions exist, and the New User Wizard restarts.



Regardless of the options that are selected, messages accessed by LMS users are always played in order from oldest to newest.

To create an IMAP user profile

1. Follow all the steps in [To start the New User Wizard on page 88](#).
2. On the Messaging menu select **IMAP** from the **Email Storage** list.
3. Type a mailbox number in the Mailbox box.
4. From the **Select the server that handles contact information** list, select a server.
5. Select the answering mode for the new user by choosing one of the following:
 - **Personal Assistant**—provides standard voicemail features and is the default call answering mode
 - **Personal Business Assistant**—provides more advanced features for the user who wants to create personalized services and announcements
6. Select one of the following operator types:
 - **System**—the general operator, typically a company receptionist, where calls are directed to when a caller presses “0” in a voicemail greeting
 - **Personal operator**—a specific operator, for example a personal receptionist or an individual user in a particular department or location, where calls are directed to when a caller presses “0” in a voicemail greeting
7. If you selected **Personal operator**, type the operator’s extension number in the box.
8. Select **Next**.
9. Review the Summary of the information for the new user.



*If there is information in the Summary box that is incorrect, you can select **Back** to the appropriate menu and change it.*

10. Select **Next** to create the new user profile.



You cannot use the same name, extension, or mailbox number as another user. The extension number cannot be the same as a port extension. The UC server displays a warning if any of the above conditions exist, and the New User Wizard restarts.

11. Select **Next**. Please wait while the program files are copied to the computer. The Setup Complete menu appears when the UC client application is installed.

12. Select **Finish**.

Configuring User Profiles

You can change the following types of user profile information, depending on the type of user you have configured for the system:

- Configuring a user's general information
- Configuring a user's message storage
- Configuring a user's message waiting indicator
- Configuring a user's identity
- Configuring a user's phone (refer to [Managing phones on page 123](#))
- Configuring a user's authentications



The Messaging tab is unavailable for Active Directory Windows users. For information about Active Directory, refer to [Understanding Active Directory on page 70](#). For information on how to change, and delete user profiles from Active Directory, refer to [Using the Active Directory Users and Computers Console on page 71](#).

Configuring a user's general information

You can change the following types of general user information:

- User name
- Including the user in or removing the user from the dial by name directory
- Time zone
- Call answering mode (PA/PBA)
- Operator (the extension the caller may be directed to if the user is unavailable)

User profile information includes the first, last, and display name of the user, as well as the option to include the user in the dial by name directory.

Each user profile has a time zone setting that indicates in which time zone the user is typically located. The time zone setting affects the times and dates that appear on voice, fax, and email messages when users access their mailbox using the telephone user interface. For workers who travel, or are relocating to another site for a period of time, you can change each user's time zone settings, or configure the time zone setting for multiple users.

You can specify the call answering mode, either Personal Assistant (PA) or Personal Business Assistant (PBA).

- The PA user is the default call answering mode, and includes access to all of the standard voicemail features.
- The PBA user has access to more advanced features, and is for users who want to create personalized call answering services and announcements.

When a user is unavailable, callers are transferred to an operator's extension by pressing a button in a voicemail greeting, typically **0**. You can specify the operator for the user, either System or Personal.

- A System operator is the general operator, and the call is typically transferred to the extension of the company receptionist.
- A Personal operator is an alternative to the System operator, and callers are directed to a different extension, for example, to a personal receptionist, or to an individual in a particular department or location.

You can specify the same operator number for several users.

To change a user's name information

1. In the Users content pane, right-click the user name that you want to modify, and select **Open**.
2. Change any of the following options:
 - **First name**
 - **Last name**
 - **Display name**



*If the user is a Windows user with access to Active Directory, you can select **Browse** to search the system for user name information. For information about Active Directory, refer to [Understanding Active Directory on page 70](#).*

3. Enable **Include in dial by name directory** if you want the user to be accessible by the dial by name mode.
4. Select **OK**.

To change the time zone for a single user

1. Right-click the user whose time zone setting you want to change, and select **Change time zone**.
2. Select the time zone from the list available.



*You can also change the time zone setting by right-clicking the user(s) that you want, selecting **Open**, and selecting a time zone from the list available.*

To change the time zone for multiple users

1. Press **Shift** and select each user for which you want to change the time zone setting.
2. Right-click in the content pane, and select **Change time zone**.
3. Select the time zone from the list available.



You cannot change the time zone setting for the system profile. The system profile time zone setting is always the same as the time zone of the server.

To change a user's call answering mode

1. In the Users content pane, right-click the user name that you want to modify, and select **Open**.
2. Select one of the following answering modes for the user:
 - **Personal Assistant (PA)**—this user has access to all the standard voicemail features. This is the default call answering mode.
 - **Personal Business Assistant (PBA)**—this user has access to more advanced features than the PA user. This call answering mode is for users who want to create personalized call answering services and announcements.

To select a user's operator

1. In the Users content pane, right-click the user for which you want to select an operator, and select **Open**.
2. Select one of the following operators for the user:
 - **System**—the general operator, typically a company receptionist, where calls are directed to when a caller presses “0” in a voicemail greeting.
 - **Personal operator**—a specific operator, for example a personal receptionist or an individual user in a particular department or location, where calls are directed to when a caller presses “0” in a voicemail greeting. If you selected **Personal operator**, type the operator's extension number in the box.
3. Select **OK**.



The operator's extension must be longer than one digit. An operator number does not need to be a user extension; it can be a phantom mailbox.

To select an operator for multiple users

1. Press **Shift** and select all users for which you want to specify the same operator number.
2. Right-click the selected user names, and select **Set operator identity**.
3. Type the extension number that you want in the **Operator identity** box.



The operator extension must be longer than one digit. An operator number does not need to be a user extension; it can be a phantom mailbox.

4. Select **OK**. If you leave the **Operator identity** box blank, or enter the system operator extension, you are prompted to confirm the selection.



If you do not specify an extension number for a user, callers are transferred to the system's operator extension number when they dial '0'.

Configuring a user's message storage

For each user profile you can specify the email storage type, mailbox, and server, which determines where voice, fax, and email messages are stored. You can also select a server to handle contact information, including contact matching for messages, call routing based on contacts, and other mailbox functions such as contact searches for managing your mailbox, forwarding messages to contacts, and reply or callback to contacts.



*The Messaging tab is linked to the user profile mailbox selection. The Messaging tab is not visible for Microsoft Active Directory® users where **Use mailbox defined by Active Directory** is enabled. For more information, refer to [Understanding Active Directory on page 70](#).*

When selecting a message storage server, consider the following:

- Selecting **Exchange** means that the user's messages are stored on Microsoft® Exchange Server.
- **IMAP** (Internet Message Access Protocol) is a method of accessing and storing email messages. This is the preferred configuration for Lotus Notes and Lotus Domino servers.
- Selecting **Local Message Store** means that the user's messages are stored locally, on the UC server.

When you configure a user's messages to be stored on Exchange Server, there are some additional options available related to message delivery, and to mailbox monitoring, which affects message waiting indicators.

The message delivery mode determines whether voice and fax messages are sent from the system mailbox or are stored directly in the user's mailbox. This setting affects whether:

- User defined message rules in Microsoft Outlook® or Microsoft Exchange Server are applied
- Messages are sent to third-party applications such as BlackBerry® Enterprise Server
- The caller's phone number appears in the From line of the message

The settings below determine message delivery for the individual user selected, however message delivery can also be configured system-wide.

Use system-wide setting

Use the system-wide setting for message delivery configured for Exchange Server. This is the default value. For more information on the system-wide setting, refer to [Configuring Exchange Server on page 138](#).

Send messages from system mailbox

Because voice and fax messages are sent from the system mailbox:

- User defined message rules set in Outlook or Exchange Server are applied.

For example, a user's rule might direct messages from a particular contact to a specific folder.

- Messages are sent to any third-party applications such as BlackBerry Enterprise Server.
- The message appears to be from the system mailbox, so the From line does not include the caller's phone number.

Impersonate caller and store to mailbox

Because voice and fax messages are stored directly in the user's mailbox, rather than being sent:

- User defined message rules in Outlook or Exchange Server are not applied.

For example, a user's rule directing messages from a particular contact to a specific folder will not be applied.

- Messages are not sent to third-party applications such as BlackBerry Enterprise Server.
- The From line includes the caller's phone number, or the caller's name from a contact that matches the phone number.

Mailbox monitoring ensures that when a user accesses their voicemail messages through Microsoft Outlook Web Access or through a mobile device, that their message waiting indicator on their phone updates accordingly. Without mailbox monitoring enabled, the UC server is unaware when a user accesses their voicemail messages through Outlook Web access or through a mobile device. For example, a user's message waiting indicator on their phone may continue to flash even after they've accessed all their new messages.



Mailbox monitoring works in conjunction with the message waiting indicator options set. For example, the message waiting indicator options determine if the message waiting indicator is turned off when the first message is read, or when all new messages have been read. For more information, refer to [Configuring message waiting indicator options on page 21](#).



In order to enable mailbox monitoring for individual users, you must also configure Exchange Server to enable mailbox monitoring. For more information on configuring Exchange Server to enable mailbox monitoring, refer to [Configuring Exchange Server on page 138](#).



Performance issues may arise when mailbox monitoring is enabled. For more information, refer to [Configuring Exchange Server on page 138](#).

Use server default	Use the server default setting selected when configuring Exchange Server. This is the default value.
Enabled	Enabling mailbox monitoring opens the Exchange Server mailbox and starts monitoring the mailbox at system start up. This applies only to the Inbox, and other folders are not monitored. When the user accesses messages through Outlook Web Access or through a mobile device, the message waiting indicator synchronizes accordingly.
Disabled	If mailbox monitoring is disabled, then when the user accesses messages through Outlook Web Access or through a mobile device, the message waiting indicator does not synchronize accordingly. The message waiting indicator turns on when a new message arrives, and it is managed, according to the message waiting indicator options set, when messages are accessed through the regular Outlook client.

To configure a user's message storage

1. In the Administration navigation pane, select **Users**.
2. In the Users content pane, right-click the user whose information you want to change, and select **Open**.
3. In the **Edit User** dialog box, select **Messaging**.
4. Select an email storage type from the list available.



The options that appear on the Messaging tab depend on the Email Storage type selected.

To store a user's messages locally

1. On the **Messaging** tab, for Email Storage select **Local Message Store**.
2. Select the **Enable integrated messaging client** check box, if you want the user to have access to integrated messaging. Integrated messaging adds a Mailboxes navigation bar to a user's UC client. The user can then manage their voice and fax messages visually through the Inbox and Sent Items folders in UC client.
3. Select a server from the **Select the server that handles contact information** list.

To store a user's messages on Exchange Server

1. On the **Messaging** tab, for email storage select **Exchange**.
2. Select an Exchange Server mailbox from the list available.
3. From the list of servers available, select a server to handle the user's contact information.

4. Select a message delivery mode from the following options:
 - **Use system-wide setting**
 - **Send messages from system mailbox**
 - **Impersonate caller and store to mailbox**
5. Select a mailbox monitoring mode from the following options:
 - **Use server default**
 - **Enabled**
 - **Disabled**

Configuring a user's message waiting indicator

You can set how the message waiting indicator operates on users' phones after the UC server receives a message. The instructions below describe how to set the message waiting indicator behavior for an individual user, however you can also set the message waiting indicator behavior system wide. Users can also set their own message waiting indicator behavior.

Because message waiting indicator behavior can be set by both the administrator and the user, and the administrator can set it both system wide and individually, sometimes these settings may differ. The message waiting indicator behavior in effect corresponds to the last mode set. Thus, if you set the message waiting indicator behavior for an individual user, then this setting takes effect rather than the system wide or the user selected mode. However, users can reset the message waiting indicator behavior by selecting an mode themselves.



For message waiting indicators to update when messages are accessed in Microsoft Outlook Web Access or on a mobile device, you can configure Microsoft Exchange Server to monitor for changes in users' mailboxes, and for the UC server to adjust the waiting light accordingly. For more information, refer to [Configuring Exchange Server on page 138](#).

To change a user's message waiting indicator behavior

1. In the Users content pane, right-click the name of the user whose message waiting indicator behavior you wish to change.
2. Select **Open**.
3. Select the **message waiting indicator** tab.
4. Choose one of the following options.

Use system defaults

The message waiting indicator behavior for the user is set according to the system wide settings made in the **Tool > Options > message waiting indicator** dialog.

This is the default setting for new and upgraded users.

Never turn on or off

The UC server does not manage the message waiting indicator

Turn on when new message arrives**Turn off when first message is accessed in mailbox**

The light turns off when the user accesses the first new message, regardless of whether the user disconnects before listening to the message.

Turn off when no more new messages in mailbox

The light turns off when the user accesses the last new message.

5. Select **OK**.



Any changes to the message waiting indicator options come into effect immediately.



For information on setting the message waiting indicator options system wide, refer to [Configuring message waiting indicator options on page 21](#)



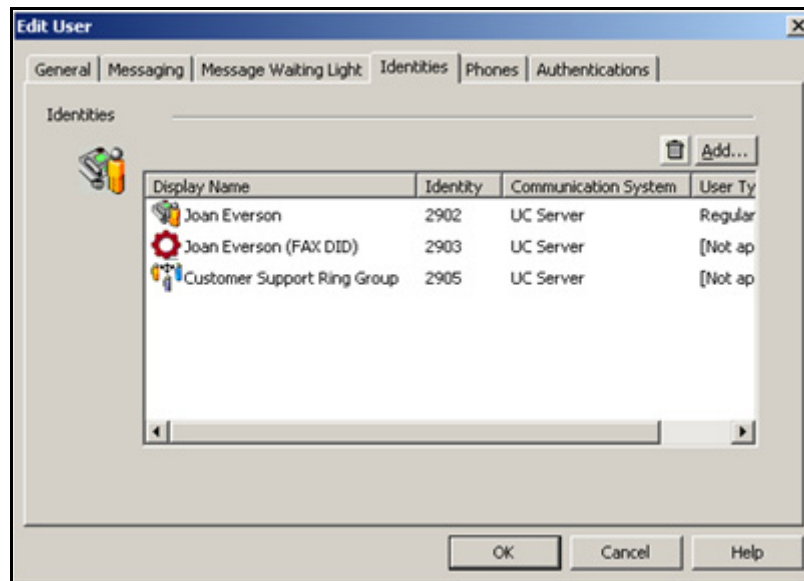
For troubleshooting information, refer to [Troubleshooting Message Waiting Indicators on page 317](#).

Configuring a user's identity

An identity can be an extension assigned to a user, an auto attendant, or receive incoming faxes. An identity can refer to a user extension, a hunt or ring group, an auto attendant, a fax extension, or an email address. An identity is associated with a specific extension or SIP identity for a hard or soft phone.

A user can have up to two identities, for example, one for a telephone, and another for a softphone. Any additional identities require an additional license. Users can have more than one identity, but identities are always associated with only one user.

You can create a fax identity to receive incoming faxes for a user, and direct the faxes to the user's Inbox.



You can start the New Identities Wizard to create an identity for a user using the Users content pane. An identity that you create in the Users content pane also appears, and can be managed, in the Identities pane.

When you create an identity, SIP devices require both a SIP authorization and password to successfully register with UC server SIP registration. The SIP authorization and password are associated with a user identity. Some phones, for example, Polycom phones, support automatic configuration. For more information, refer to [To get the SIP authorization and password on page 124](#)



When configuring an identity on a SIP telephone that does not support automatic configuration, consult ADTRAN integration notes and manufacturer documentation for the specific configuration for your environment.

To create an identity in the Users content pane

1. Right-click the user to which you want to associate an identity.
2. Select **Create Identity**. The New Identity Wizard starts. For information about using the New Identity Wizard to create a new identity, refer to [Using the New Identity Wizard on page 114](#).

To add an identity to an existing user profile

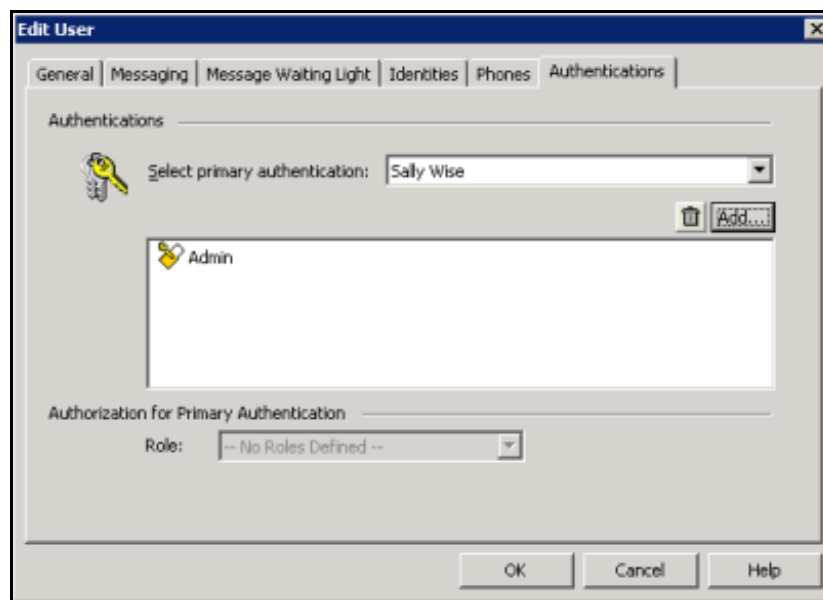
1. In the Users content pane, right-click the user name that you want to add an identity to.
2. Select **Open** to open the User dialog box.
3. Select the **Identities** tab.
4. Select **Add...** to add an identity to the user profile.
5. Select **OK**.

To delete an identity from an existing user profile

1. In the Users content pane, right-click the user name that you want to delete an identity from.
2. Select **Open** to open the User dialog box.
3. Select the **Identities** tab.
4. Select the identity that you want to delete.
5. Select the **trash** icon.
6. Select **OK**.

Configuring a user's authentications

An authentication is a means by which a person can access the system, for example, a user name or password. Many user profiles can have many authentications, and vice versa. Therefore, when an authentication is used to access the UC server, the user might have access to several user profiles, each with several identities. For more information about managing authentications using the Authentications user pane, refer to [Enabling Access to User Profiles using Authentications on page 105](#).



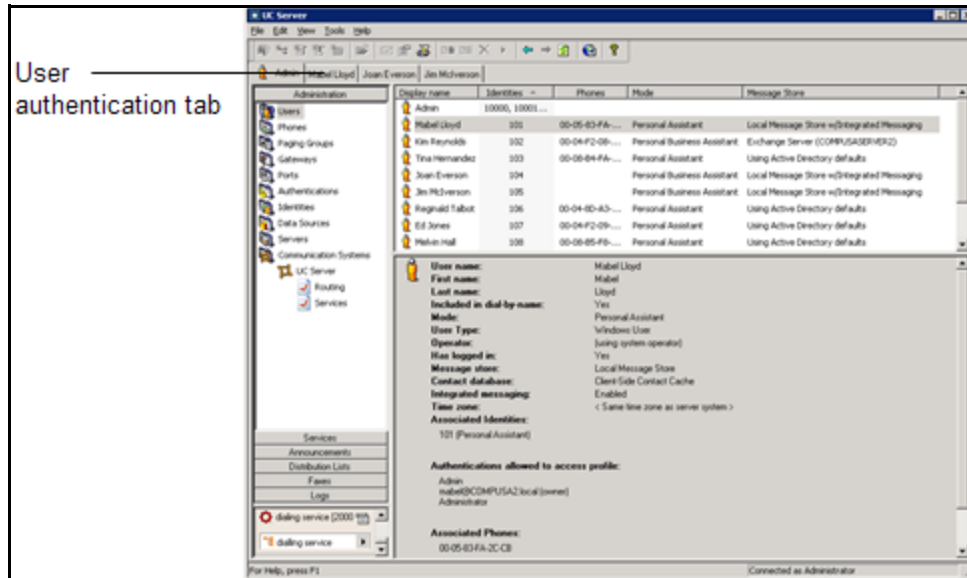
The primary authentication is the main authentication that owns the user profile. When a user logs in with the primary authentication, the user profile is on the main tab.



Although it is possible to remove the primary authentication, ADTRAN does not advise doing so.

When you add an authentication to a user profile, the authentication then has access to another user's UC server client application. The profiles that a user has authorization to access are displayed as a horizontal list of tabs across the top of the application window. The user selects the tab of another user to change their greetings, change transfer options, or any other function that they might be required to perform. You can

change some aspects of authentications using the Users content pane. For full authentications management, use the Authentications pane.



You must restart the application to view changes to user authentication tabs.

To add an authentication to an existing user profile

1. In the Users content pane, right-click the user name that you want to add an authentication to.
2. Select **Open** to open the User dialog box.
3. Select the **Authentications** tab.
4. Select **Add...** to add an authentication.
5. Select a user name from the Select Authentication list.
6. Select **Select**.
7. Repeat steps 4 to 6 until you have added all the users that you want.
The users that you select appear in the Authentications list.
8. Select **OK**.

To delete an authentication from an existing user profile

1. In the Users content pane, right-click the user name that you want to remove an authentication from.
2. Select **Open** to open the User dialog box.
3. Select the **Authentications** tab.
4. Select the authentication that you want to delete.

5. Select the **trash** icon.



Trash icon

6. Select **OK**.

Resetting a user's PIN

Users enter a PIN when they access their mailbox using the telephone. Reset the PIN for users on the system for added security, or if a user forgets the telephone PIN to retrieve messages over the telephone.

To reset a PIN for a user

1. Right-click the user name for which you want to reset the PIN.
2. Select **Reset PIN #** to open the **Change Personal Identification Number (PIN)** dialog.
3. Type a new number in the **New PIN** box.
4. Retype the new number in the **Confirm new PIN** box.
5. Select **OK**.



NOTE

The new PIN takes effect immediately.

Deleting a user

To delete an existing user profile

1. Right-click a user name.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion. You have deleted the UC server user profile. All that user's data is lost. For example, all services the user created, all greetings the user recorded, and all announcements are deleted.



CAUTION

If the local message store user also has messages stored on the UC server, the messages are deleted.



NOTE

The user's email account is not affected when you delete the UC server user profile.

Enabling Access to User Profiles using Authentications

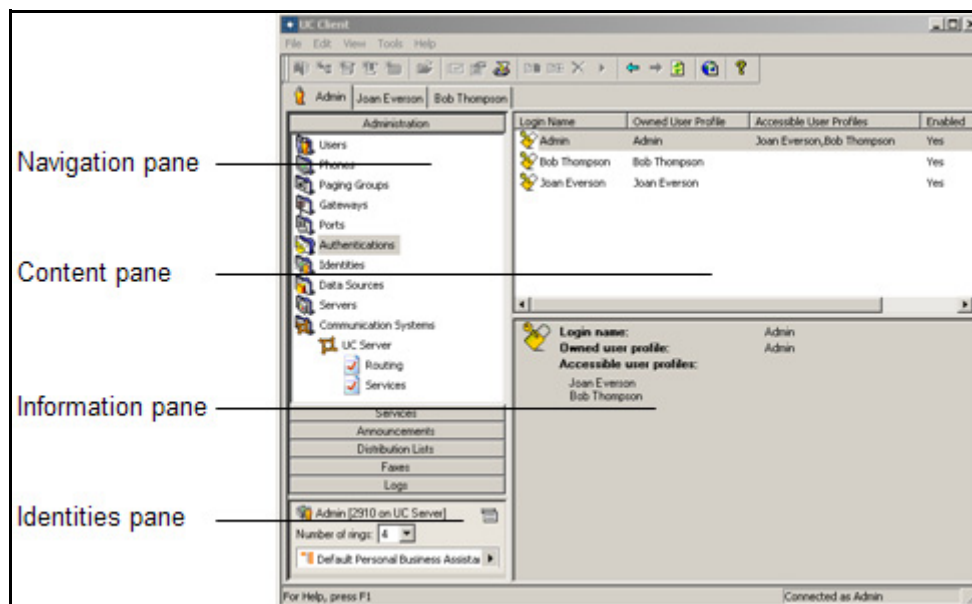
UC server allows you to manage multiple users using a single client application. An authentication is a set of credentials that either allow or deny access to system resources. You can allow specific users to have access to the profiles of other users. For example, department managers might want to have access to all the employee accounts so that they can change everyone's service at once.

There are two types of authentication:

- **Windows user authentication**—whose credentials are based on the single sign-on Active Directory account into which the user is logged
- **Local user authentication**—which is created and managed by the UC server. When the authentication type is local, users must type a user name and password each time they log on.

Authentications are either dependent on an Active Directory user, or are managed by the UC server. If authentications depend on Active Directory, single sign-on can be used for managing the associated profiles and for automatically viewing voice and fax messages for Exchange or Lotus Notes integration.

Authentications are automatically created when a user is added using the New User Wizard. Manage authentications only when you want to add authentications that have not already been created for a Windows user, or that do not have an associated user profile.



Single sign-on

Single sign-on is accomplished in Microsoft networks by ensuring the following:

- a Windows Active Directory user is created on the domain
- a UC server Windows user is created

Delegation of control

The administrator can grant other users the ability to manage aspects of the system. For example, if the administrator adds the authentication “Bob Jones” to the Administrator user profile, “Bob Jones” can now manage the UC server.

Authorization roles

You can also enable Windows authentication, which allows you to assign a role to a Windows user. Authorization is the process by which it is determined whether a user can perform a certain operation on some object or objects (for example, identities, profiles, ports, announcements, faxes). You can add or remove users from specific roles. If you have created a custom authorization store on your system, you can choose from the following roles:

- **Administrator** — authentications have full access, for example, are fully able to manage system objects (like users and ports), and they can create, change, and delete objects in shared folders
- **Read Only Administrator** — authentications can do almost everything that Administrator users can do, except that they cannot create, change, and delete system objects
- **Restricted User** — authentications with capabilities typically associated with Personal Assistant and Personal Business Assistant users, except that they cannot create or delete announcements, and can only read services
- **Standard User** — authentications with capabilities typically associated with Personal Assistant and Personal Business Assistant users
- **Executive Assistant** — authentications are the same as Standard Users, but can also manage objects that belong to other profiles to which they have access. For other profiles, their authorization does not allow them to create or delete announcements, and allows them to only read services.
- **Restricted Executive Assistant** — authentications that have Restricted User capabilities for their own profile and the same authorization for other profiles to which they have access
- **Power User** — authentications similar in capability to Executive Assistant, except that they can also create or delete announcements for other profiles and have full authorization for services



If there is no custom store, all Windows authentications are treated as if they belong to the Administrator role.

A user’s authorization for a specific profile depends on the following:

- The role associated with a user’s authentication
- The user’s profile type, for example, System, Personal Business Assistant, or Personal Assistant

To create a new authentication for an existing user profile

1. Right-click anywhere in the Authentications content pane, and select **New**.
2. Type a login name for the new authentication in the **Login name** box.
3. Make sure the **Enabled** check box is enabled.
4. Type a password in the **Password** box.

5. Retype the password in the **Confirm** box.
6. Select the **Change** button to select an owned user from the list of available user profiles.
7. Select the **Add** button to add accessible users to the new authentication.
8. Select **OK**.

To assign a Windows user to a role

1. Right-click anywhere in the Authentications content pane, and select **Enable Windows Authentication**.
2. Select **Browse**.
3. Select a user from the Select User box.
4. Select **OK**.
5. In the **Authorization** section of the Enable Windows user box, select a role from the **Role** list. The list is disabled if there are no authorization roles available.
6. Select **OK**.

To add an accessible user to an existing user profile

1. In the Authentications content pane, right-click the Login Name of the user to which you want to allow another user access, and select **Open**.
2. In the **User associations** section, select the **Add** button.
3. In the **Select accessible user profile** box, select the user that you want to have access.
4. Repeat step 4 until all the users that you want are listed in the Accessible users box.
5. Select **OK**.

To remove an accessible user from a user profile

1. In the Authentications content pane, right-click the Login Name of the user for which you want to remove another user's access, and select **Open**.
2. In the **Accessible users** section, select the name of the user that you want to remove.
3. Select the **trash** icon.



Trash icon

4. Select **OK**.

Managing User Profiles from EBS

If your system uses Windows Essential Business Server (EBS) 2008 then you can manage UC server users from within the EBS' Administration Console.



To access UC server user profiles from EBS, the UC client must be installed on EBS.

When the UC client is installed on EBS, then under Business Applications a tab is added for UC server Users. The information provided in the tab is the same information that is available from the UC client Users contents and preview panes. Therefore, you can add, edit, and delete UC server users from within the EBS Administration Console, just as you would in the UC client. Changes made to UC server user profiles from EBS are automatically updated in the UC client. From within the UC Server Users tab, you can also open the UC client, or connect to another server.

To access NetVanta UC Server user profiles in EBS

1. Run EBS' Administration Console and select the **Business Applications** tab.
2. Select the **UC Server Users** tab.
3. If prompted, select the server to login to.
4. If required, login as Administrator, or a user with administrative privileges. A list of UC server users appears, with the same headings and information available that appear in the UC client Users content pane.
5. Select a user. The user's information appears in the pane below, similar to the preview pane in UC client.

To add a NetVanta UC Server user from EBS

1. From the right-hand pane, select **Add user**.
The UC server New User Wizard opens.
2. Follow the steps for the New User Wizard to add a new user. Refer to [Using the New User Wizard on page 87](#).

To edit a NetVanta UC Server user from EBS

1. Select a user.
2. From the right-hand pane, select **Edit user**. The Edit user dialog opens.
3. Follow the steps for the Edit User dialog to edit a user profile. Refer to [Configuring User Profiles on page 93](#).

To delete a NetVanta UC Server user from EBS

1. Select a user.
2. From the right-hand pane, select **Delete user**.
3. At the prompt, confirm that you want to delete the user.

7 Managing User Extensions and Identities

This chapter introduces user extensions and identities and how to manage them. It includes the following topics:

- *Overview on page 109*
- *Auto attendant Identities on page 112*
- *Ring and Hunt Group Identities on page 113*
- *Identities and ITSP Gateways on page 114*
- *Using the New Identity Wizard on page 114*

Overview

An identity, which is defined as a dialable entity, corresponds to any of the following: user extension, auto attendant, or email address. Using the Identities menu, you can create a new **user**, or **attendant**, or **group** identity, and you can edit or remove an identity.

The UC server uses the following concepts as identities:

User identity

- Traditionally, an extension number that is dialed by another telephone or automated attendant
- On which call forwarding is configured, and is intended to be used on ucCompanion, a soft-phone, or a hard-phone

Attendant identity

- An entry in the dial plan that is typically used for an automated attendant or a service (for example, Interactive Voice Response, pre-screening, and so on)
- Differs from a user identity because they cannot be associated with a telephone, and no call forwarding is configured

Group identity

- Contains groups of user identities
- Allows you to configure call forwarding behavior (also referred to as overflow behavior) for the group

Depending on the base bundle type in use, a user may have one or more identities by default (typically one for basic bundles and two for other bundles). For example, when two identities are available for each user, one identity can be used for a telephone and another for a softphone. Any additional identities requires an additional license. To increase the number of identities, contact your reseller or ADTRAN Sales. To contact ADTRAN Sales, you can email them at sales@adtran.com, or visit the ADTRAN website at www.adtran.com.

To determine the maximum number of identity licenses per user

1. From the Administrator account, select **Help > License** Information.
2. Go to the **Number of additional per user SIP identities**, and note the license value.
3. Go to the **Number of additional pooled SIP identities**, and note the license value.
4. Add the sum of the two values noted above, to determine the maximum number of identity licenses per user.

You can associate multiple identities to a single user profile and each identity can have a different behavior that is managed by the owner of the profile. This behavior can be defined by assigning a service to the identity by the administrator when the service is created. Services are essentially customized ways of answering calls. By default:

- The *Default Trunk Service* is assigned to identities associated with the administrator account.
- The *Default Personal Business Assistant* is assigned to identities associated with user accounts.

Administrators can only assign services in the user's service folder or in the *Shared* services folder to identities associated with a user. Similarly, administrators can only assign services in the administrator's services folder or the *Shared* services folder to identities associated with the administrator account. Once created:

- The administrator can change the service attached to identities associated with the administrator account, but cannot automatically change the services of an identity associated with a user account.
- Users can change the services attached to the identities associated with their account through the UC client.

If an administrator wants to change the service associated with a user identity, then the administrator can temporarily add their user authentication to the authentications of the user account. This allows the administrator to access the user account through a new tab in the UC client, once they close and re-launch the UC client. The administrator can then modify the service associated with the user identity in the user tab and then subsequently remove their authentication from the user account. The tab is removed when the UC client is restarted. For more information about services and how to build and activate them, refer to [Managing Services on page 214](#), or [Creating Custom Call Answering Services on page 213](#).

Use an attendant identity to connect a caller directly to an auto attendant or to a service (for example, Interactive Voice Response, pre-screening, and so on). When callers dial in to the system, they can access, for example, a user extension, or the auto attendant, or a hunt group.

You can create a fax identity to receive incoming faxes for a user, and direct the faxes to the user's Inbox.

An identity is associated with a communication system: identity = communication system + address. Typically an identity is a numeric extension, however, in SIP, an identity can also be alphanumeric.



You can have the same address on different communication systems. For example, an identity on SIP can equal 2000, and a communication system extension can be 2000.

When you create an identity, SIP devices require both a SIP authorization and password to successfully register with UC server SIP registration. The SIP authorization and password are associated with a user identity. Some phones, for example, Polycom phones, support automatic configuration.



When configuring an identity on a SIP telephone that does not support automatic configuration, consult ADTRAN integration notes on the ADTRAN website at <http://kb.adtran.com>, and manufacturer documentation for the specific configuration for your environment.

The **New Identity Wizard** steps you through the process of creating a new identity.



The New Identity Wizard is dynamic and prompts you to enter different information depending on the configuration options that you choose. For example, configuring an identity type, associating a user with an extension number, or defining call answering behavior. The procedures below are only a few of the possible wizard menus that you might view when creating an identity.

You can create the following types of identities using the wizard:



User Identity

This type belongs to a user and is typically associated with ucCompanion, a soft-phone or a hard-phone.



Attendant Identity

This type allows calls to route directly to an auto attendant or to a service (for example, Interactive Voice Response, pre-screening, and so on)



Group Identity

This type allows you to create either a Ring group or Hunt group extension, which distributes calls to members of these group types.

Ring Group

A **Ring** group creates an identity that distributes calls to all members of the ring group at the same time.

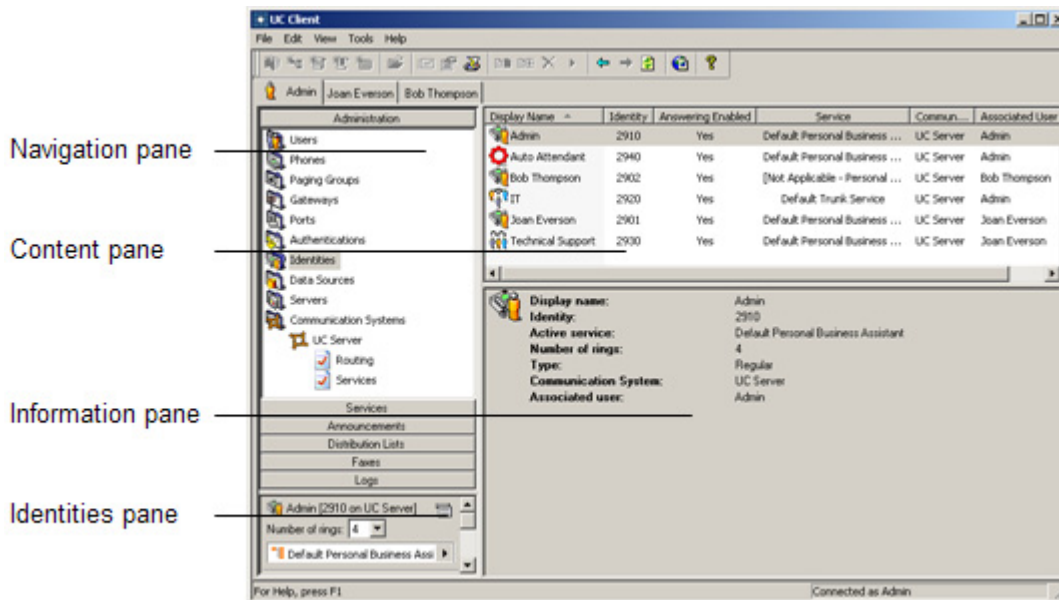


Hunt Group

A **Hunt** group creates an identity that distributes calls to members of the hunt group sequentially, based on availability.

Each identity type has a distinctive icon that appears in the content pane. By default, the content pane displays only the auto attendant service. To change the view to display all the extensions on the system,

select the **View** menu, and **Display Identities for all Profiles**.



Auto attendant Identities

An attendant service connects callers directly to an auto attendant or to a service (for example, IVR, pre-screening, and so on). When the UC server was configured to work with your system, each of the attendant lines was set to run a default service. This means that the attendant lines already provide basic extension dialing, dial-by-name support, operator access, and the ability for users to access and manage their mailboxes remotely.

For the UC server to use a new attendant, the Communication System must be programmed to do so. If automatic Communication System is supported for your Communication System, the UC server can program the Communication System so that incoming calls for the new attendant are directed to a port that is connected to the UC server.

The UC server might also support DID (direct inward dial) attendants, depending on the switch compatibility.

You can create an attendant service and configure it to receive faxes. Incoming faxes are then directed to the user's mailbox, for example, through Microsoft Outlook, Microsoft Exchange Server, Lotus Notes, or through an Integrated messaging client (a Mailboxes navigation bar on the user's UC client) or IMAP4. The message waiting indicator (MWI) flashes to indicate that there is a new message waiting, and a voicemail message indicates that a fax has been received.

You can assign a fax identity to a new user using the New User Wizard, by entering a Direct Inward Dial (DID) fax number for the user. Alternatively, you can create a fax identity for an existing user using the New Identity Wizard, and selecting the Receive Fax mode. Either method has the same result in that the fax number or address entered is configured to receive incoming faxes, and incoming faxes are directed to

the user's Inbox.



You cannot configure or delete a user DID fax attendant using the identities menu. You must manage fax attendants using the Users menu.

The options available for an attendant identity depend on the type of user selected. Admin and Personal Business Assistant (PBA) users can be configured to run a service. The PBA and Personal Assistant (PA) users can be configured to receive faxes in their mailbox.



For information about selecting a male or female voice for system announcements, or the order in which callers enter a name, either first or last, refer to [Setting Auto attendant Options](#) on page 29.

Ring and Hunt Group Identities

You can create ring or hunt groups, and you can include external numbers as part of the group.

Ring Group A ring group is a group of identities that are all contacted simultaneously when a call comes in. For example, a sales department might include several sales agents in a ring group. When a call comes in, the call rings simultaneously at each agent's phone, so that any available agent can take the call.

Hunt Group A hunt group is a group of identities that are contacted in sequential order when a call comes in, until the call is answered. For example, a customer service department might have several agents in a hunt group. If the first agent does not answer the call after the specified number of rings, then the call is forwarded to the next agent in the group, until the call is answered.

When you add an external number to a group identity, it is automatically associated with one of your external gateways. To support the feature, the external SIP PSTN gateway must provide Answer Supervision. Answer Supervision is the ability for the phone company to signal that the remote party has answered the call. For information about your gateway's ability to provide Answer Supervision, refer to the documentation provided with your gateway.



If you add an external number to a ring group or hunt group and your gateway does not support true answer supervision, the group stops presenting the call to all the other internal extension numbers and rings only the external telephone number.

Configuring mobile twinning using a ring group identity

Mobile twinning refers to *twinning* devices so that using one phone number, a call rings at more than one device simultaneously. For example, an internal extension can be twinned to a cell phone. When a call comes in, both devices ring simultaneously. If a user is away from their desk, they can then answer the call on their cell phone. This allows mobile users to be available for calls while away from their desk or their

office. The benefits of mobile twinning include:

- Being accessible by one-number, and having one voicemail box to manage.
- Twinned calls that are placed to your cell phone can easily be moved to your desk phone, using the connected call pickup feature.
- Mobile twinning increases your availability to customers, while minimizing costs.

The connected call pickup feature allows users to seamlessly transfer a call from one device to another, for example, from a cell phone to a desk phone, resulting in cost savings. For more information, refer to [Changing phones during a call on page 164](#).



The connected call pickup feature is very powerful in that it allows any user to pickup any connected telephone call that went through the UC server, as long as they know the number of the phone that answered the call, for example, the extension number or the cell phone number. If your company policy restricts this behavior, then follow the instructions to disable connected call pickup. For more information, refer to [Restricting picking up connected calls on page 50](#).

Identities and ITSP Gateways

If you are using an ITSP gateway, check whether your ITSP requires the use of the E.164/ENUM prefix, that is, a + symbol before the number. An incoming call must match an identity exactly in order for the call to be routed accordingly. However, the + symbol is also an operator within a regular expression that means, match the preceding element one or more times. If an incoming call comes through an ITSP gateway with a + symbol before the number, it is important that the UC server matches the + character, and not the + regular expression operator.

If you are using an ITSP gateway that requires the use of the E.164/ENUM prefix, the + symbol, then when creating your identities, add a \ which comments or exceptions out the regular expression operator +. Follow the \ with a +, to indicate to the UC server to match the + character. For example, instead of entering the identity, **613-555-1234**, you would enter, **\+613-555-1234**. A call coming through the ITSP gateway as **+613-555-1234** is then routed accordingly to **613-555-1234**.

Using the New Identity Wizard

After you finish entering information on a menu, select the **Next** button to proceed to the next menu. If you are missing information that the UC server needs to create an identity, a warning displays with instructions about the type of information that is missing. You can move **Back** through the menus at any time to change the information that you entered. You can also select **Cancel**, which closes the wizard and deletes all the information that you entered. Restart the wizard to begin again.

You can also configure a phantom identity. A phantom identity is not physically configured on a communication system, which means that there is no message waiting indicator functionality. Also, any transfer to a phantom identity automatically runs the behavior associated with that identity. A phantom identity does not ring at a phone, but goes directly to voicemail.

To start the New Identity Wizard

1. Right-click anywhere in the Identities content pane.
2. Select **Create Identity...** to display the New Identity Wizard welcome menu.
3. Select **Next** to proceed with creating a new identity.

To select a new identity type

1. Follow all the steps in [To start the New Identity Wizard on page 115](#).
2. On the Select Identity Type menu, select a communication system from the **Select a communications system** list.
3. From the **Select a user profile** list, select the user profile to which you want to assign the new identity.
The user profile that you select can manage the behavior of that identity.
4. In the **Select the class of identity** section, enable one of the following buttons.
 - **User**—creates an identity that is associated with ucCompanion, a soft phone, or a hard phone
 - **Attendant**—creates an auto attendant identity that handles calls as an automated attendant or service (for example, IVR, pre-screening, and so on)
 - **Ring group**—creates an identity that distributes calls to all members of the ring group at the same time
 - **Hunt group**—creates an identity that distributes calls to members of the hunt group sequentially, based on availability
5. Select **Next** to proceed.

To configure a new user identity

1. Follow all the steps in [To start the New Identity Wizard on page 115](#).
2. From the **Select a communications system** list, select a communication system.
3. From the **Select a user profile** list, select a user profile to which you want to associate the identity.
4. Select **User** as the new identity type.
5. On the Configure User Identity menu, type a name for the new user in the **Name** box.
6. In the **Address** box, type an address for the new user.
An address is typically an extension number.



Ensure that you do not add an extension that is restricted in the dial plan. For example, by default the UC server restricts the following dialing prefixes: 0, 1, 411, 611, 911, 1900, and 1976. If a call is routed to an extension number starting with one of these restricted numbers, then the call will not be completed. For more information, refer to [Routing calls using the dial plan on page 43](#).

7. If you want to configure the identity as a phantom identity, enable the Phantom check box.
8. From the **Run service** list, choose the service that you want to assign to the new identity.
9. Select **Next** to proceed.

10. On the Configure SIP Authentication menu, type a password in the **Password** box. The password allows the phone to connect to the communication system.
11. Select **Finish** to create the new user identity.



*The new user identity appears in the Identities content pane beside its distinctive icon. By default, only the auto attendant identities appear, but you can display all the identities by selecting **View > Display User Identity Details**.*

To configure an attendant identity

1. Follow all the steps in [To start the New Identity Wizard on page 115](#).
2. From the **Select a communications system** list, select a communication system.
3. From the **Select a user profile** list, select a user profile to which you want to associate the identity. The user profile that you select is able to manage the behavior of this identity.
4. Select **Attendant** as the new identity type.
5. On the Configure Attendant Identity menu, type a name for the new attendant identity in the **Name** box.
6. In the **Address** box, type an address for the new attendant identity. An address is typically an extension number.
7. From the **Run service** list, choose the service that you want to assign to the new attendant identity. This option is available to Admin and PBA users only.
8. Select **Next** to proceed.
9. Select **Finish** to create the new identity.



*The new identity appears in the Identities content pane beside its distinctive icon. By default, only the auto attendant identities appear, but you can display all the identities by selecting the **View menu > Display Identities for all Profiles**.*

To configure a fax attendant identity

1. From the **Admin** tab, select **Identities**.
2. In the content pane on the right, right-click and select **New Identity**.
3. Select **Next** at the Welcome dialog.
4. Select a communication system.
5. Select the user for whom you want to configure a fax identity. The user profile that you select is able to manage the behavior of this identity.
6. Select **Attendant** identity, and select **Next**. The options that appear on the Configure Attendant Identity dialog depend on the user profile selected in the previous dialog. Admin users do not have the Receive Fax mode.
7. Type a display name for the fax attendant identity. This is the display name that appears in the Identities content pane.

8. Type an address for the fax attendant identity. An address is typically an extension number, but it can be any string of numbers that can be dialed and is unique to a user, such as an email address. The address is the identity number that callers use to send faxes to the user.
9. Select **Receive Fax** to configure the attendant identity to receive incoming faxes, and select **Next**. Incoming faxes are then directed to the user's Inbox. The MWI flashes, and a voicemail message indicates that a new fax has been received. This option is unavailable for Admin users.
10. Select **Finish** to create the fax attendant identity.

To configure a new group identity

1. Follow all the steps in [To start the New Identity Wizard on page 115](#).
2. Select **Ring group** or **Hunt group** as the new identity type.
3. Select **Next**.
4. In the **Name** box, type a name for the new group identity.
5. In the **Address** box, type an address for the new group identity.
6. Select **Add...** to open the Select a Member Identity box.
7. If you want to add an external number to the group identity, select **Add External**.



If your gateway does not support true answer supervision, the ring group stops presenting the call to all the other internal extension numbers and rings only the external telephone number.

8. Select the user identity that you want to add to the group.
9. Select **Select**.
10. Repeat steps 6 to 8 until all the identities that you want in the group are listed in the Members box.



Use the up and down arrow keys to change the order of the identities in the group.

11. Select **Next**.
12. From the **Determine the number of rings before overflowing to the next member of a hunt group and from a hunt/ring group to an alternate call answering behavior** list, select the number of rings.
 - For a ring group, the number of rings determines the number of rings that are made simultaneously to all the members of the ring group before the call is redirected to the overflow behavior.
 - For a hunt group, the number of rings determines the number of rings that are made to a member of the hunt group before the call is redirected to the next member of the group. The overflow behavior only occurs after the number of rings has been made to the last member of the hunt group.
13. From the **Run service** list, select an overflow service for the group identity. If the call is not answered after the specified number of rings for a ring group, or the specified number of rings for each member of a hunt group, the service that you select answers the call.

14. Select **Next**.
15. Select **Finish**.

To configure mobile twinning using a ring group identity

1. Create a ring group identity for the user, using the steps above in *To configure a fax attendant identity on page 116*.
2. Include any numbers in the ring group for which the user can be contacted at, for example, both their internal extension number and their cell phone number. For example, if the extension number is 300, then change the extension identity to 3001, create a ring group as 300, and include 3001 and the external number as members of that ring group. When a call is transferred to 300, it rings at both 3001, their desk telephone, and their cell phone number. A call that is answered on a users cell phone number (the call must have been made to, transferred, twinned or forwarded from the UC server) can then be picked up on another phone, such as their desk phone. From the phone that the user wants to pickup the call on, the user dials the connected call pickup feature code *27 <called phone number>. The called phone number is the number of the phone that initially answered the call, such as the cell phone number. If restrictions were set on using this feature, then the user may only need to dial *27 to pickup the call from their desk phone.

To display all the identities on the system

- From the Identities content pane, select the **View** menu > **Display Identities for all Profiles**.

To edit a group identity

1. In the Identities content pane, right-click the group identity that you want to edit.
2. Select **Open**.
3. Type changes in any of the following boxes:
 - **Display Name**
 - **Address**
 - **Members**—type the addresses of the members that you want to include in the group
 - **Number of Rings**—type the number of rings after which you want the call to be redirected. For a ring group, all members will ring simultaneously after which the call is redirected to the overflow behavior. For a hunt group, each member of the group will ring individually, after which the call is redirected to the next member of the group. If the last member of the group does not answer, then the call is redirected to the overflow behavior.
4. If you want to change the group type, choose a new type from the **Group Type** list.
 - A Ring group creates an identity that distributes calls to all members of the ring group at the same time.



Ring Group

- A Hunt group creates an identity that distributes calls to members of the hunt group sequentially, based on availability, followed by the default overflow behavior.



Hunt Group

5. Select the up and down arrows if you want to change the order of identities in the group.
6. If you want to change the group identity overflow service, select **Select...**, and choose a new service from the Select Service box.
7. Select **OK**.

To edit an attendant identity

1. In the Identities content pane, right-click the attendant identity that you want to edit.



Attendant Identity

2. Select **Open**.
3. Type changes in any of the following boxes:
 - **Display name**
 - **Address**
 - **Number of rings**—the number of rings before the overflow behavior is initiated
4. In the Call answering section, enable or disable the **Enable** check box.
5. If you want to change the attendant identity service, select **Select**, and choose a new service from the Select Service box.
6. Select **OK**.

To assign a user identity to a new phone

1. In the Identities content pane, right-click the user identity that you want to assign to a phone.



User identity

2. Select **Assign to Phone...**
3. Enable the **Create new phone** button.
4. Type the MAC address in the **MAC Address** box.



The MAC address of each phone is printed on a sticker that is located on the phone.

5. Select a phone type from the **Phone Type** list.
6. Select **OK**.

To assign a user identity to an existing phone

1. In the Identities content pane, right-click the user identity that you want to assign to a phone.



User identity

2. Select **Assign to Phone...**
3. Enable the **Select existing phone** button.



*Only telephones that have available buttons are displayed in the existing phones list. To view all phones on the network, enable the **Show all phones** check box.*

4. Select the existing phone to which you want to assign the identity.



Make sure that the existing phone that you select has at least one available button.

5. Select **OK**.

To remove an identity

1. In the Identities content pane, right-click the identity that you want to remove.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.

To change the number of rings using the Identities pane

1. On the Identities pane, select the **Number of rings** box that corresponds to the identity that you want to change.
2. Select the number of rings that you want for the identity. The default number of rings is 4.

To edit an active service using the Identities pane

1. On the Identities pane, select the active service icon.



Active Service icon

2. Select **Edit Active Service**. The service that is active on the identity opens in the Service Editor. For information about editing services using the Service Editor, refer to [Creating Custom Call Answering Services on page 213](#).

To locate an active service using the Identities pane

1. On the Identities pane, select the active service icon.



Active Service icon

2. Select **Locate Active Service**.

The Services navigation pane opens with the active service highlighted in the Services content pane.

To change a service using the Identities pane

1. On the Identities pane, select the arrow in the services list box that corresponds to the identity that you want to change.
2. Select **Services**.
3. Select the service that you want for the identity.



For information about services and how to build and activate them, refer to [Creating Custom Call Answering Services on page 213](#).

8 Installing and Configuring SIP Telephones

This chapter presents basic SIP phone installation and configuration information for the UC server:

- [Overview on page 123](#)
- [Managing phones on page 123](#)
- [Installing SIP phones on page 128](#)
- [Configuring SIP phones on page 129](#)

Overview

The SIP solution provides extensive telephone features for the PBX end user, superior voice quality, customer cost-savings, support for real-time communications, and improved efficiency of communications.

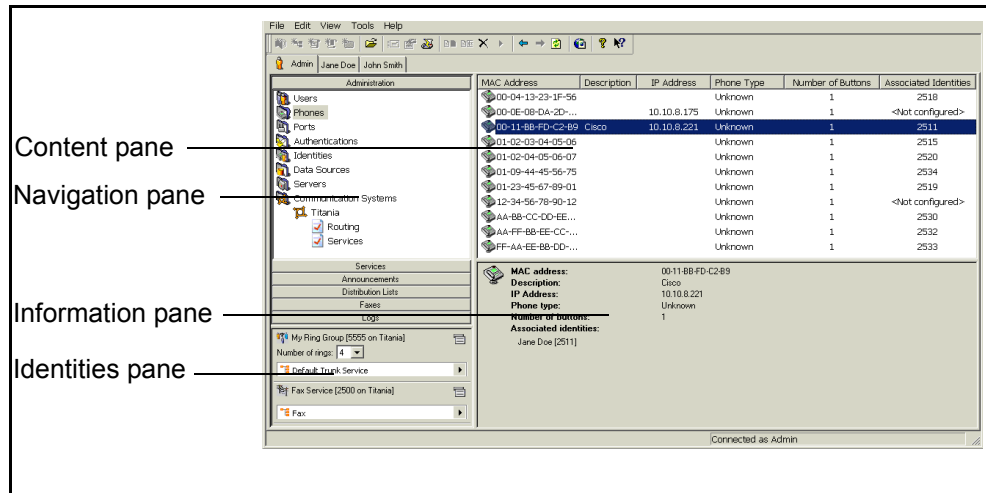
Managing phones

When you physically connect a SIP phone that supports automatic configuration to the network, the phone automatically appears in the Phones pane of the UC server Administrator Assistant. ADTRAN-supported phones, such as Polycom, must also be connected to the same LAN segment as the UC server so that they can automatically configure. The MAC address cannot be preserved if the phone is on the far side of a router.

Every SIP phone that is connected to the network appears in the Phones pane, whether the phone is configured with identities or not. SIP phones retrieve configuration information from the server.

Each SIP phone has a MAC address, which is printed on a sticker located on the telephone. If you have a barcode reader, you can scan the MAC address sticker to input the address. You also need an IP address for each phone, which is automatically assigned after you connect the phone to the system.

Using the Phones pane, you can add and assign identities, change phone information, remove a phone from the network, restart a phone, or reload phone configurations. For information about assigning a SIP phone to new a user profile using the New User Wizard, refer to [Using the New User Wizard on page 87](#).



Adding and removing phones

The phones that you add do not have to be automatically configured phones. You can add any SIP phone, but you must configure the phones manually, according to the manufacturer’s instructions. To complete the configuration, and for the phone to connect to and register with the communication system, you must supply a SIP authorization and password. The SIP authorization and password are associated with a user identity.

To get the SIP authorization and password

1. In the Administration pane, select **Identities**.
2. In the **Content** pane, right-click the user identity to which you want to assign a phone.
3. Select **Open**.
4. In the **Identity** dialog box, select the **SIP Authentication** button. Note the User/login name and Password.

To add a phone with automatic configuration

1. Unpack the telephone that you want to add.
2. Provide power to the phone.
3. Connect the LAN cable to the same subnet as the UC server. The phone automatically appears in the Phones navigation pane.



The telephone cannot make an outgoing telephone call or receive a call until it is properly configured by the UC server Administrator.

To add a phone using a barcode scanner

1. Select the IP address field in the Phones pane of the Administrator tab.
2. Scan the MAC address on the phone. The phone automatically appears in the Phones navigation pane.

To add a phone manually

1. Right-click anywhere in the Phones content pane.
2. Select **New Phone**.
3. In the Phone dialog box, type the following general information:
 - **MAC address**—find it printed on a label that is attached to the phone
 - **IP address**—leave the box blank if you do not know the IP address. When you connect a SIP phone to the network, the IP address for the phone automatically appears in the Phones content pane.
 - **Description**—type a description for the phone to distinguish it from other phones on the network. A description is optional.
 - **Phone type**—choose a phone type from the options available in the list
 - **Number of buttons**—the number of buttons is automatically displayed, depending on the phone type that you choose.
4. Select **OK**.

To remove a phone from the network

1. In the Phones pane, right-click the phone that you want to remove.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.



If the phone that you remove is still connected to the network, it might appear in the un-configured phones area.

Managing phones and identities

You can assign identities to phones, change the button number for the identities, and remove identities.



When you assign an identity to a phone, the phone immediately reboots.

To assign an identity to a new phone

1. Right-click anywhere in the Phones content pane.
2. Select **New Phone...**
3. In the Phone dialog box, type the general information for the new phone.

4. In the Assigned identities section, select the **Add...** button.
5. From the Select Identity box, select the identity that you want to assign to the new phone.
6. Select **Select**. The identity that you assign to the phone appears in the assigned identities section of the Phone box.



If the phone type has several buttons, the identities are assigned to the buttons in numerical order. You can assign as many identities as there are buttons on the phone.

7. Repeat steps 4 to 6 for all the identities that you want to assign to the new phone.
8. Select **OK**.



You can also assign an identity to a phone using the Identities navigation pane. For information about assigning an identity using the Identities pane, refer to [To assign a user identity to a new phone on page 119](#), or [To assign a user identity to an existing phone on page 120](#).

To assign an identity to an automatically configured phone

1. In the Phones pane, right-click the automatically configured phone to which you want to assign an identity.
2. Select **Open**.
3. In the Assigned identities section, select **Add**.



The Add button is only available if there are available buttons on the configured telephone.

4. Select the identity that you want to add.
5. Select **Select**. The new identity appears in the assigned identities list, assigned to the next available button.

To change the button number for identities on a new phone



Up or down arrows

1. Follow all the steps in [To assign an identity to a new phone on page 125](#).
2. Select the identity whose button number you want to change.
3. Select the up or down arrow to change the button number for the identity.
4. Select **OK**.

To delete an identity on a new phone

1. Select the identity that you want to delete.
2. Select the **trash** icon.



Trash icon

3. Select **OK**.

To change information associated with a phone

1. In the Phones content pane, right-click the phone that you want to change.
2. Select **Open**.
3. Change the information about the phone that you want.



You can change only some of the information associated with a phone, for example, IP address or description. If you want to reconfigure the phone, remove it from the network and re-install it using [To add a phone manually on page 125](#).

4. Select **OK**.

Restarting phones and reloading phone configurations

You may want to restart a phone if you are experiencing problems with the phone or the network. You may want to reload the system configuration for the phones following changes to the configuration. Some phones, such as Polycom phones, automatically restart when the phone configuration is reloaded, while other phones, such as snom phones, do not automatically restart.

To restart a phone

1. In the Phones content pane, right-click the phone that you want to restart.
2. Select **Restart Phone**. All currently selected phones restart.



This option only applies to ADTRAN Connect PLUS™ phones.

To reload a phone's configuration

1. In the Phones content pane, right-click the phone that you want to reload the configuration for.
2. Select **Reload Configuration**. The selected phones' configurations reload. Depending on the make and model, this may restart the phone.



This option only applies to ADTRAN Connect PLUS™ phones.

Installing SIP phones

The UC server allows you to quickly install SIP phones to the network. If you install a SIP phone to the same network on which the UC server and FTP server reside, an IP address is automatically and dynamically assigned to each piece of hardware on the network. The UC server also recognizes the MAC address of the SIP phone, provided it is installed on the same network.

If you are installing SIP phones on a different network, the UC server also gives you the option of entering MAC addresses manually, or using a barcode scanner. If you are installing a large number of SIP phones on the network, ADTRAN recommends that you use a barcode scanner.



If you install a SIP phone on a network on which the UC server and FTP server do not reside, the telephone does not retrieve the correct IP and MAC address.

The procedures below are for general setup of most SIP-enabled telephones.



External problems that affect the IP connection, or the availability of a SIP connection from your service provider, might interfere with IP trunk availability. Also, local power outages can temporarily interfere with SIP phone operation.

You can reduce loss of phone service in the following ways:

- Use a PSTN gateway to provide backup phone service.
- Use an Un-interruptible Power Supply (UPS) to maintain SIP phone operation during short local power outages.
- Install an analog phone on an analog trunk for basic and emergency phone service in the case of an extended power outage.

When you plug in a telephone that has not been configured, automatic configuration requires a DHCP server. From the DHCP server, the telephone can retrieve the following information:

- IP address
- network information

- boot server (FTP or TFTP server)

When you install a telephone with automatic configuration, it searches for a boot server. If a boot server is available, the phone attempts to retrieve, from the DHCP server, network and configuration information, and its software version. If the phone cannot retrieve configuration information, it downloads a default configuration and applies the default settings.



All SIP devices have a registration setting that defines the SIP server, SIP port, user name, and password. The default configuration includes the IP address of the UC server and a fake SIP registration port.

The attempted SIP registration creates a new phone entry in the UC server phone management window.

Configuring SIP phones

For telephones that are not automatically configured, you can view the SIP authentication or password.



For information on configuring the message waiting indicator, refer to [Configuring a user's message waiting indicator on page 99](#). For information on setting the order messages are played over the telephone, refer to [Setting the message playback order on page 23](#).

To view the SIP authentication

1. If all identities are not displayed, select **View > Display Identities for all Profiles...**
2. On the Identities pane of the Administrator application, select the Active Service Icon for the identity for which you want to view SIP authentication.



Active Service icon

3. Select **SIP Authentication**.

9 Configuring the NetVanta UC Server

The UC server is typically configured at the time of installation, using the Server Configuration Wizard. However, further configuration may be required when hardware is replaced, or when the network or system is upgraded. This chapter presents information on configuring the UC server and includes the following topics:

- [Configuring Gateways on page 131](#)
- [Configuring Ports on page 135](#)
- [Configuring Servers on page 137](#)
- [Configuring Communication Systems on page 147](#)
- [Configuring Licenses on page 152](#)

You can configure gateways, ports, servers, and communication systems from the Administration navigation bar.

Configuring Gateways

The **gateways** section allows you to define a gateway, including its host name (or IP address), device type, and description with each gateway on the system. Enter information for each of the gateways that are configured on the communication system server.

A gateway allows callers to phone people that are external to the UC server, through PSTN gateways, Internet Telephony Service Providers (ITSPs), and SIP Peers. Gateways are available from many different vendors.



Make sure to consult ADTRAN integration notes and manufacturers' documentation for each gateway on the system.

Several types of gateways can be used with the UC server. During the installation and configuration of the UC server, the Server Configuration Wizard provides the following gateway options:

- ITSP Gateways
- PSTN Gateways
- SIP Peers
- Gateway Groups
- None



For ITSP gateways, select PSTN gateway.

More than one gateway type can be added, but each gateway is configured separately.

To complete the gateway programming, an identity must be created for incoming calls to be routed through the gateway. Refer to *Managing User Extensions and Identities on page 109* for more information. The routing or dial plan must also be completed to direct outgoing calls through the gateway. Refer to the *Configuring Communication Systems on page 147* section for more information.

ITSP Gateways

The UC server provides support for ITSPs. ITSPs offer Voice over Internet Protocol (VoIP) services using broadband connections. VoIP is a combination of hardware and software that enable users to make and receive telephone calls over the Internet. The voice data is transmitted in packets using Internet Protocols (IP) instead of using the traditional PSTN. ITSPs terminate a PSTN telephone number and associate it with a SIP identify.

Benefits of ITSPs

- Using an ITSP provides a number of benefits, primarily with regards to cost savings.
- Lower PSTN service costs
- Lower long distance costs
- Calls can be sent and received through a VoIP phone, regardless of your location, provided you have an Internet connection.
- A single provider can provide local telephone numbers in different geographic locations.
- Disaster Recovery options as ITSP services can be connected from different locations.

NetVanta UC Server Support for ITSPs

The UC server supports ITSP gateways, using a SIP-aware firewall that contains an Application Layer Gateway (ALG), such as the Ingate SIParator. The SIP-aware firewall allows the network address translation (NAT) traversal of SIP trunking calls to various service providers from the UC server. The UC server also supports dual network interface controllers (NICs) on servers.



To use an ITSP gateway, select the Public Switched Telephone Network (PSTN) as the gateway type.

For more information on Ingate SIP trunking, refer to the *NetVanta ITSP Configuration* technical note. This document is available on your installation CD, and from the ADTRAN website at <http://kb.adtran.com>.

PSTN Gateways

The UC server can provide public switched telephone network (PSTN) access through SIP PSTN gateways. SIP PSTN gateways convert the SIP protocol (signaling and voice) to traditional PSTN (Time Division Multiplexing) technology. SIP PSTN gateways can use either analog or digital lines (T1 in North America).

SIP Peers

The SIP Peer allows for the networking of multiple UC servers. The default port is 12555.

Gateway Groups

If you have more than one gateway, you can organize them into groups. Dial plan entries route calls to alternate trunking facilities if the facilities are busy or unavailable. When a call matches the dial plan entry, the call is routed to the first member of the gateway group. If the first member is unavailable, the next member of the group is used. When a call is forwarded to another gateway is determined by the dial plan entry. That is, in a dial plan entry the field Call next member after X seconds, determines how long to wait when a gateway is busy, there is no answer, or an error, before trying the next gateway in the gateway group.

A gateway group might be used for example, if two trunking facilities are available, PRI and analog. The analog facilities can be used as a second choice when the PRI lines are busy or unavailable. A flexible dial plan routes incoming and outgoing calls to individual or groups of gateways. A gateway group provides the following capabilities:

- Increased capacity
- Fault tolerance/Disaster Recovery - This provides overflow capability in the event that a gateway device is busy or unavailable.
- Least Cost Routing (LCR) alternatives - A gateway group can contain gateways that are connected to different carriers, such as local and long distance providers, or can contain gateways in different geographic locations.
- Different technologies of trunking facilities can be combined seamlessly. Gateway groups can combine analog facilities, digital facilities, SIP trunking services, and SIP Peers.

Configuring Gateways

Gateways are initially configured when the UC server is installed, using the Server Configuration Wizard, Add Gateways Wizard. However, gateway information can also be modified using the UC client. Changes to gateways do not require a system or service restart in order for the changes to take effect.

Deleting a gateway removes the gateway host information from the SIP communication system. Deleting a gateway that is part of a gateway group removes the gateway from the group. Deleting the last member of a gateway group removes the gateway group. Deleting a gateway removes dial plan entries if it is the only gateway using that entry, or if it is the last gateway in a group that is the only gateway configured for the dial plan entry.

To add a gateway

1. From the Administration Navigation pane, select **Gateways**.
2. Right-click in the Content pane, and select **New Gateway**.
3. Select the gateway type to be used in your environment.
Follow the instructions for your gateway type.

To configure PSTN gateways

Enter the host name for the PSTN gateway.

To configure SIP Peers

1. Enter the host name for the SIP Peer. This is the current IP address or DNS name of the host.
2. Enter the port number. This has a default value of **12555**, which is suitable for most networks.

To create a gateway group

1. From the Available gateways list, select a gateway and select **Add** to add the gateway to the gateway group.
2. Repeat the step above until you have added all the gateways you want to your gateway group.
3. Select **Next** when you have finished creating your gateway group.

To configure the gateway

1. Enter a unique name for the gateway. This information is for display purposes only, on the UC server and in the database.
2. Select the communication system for which the gateway is associated. The default is the UC server.
3. Enter a brief description of the gateway. This information is for display purposes only.

To Edit a Gateway

1. From the Administration Navigation pane, select **Gateways**.
2. From the Content pane, double-click on the gateway you want to edit.
3. From the Edit Gateway dialog, General tab, the following gateway information can be edited, depending on the gateway provider:
 - Name - Enter a gateway name, used for display purposes only in the UC server.
 - Description - Enter a brief description of the gateway, for information only.
 - Host - Enter the gateway host name, if applicable.

To Delete a Gateway



Deleting a gateway removes the dial plan entries if it is the only gateway configured for the dial plan entry, or if it is the last gateway in a gateway group that is the only gateway configured for the dial plan entry.

1. From the Administration Navigation pane, select **Gateways**.
2. Right-click on the gateway you want to delete, and select **Delete**.



When you delete any communication system configuration category entry, you are not prompted to confirm the deletion. Make sure that you want to delete the entry before doing so.

Configuring Ports

The UC server ports provide a real-time view of the system activity. A port represents an active call into the UC server, and allows you handle incoming calls and personalized communication services.

You can view the following types of port activity:

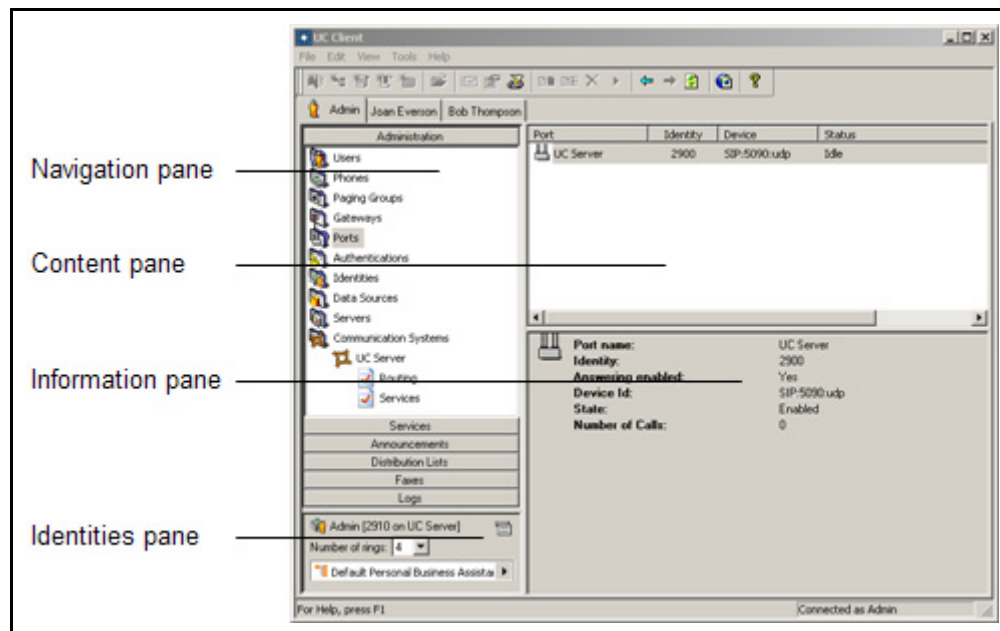
- Incoming calls
 - voice and fax
 - includes calling line ID, if available
- Outgoing calls
 - handset playback
 - port auditing
- Outgoing faxes

A port represents a physical resource and associated hardware in the attached communication system.

You can add and manage ports only if a communication system already exists. For information about managing communication systems, refer to [Configuring Communication Systems on page 147](#).

The UC server can connect to multiple communication systems at the same time, provided the integration hardware and software components support it. For example, the UC server can simultaneously support the integrated SIP server and Avaya IP Office.

To manage ports, in the navigation pane, select the Ports folder. Port names are displayed in the content pane.



Configuring ports for all types of communication system

You can configure the UC server ports. Ports are the physical connections on the telephony card that connect the UC server to the communication system.



If you remove a port that is responsible for activating or deactivating message waiting indicators, you must first delete the message waiting indicators status from the attached communication system.

If you integrate additional communication systems, the hardware and software components must be installed before you can configure ports for the communication system. If your system uses a traditional communication system, you must configure a port for each configured communication system port. If your system has an integrated SIP communication system, you require only one port to handle multiple simultaneous calls.

To add a new port so that the NetVanta UC Server answers it

1. Right-click anywhere in the content pane.
2. Select **New** to open the Port dialog box.
3. In the **Name** box, type a name for the new port.
4. In the **Extension** box, type the extension number associated with the new port.
5. Enable the **Enable use of port for** check box if you want the UC server to answer calls on that port.
6. Enable any of the following check boxes to configure port management:
 - **Handset call**
 - **Pager notification**
 - **Message delivery**
 - **message waiting indicator changes**
 - **Fax transmission**
 - **Port audit**
7. Select a communication system from the **Pbx** list.
8. Select a device from the **Device** list. Devices are channels on the Dialogic telephony card.



Information about the port device that you select appears in the Information section.

9. If you select a SIP port, type a port number in the **Port number** box, and choose a protocol from the **Protocol** list.
10. Select **OK**.

To edit a port

1. Right-click the port that you want to edit.
2. Select **Open** to open a Port dialog box that has the name of the port.

3. If you want to rename a port, type a new name in the **Name** box.



You cannot change the extension number of an existing port. To change an extension number, you must remove and then re-create the port.

4. If you want to disable the port, disable the **Enable use of port for** check box.
5. If you want to enable the port, enable any of the following check boxes to edit the configuration of port management:
 - **Handset call**
 - **Pager notification**
 - **Message delivery**
 - **message waiting indicator changes**
 - **Port audit**
 - **Fax transmission**
6. If you want to change the device, select a new device from the **Device** list. Devices are channels on the Dialogic card.
7. Select **OK**.

To delete a port so that the application no longer answers it

1. Right-click the port that you want to delete.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.



It is not advisable to delete or change the primary port on the Norstar communication system. If you do so, all identities trunks and extensions must be re-configured so that incoming calls are directed to the new primary port. This is a time-consuming process during which calls might not be answered.

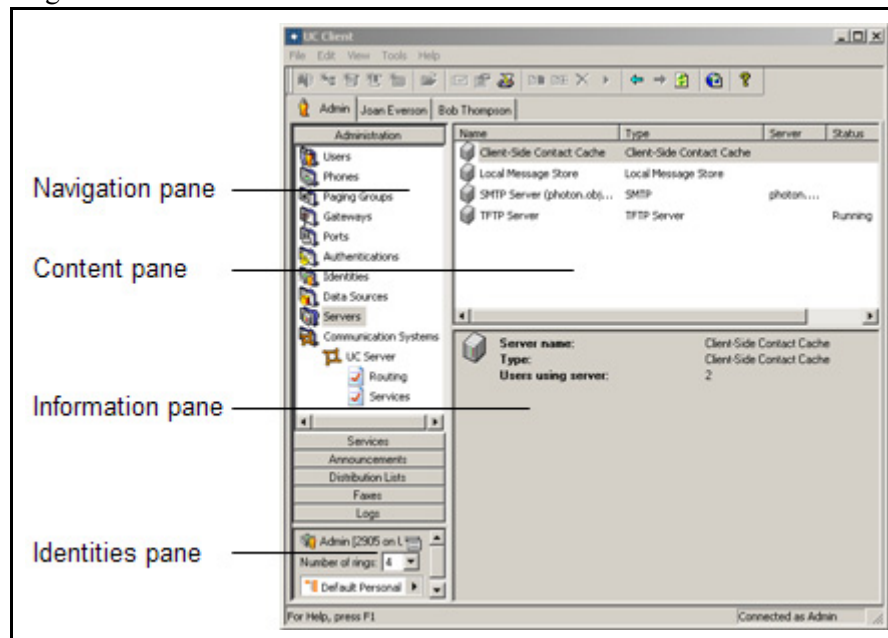
Configuring Servers

In the Administration navigation pane, you can use the Servers pane to add servers. You can add three types of server: Microsoft Exchange Server, IMAP4, and SMTP. Lotus Notes and Lotus Domino servers use the IMAP4 integration.

Adding a server lets you configure a variety of functions. For example, adding an Exchange Server allows you to store and retrieve voice, fax, and email messages, configure the Send email function, and integrate your contact list with UC server clients.

For non-Exchange Server set-ups, you can use an SMTP server to send email notification. The IMAP server configuration is preferred for use with Lotus Notes.

You can also change server information and delete a server.



For information on setting mail server message buffering options, refer to [Setting mail server message buffering on page 39](#).

Configuring Exchange Server

When configuring Microsoft Exchange Server, there are several options available to consider. The following sections describe these options.

Configuring message delivery

When you configure Exchange Server, you can set the message delivery mode system-wide to determine whether voice and fax messages are sent from the system mailbox or are stored directly in users' mailboxes. This setting affects whether:

- User defined message rules in Microsoft Outlook® or Microsoft® Exchange Server are applied
- Messages are sent to third-party applications such as BlackBerry® Enterprise Server
- The caller's phone number appears in the From line of the message.



You can also set the message delivery mode for individual users. For more information, refer to [Configuring a user's message storage on page 96](#).

Send messages from system mailbox

Because voice and fax messages are sent from the system mailbox:

- Any user defined message rules set in Outlook or Exchange Server are applied. For example, a user's rule might direct messages from a particular contact to a specific folder.
- Messages are sent to any third-party applications such as BlackBerry Enterprise Server.
- The message appears to be from the system mailbox, and the From line does not include the caller's phone number.

Impersonate caller and store to mailbox

Because voice and fax messages are stored directly in the user's mailbox, rather than being sent:

- User defined message rules in Outlook or Exchange Server are not applied. For example, a user's rule directing messages from a particular contact to a specific folder is not applied.
- Messages are not sent to any third-party applications such as BlackBerry Enterprise Server.
- The From line includes the caller's phone number, or the caller's name from a contact that matches the phone number.

Synchronizing message waiting indicators for messages accessed through the web or mobile devices

You can configure Exchange Server to monitor users' mailboxes to determine when a new message arrives or is read, or when a message is marked as read or unread. The advantage is that when a user accesses their voicemail messages through Microsoft Outlook Web Access or through a mobile device, the UC server adjusts the message waiting indicator accordingly on the user's phone, to turn it on or off. Without mailbox monitoring, the UC server is unaware when a user accesses their voicemail messages through Outlook Web access or through a mobile device. For example, even after a user has accessed all their new messages, the message waiting indicator on their phone may continue to flash.

Mailbox monitoring can however, negatively impact system performance. Therefore, you may want to enable mailbox monitoring only for those users who require it.

**NOTE**

Mailbox monitoring works in conjunction with the message waiting indicator options set. For example, the message waiting indicator options determine if the message waiting indicator is turned off when the first message is read, or when all new messages have been read. For more information, refer to [Configuring message waiting indicator options on page 21](#). For information on troubleshooting message waiting indicators, refer to [Troubleshooting Message Waiting Indicators on page 317](#).

**NOTE**

If mailbox monitoring is enabled, then you can also configure mailbox monitoring for individual users. For more information, refer to [Configuring a user's message storage on page 96](#).

Enable mailbox monitoring

Opens the Exchange Server mailboxes of selected users at system start up. The UC server monitors each mailbox for changes such as a new message has arrived or is read, or a message is marked as read or unread. This applies only to the Inbox, and other folders are not monitored. When a user accesses messages through Outlook Web Access or through a mobile device, depending on the message waiting indicator (MWI) settings selected, the MWI turns on or off accordingly.

Default setting for users

If mailbox monitoring is enabled, then this mode turns mailbox monitoring for users on or off by default.



Mailbox monitoring could negatively impact system performance. Mailbox monitoring increases the traffic between the Exchange Server and the UC server. Depending on the number of users configured for mailbox monitoring, the bandwidth between the Exchange Server and UC server, and the CPU power, performance may be affected. On systems with more than 100 users, system design and engineering must consider the increased load that keeping open large numbers of mailboxes will have on the Exchange Server. Enabling monitoring of more than 500 mailboxes requires careful consideration and probably should not be done without a slow roll-out procedure to validate the performance of both Exchange Server and UC server in the specific customer environment. If your Exchange Server is at or near capacity, then you may want to disable mailbox monitoring.

To add an Exchange Server

1. In the Administration navigation pane, select **Servers**.
2. Right-click in the Servers content pane, and select **New > Exchange** to open the Exchange Server dialog box.



You can add only one Exchange Server. If you have already added an Exchange Server, the option is unavailable.

3. Type a name for the server in the **Name** box. This name is used as a label to distinguish between different message servers, and appears in the list of server names.
4. Type a machine and mailbox name in the boxes.

Machine name The Machine name corresponds to the actual name of the Exchange Server that the UC server is connected to.

Mailbox name The Mailbox name is the Exchange Server mailbox name that has special access privileges in Exchange Server.



If you change a mailbox name, the corresponding Exchange Server account might also need to be configured.

5. To select the mailbox from which you want email notification sent, enable one of the following:

System mailbox	The email notification appears to come from the configured UC server Exchange Server account.
User mailbox	The email notification appears to come from the user's mailbox.



The sender information for Exchange Server applies only if the Exchange Server is configured to send email notification.

6. In the Message delivery section, select one of the following options:

Send messages from system mailbox	Voice and fax messages are sent from the system mailbox.
Impersonate caller and store to mailbox	Voice and fax messages are stored directly in the user's mailbox, rather than being sent.

7. In the Mailbox monitoring section, check Enable mailbox monitoring if you want to enable mailbox monitoring for any or all users.

Enable mailbox monitoring	Opens the Exchange Server mailboxes of selected users at system start up. The UC server monitors each mailbox for changes in new message status, and depending on the MWI settings, adjusts the MWI accordingly.
Default setting for users	If mailbox monitoring is enabled, then this mode turns mailbox monitoring for users on or off by default.



*The Information pane shows the connection state of the Exchange Server as either Connected or Disconnected. The connection state is Disconnected if the Server account has no mailbox, or if the Server account lacks permissions to access the Exchange Server. For instructions about configuring a mailbox and permissions, refer to **Appendix A** of the NetVanta UC Server Configuration Guide.*

Configuring IMAP Server

IMAP (Internet Message Access Protocol) is a method of accessing and storing email messages. The UC server can integrate with IMAP version 4rev1 servers conforming to RFC 2060. For more information on IMAP version 4rev1, refer to www.ietf.org/rfc/rfc2060.txt. Add an IMAP server when you need to integrate the server with Lotus Notes, Lotus Domino, Novell GroupWise, Google Gmail, or other servers.

The UC server can use mailboxes on an IMAP server to store voice and fax messages in order to provide users with unified messaging functionality. In addition, users licensed for text-to-speech (TTS) can listen to their email messages from their mailbox over the telephone.

IMAP servers vary in the methods they support for retrieving a user's voice and fax messages, and these

methods vary in the speed at which they retrieve such messages.



To optimize IMAP server performance and to minimize delay between when a user logs into the telephone user interface (TUI) to check their mailbox, and when the UC server can play mailbox and message information, such as the number of new messages, ensure that you select the fastest search method available that your IMAP server supports. For more information, refer to [Determining a Method of Filtering Messages for IMAP Servers](#) on page 143.



You can add multiple IMAP servers. This can be used to connect to different servers, or to create different servers with different configuration options.

To add an IMAP server

1. Right-click anywhere in the Servers content pane.
2. Select **New > IMAP** to open the IMAP4 Configuration dialog box.
3. Type a server name in the **Name** box. The server name is used to distinguish the IMAP server from other server types. This is a descriptive name and cannot be changed after creating the server. Ensure that you use a distinct and meaningful name.
4. Type a host name in the **Host name** box. The host name or IP address must be the host name or IP address of the IMAP server that the mailboxes are on. In some cases, such as Gmail, this may point to a proxy instead.
5. Type a port number in the **Port number** box. The port number is the Internet protocol port number of the IMAP service on the target server. The default port number for IMAP servers is 143.
6. If you want the messages that users delete to be permanently deleted from the server, enable the **Permanently delete messages flagged for deletion** check box. Some IMAP servers may require this to be set in order for messages to actually be removed from menu in the user's email client. By default, when messages are deleted from an IMAP server, they are only marked for deletion. Enabling this mode causes deleted messages to be removed from the mailbox after a message is marked for deletion by the UC server. With some IMAP servers, user's may still see messages in their mailbox after deleting them using the telephone user interface (TUI). If this is the case, then enabling this mode should resolve this issue.
7. If you want to disable message waiting indicators, enable the **Disable message waiting indicators** check box. Disable the message waiting indicators if you do not have a compatible mail client installed with the UC server voice forms. If you are using Lotus Notes or Lotus Domino then enable message waiting indicators if client integration is installed, otherwise, if you are not using UC server client forms then disable the message waiting indicators. In all other cases this should be disabled.
8. For Query optimization technique select the fastest option available for retrieving messages that your IMAP server will support.

Search custom message headers (requires SEARCH HEADER; fastest)

This is the default value. Select this mode if you are using *Lotus Notes* with Indexing turned On.

Search subject and then check message headers (requires SEARCH SUBJECT)

Select this mode if you are using *Gmail* or *GroupWise*.

No query optimization
(use if SEARCH not supported; slowest)

Select this method *only* if your IMAP server does not support the other methods available.
Select this mode if you are using *Lotus Notes* with Indexing turned Off.
Note: If this method is selected then users must be informed of the importance of not storing large numbers of messages in their inbox as this delays message retrieval.



If you are using Gmail, GroupWise, or Lotus Notes as your IMAP server, then a supported method of retrieving messages has already been determined. Select the appropriate query optimization technique noted for your server configuration. If you are not using Gmail, GroupWise, or Lotus Notes as your IMAP server, then proceed to the following section [Determining a Method of Filtering Messages for IMAP Servers on page 143](#) to determine the optimal settings.



If you select a search method that is not supported for your IMAP server, then the UC server first tries the selected method. If the selected search method returns an IMAP protocol error to the UC server, then the UC server automatically falls back to the No query optimization method and tries again. While this method will function with all server configurations, IMAP server performance may be affected and message retrieval may be delayed as the selected method is attempted and fails, and then the slowest method is implemented. Select the fastest method that is supported by your server configuration to minimize message retrieval delays.

9. Select **OK**.

Determining a Method of Filtering Messages for IMAP Servers



If you are using Gmail, GroupWise, or Lotus Notes as your IMAP server, then a supported method of retrieving messages has already been determined and you do not need to follow the steps below. If you are not using Gmail, GroupWise, or Lotus Notes as your IMAP server, then follow the steps below to determine the optimal settings.

IMAP servers vary in the methods they support for retrieving a user's voice and fax messages, and the speed at which they retrieve such messages. The procedures below involve selecting the fastest retrieval method first and testing the results to determine whether your IMAP server supports the method selected. If this method is unsuccessful then select the second fastest method and test the results again. If this method is unsuccessful then select the third method.

If an unsupported method is selected, then the UC server automatically falls back to the No query optimization, the slowest method, and tries again. For this reason the testing procedures described must be

followed to determine the fastest supported method of filtering messages.



To optimize IMAP server performance and to minimize delay between when a user logs into their telephone user interface (TUI) to check their messages and when the UC server can play mailbox and message information, ensure that you select the fastest message retrieval method available that your IMAP server supports.

To determine the optimal method of filtering messages

The steps below outline the general process used to determine the optimal method of filtering messages, as detailed in the following sections:

1. Follow the steps below *To configure IMAP for searching and retrieving messages* to select a method of filtering messages, starting with the fastest method first.
2. Follow the steps below *To test the selected method of filtering messages* to test the method selected for delay in message retrieval.
3. Follow the steps below *To check the software logs for message filtering fall back messages*.
4. Depending on the outcome of steps 2 and 3 above, follow the relevant procedures below:
 - If delay was not detected in step 2, and no message filtering fall back messages were found in step 3, then no further configuration is required.
 - If delay was detected in step 2 or the software logs included message filtering fall back messages in step 3, then the steps above are repeated, starting at step 1, and selecting the second fastest method of filtering messages.
 - If the second fastest method of filtering messages was selected in step 1, and delay was detected in step 2 or the software logs included message filtering fall back messages in step 3, then repeat step 1 only and select the No query optimization method.

To configure IMAP4 for searching and retrieving messages

1. From the Administration navigation bar, select **Servers**.
2. Right-click on the **IMAP Server** and select **Open**.
3. Select a *Query optimization technique* from one of the options listed below, starting with the fastest to the slowest method in the order presented below.

Trial order	Query optimization technique	Description
1	Search custom message headers (requires SEARCH HEADER; fastest)	This is the default value. The UC server adds custom headers to voice and fax messages so that they can be identified and retrieved.
2	Search subject and then check message headers (requires SEARCH SUBJECT)	This method first searches the Subject header for voice and fax messages. Of those messages retrieved, the message headers are then checked for the custom voice and fax message headers.

Trial order	Query optimization technique	Description
3	No query optimization (use if SEARCH not supported; slowest)	<p>This method does not use a query so all messages have information retrieved from the IMAP server, and all messages are then searched to retrieve the voice and fax messages. All IMAP servers support this method, but use this method only if SEARCH is not supported.</p> <p>Note: This method will be notably slower if there are a large number of messages in a user's inbox as all messages have header information retrieved and searched. If this method is selected then users must be informed of the importance of not storing large numbers of messages in their inbox.</p>

4. Select OK.



If you select a method that is not supported by your IMAP server, then when the SEARCH operation returns an IMAP protocol error to the UC server, the UC server automatically falls back to the No query optimization method and tries again. While the No query optimization method functions with all IMAP servers, it is the slowest method, and it will be slowed further if another method is selected and then fails before the No query optimization method is attempted. To maintain optimum IMAP server performance and to minimize message retrieval delay, it is important that you select the fastest retrieval method possible that is supported on your system, and only select the No query optimization method if neither of the other options are successful with your IMAP server.

To test the selected method of filtering messages

1. Leave a voice message for a user configured for IMAP.
2. Login to the user's voicemail using the TUI.
3. Check the user's voice messages to determine if the voice message left in step 2 is there. If there is a time delay in the message being available over the TUI (i.e., checking immediately after leaving the message fails to present the message, but checking 20 minutes later works), then this may be an indication that the selected method of filtering messages is not supported by your server.
4. Play the voice message left in step 2.

To check the software logs for message filtering fall back messages

If a message retrieval method was selected but failed (it is important to identify when the UC server falls back to the No query optimization method), then this will be noted in the software logs.

1. Navigate to the software logs. The default location is C:\OWLogs. The Software Logging Configuration Tool can be used to check the location. The most recent log is swlog.txt, and log files will rotate as they get larger.
2. Look for the following messages after the voice message was left in step 2 of the previous set of instructions.

"IMAP optimization technique may be set wrong (SEARCH HEADERS), switching to 'brute force' technique." This indicates that the IMAP server does not support the selected message retrieval method, and has switched to the No query optimization method.

"IMAP optimization technique may be set wrong (SEARCH CUSTOM), switching to 'brute force' technique." This indicates that the IMAP server does not support the selected message retrieval method, and has switched to the No query optimization method.

"Failed to brief summary information about message." This indicates that the IMAP server does not support the selected message retrieval method.

"Not connected, or unable to reconnect to server." This indicates that the server cannot be reached.

3. If you did not find one of these messages in the software logs then no further steps are required.
4. If you find one of the messages noted above in the software logs, then repeat the configuration steps above and select the next fastest method. Repeat the testing procedures (starting at *To configure IMAP4 for searching and retrieving messages*) to determine if the method selected is supported by your server. If another one of these messages is found in the software logs, then select the No query optimization method.

Configuring SMTP Server

To add an SMTP server

1. Right-click anywhere in the Servers content pane.
2. Select **New > SMTP** to open the SMTP Server Configuration dialog box.
3. Type a server name in the **Server name** box.
4. Type a name for the SMTP server in the **SMTP server name** box. For example, mail.yourcompany.com.
5. Type a port number in the **Port number** box. The default port number is 25.
6. Type the **Sender email display name**.
7. Type the **Sender email address**.
8. Select **OK**.

Configuring email notifications

To configure the Send email function

1. Select **Tools > Options**.
2. Select the **Message Delivery** tab.
3. From the **Send email using** list, select the server that you want to use.
4. Select **OK**.

Changing server information

To change server information

1. In the Server content pane, right-click the name of the server whose information you want to change.
2. Select **Open** to open the Server Configuration dialog box.



The type of information that is available for you to change varies depending on the type of server. You cannot change the information associated with the Client-side Contact server.

3. Change the information that you want.
4. Select **OK**.

Deleting servers

To delete a server

1. In the Server content pane, right-click the name of the server that you want to delete.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.

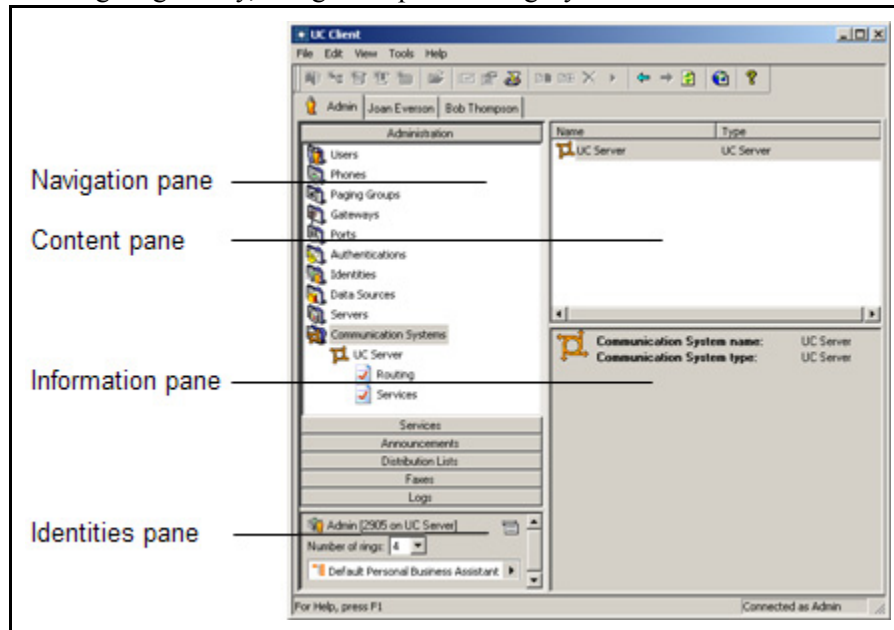


You cannot delete an Exchange Server if there are users assigned to it. You cannot delete the LMS (Local Message Store) server.

Configuring Communication Systems

You can use the Communication Systems menu to add a communication system, change information for an existing communication system, or delete a communication system. To manage communication systems, select the Communication Systems menu in the Administration navigation pane. Configure a UC server communication system using the Routing and Services categories. Configure an existing PBX connected

to the UC server through a gateway, using the Options category.



Communication systems types

The Communication Systems options allow the installer to integrate the UC server with the communication system depending on the hardware installed on the UC server. Below are some of the communication system types that are possible:

- Avaya Definity and Multivantage
- IP Office
- Avaya Merlin Magix and Legend
- Cisco CallManager
- Generic Analog
- Generic PBX (via Gateway)

A variety of PBXs can be connected to the UC server through a Dialogic Media Gateway.



*For Generic PBX (via Gateway) gateway configuration information, refer to technical note **Dialogic Media Gateway Configuration Guide**, on the ADTRAN website at <http://kb.adtran.com>. For a list of compatible PBXs, and PBX configuration information, refer to the Dialogic website at www.dialogic.com.*

Avaya Definity and Multivantage

The advanced configuration allows you to program the feature codes to correspond to what is currently configured on the communication system. Configuring message waiting indicator activation and de-activation codes makes sure that the communication system turns on or off the message waiting indicator.

The message indicator activation code activates the message waiting indicator for user extensions. The

message waiting indicator is lit when there are new messages in a user's mailbox.

The message indicator de-activation code de-activates the message waiting indicator for user extensions after they listen to their messages.

IP Office

The UC server can function as a centralized voicemail service for a main office and one or more remote IP Office sites using a remote agent application.

When calls at a remote site are unanswered and are redirected to the remote site voicemail hunt group, the remote agent first sends the call information to the UC server and then redirects the call to the main site IP Office voicemail hunt group. The UC server uses the information sent by the remote agent in conjunction with the call information supplied by IP Office at the main site to execute the appropriate service.

Requirements at the Remote IP Office Site

- Small Community Networking enabled
- Exactly one remote agent application must be installed at each of the remote IP Office sites, and the firewall configured accordingly.
- Avaya's TAPI driver must be installed on each machine that hosts a remote agent.
- A valid CTI Link Pro license for each IP Office.
- All extensions, hunt group numbers, and display names must be unique across all IP Office sites.
- Each remote IP Office site must have a voicemail hunt group where unanswered calls are redirected. The extensions in this hunt group are monitored by the remote agent application and must not be in use. The extensions can be one of the following:
 - Analog
 - TAPI/Wave - requires a license for the remote IP Office
 - Digital - requires a physical digital phone plugged into the corresponding digital port, but must not be used by anyone
- Each remote IP Office site must have a short-code defined that allows for forceful redirects over the IP trunk connecting the remote and main site IP Offices.

Requirements at the Main IP Office Site

- Small Community Network enabled
- The UC server must be configured for all of the remote agents and the extensions they are monitoring.

For more information on configuring IP Office remote agents, refer to the ADTRAN website at <http://kb.adtran.com> and consult the *IP Office Integration Guide*.

Avaya Merlin Magix and Legend

The Advanced communication system options allow you to configure a CTI link. The CTI link enables the

UC server to obtain external calling line ID.



You must subscribe to Calling Line ID from your local telephone company to use a CTI link.

Consult the manufacturer's documentation about how to install and configure the CTI link for your particular communication system.

Generic PBX (via Gateway)

If you are connecting the UC server to an existing PBX through a gateway, then there are a few options to configure. You must enter an IP address or a Fully Qualified Domain Name (FQDN), and a port number, to configure the gateway.

If your attached PBX does not have true answer supervision, then enable the prompt for handset playback. Without true answer supervision, then the beginning of playback and recording using the handset may get cut off. This affects message playback, announcement playback and recording, and greeting playback and recording, using the handset.

To configure PBX options for a PBX connected through a gateway

1. In the Communication Systems content pane, right-click the **Generic PBX (via Gateway)** that you want to change, and select **Open**.
2. In the Options dialog, enter the IP address or the FQDN.
3. Enter the appropriate port number, **5060** for a SIP PBX.
4. Enable the **Prompt for handset playback** mode if your system does not have true answer supervision. If the beginning of playback and recording using the handset is being cut off, then this indicates that your attached PBX does not have true answer supervision, and you should enable this check box.
5. Select **Change**.

To add a new communication system

1. Right-click anywhere in the Communication Systems content pane.
2. Select **New** to start the Communication System Wizard.
3. Type a name for the new communication system in the **Pbx name** box.
4. Select the type of communication system from the **Pbx type** list.
5. Select **OK**.

To change communication system information

1. In the Communication System content pane, right-click the communication system that you want to change.
2. Select **Open**.
3. Change the information about the communication system that you want.

4. Select **OK**.

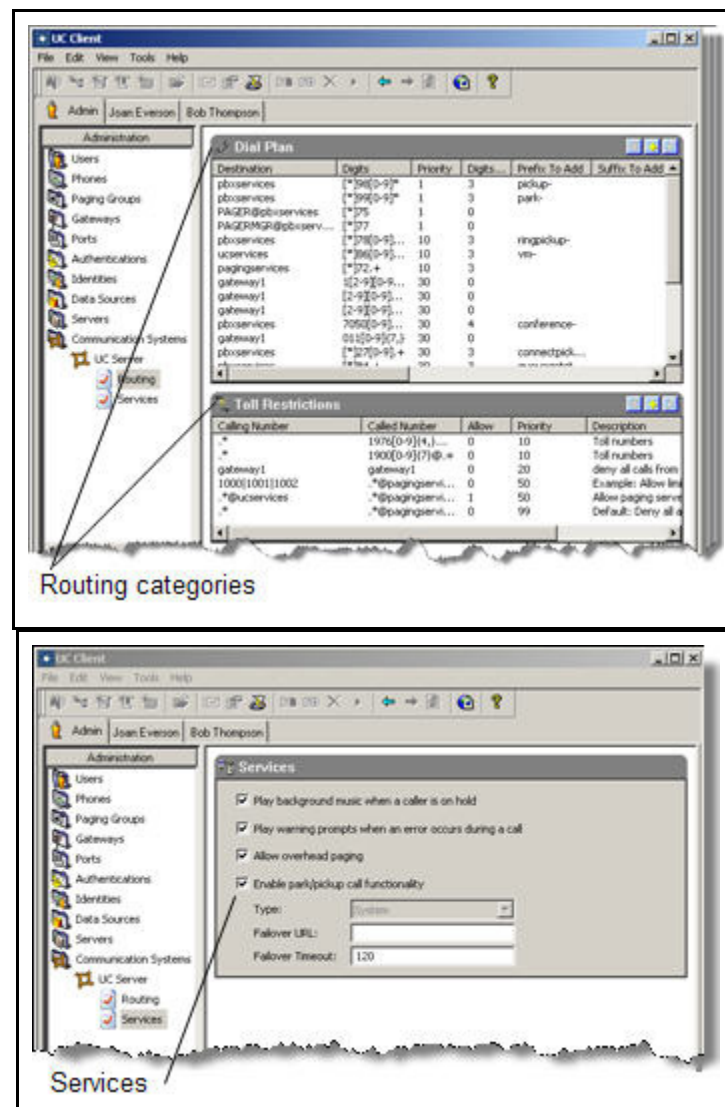
To delete a communication system

1. In the Communication Systems content pane, right-click the communication system that you want to delete.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.
4. Select **OK**.

Configuring NetVanta UC Server communication systems

For each UC server communication system on the system, you can configure the following:

- Routing** Including:
- [Routing calls using the dial plan on page 43](#)
 - [Allowing local and toll free calls on page 47](#)
 - [Restricting long distance calls on page 48](#)
- Services** [Configuring system call options on page 40](#)



Configuring Licenses

The licensing section provides information on how to manage the license keys on the UC server. ADTRAN has defined the UC server license system using the following concepts and terminology:

- **Purchase Key** — The purchase key (product key) is a unique system identifier that was assigned to you when the UC server system was ordered. The purchase key includes the core software and the bundled feature contents according to the original product purchase.

The purchase key can be identified as a unique key that XXXXX-XXXXX-XXXXX where X can be any combinations of numbers or capital letters.

- **Upgrade keys** — An upgrade key represents an upgrade to an existing product and is used to generate a license key. Upgrade keys can be purchased to increment components of UC server independently of

the core software.

The UC server software does not keep track of upgrade keys, but they are maintained and used by ADTRAN to generate the correct license key. In addition to product capabilities, upgrade keys are also used to extend software assurance. An upgrade key can be identified with the following format XXXXX-XXXXX-XXXXX-U where X can be any combination of numbers or capital letters.

- **Machine Identifier** — A unique signature that is automatically generated based on the computer platform installed hardware.
- **License Key** — An encoded key generated by ADTRAN that includes product attributes that will only work with systems that have a specific purchase key and associated machine identifier.

How to make a request for a new license key

1. A license key request can be generated by using one of the following methods:

- Authorized resellers or customers who maintain their own system may make requests electronically by using their ADTRAN website account. Once successfully logged into the ADTRAN website, a request can be made by navigating to **My Account > License Key Request**

OR

- Obtain the License Key Request form available from the product documentation section of the ADTRAN support center. Complete the form and follow the instructions in the License Key Request form.



Product purchases and license key requests are typically fulfilled by ADTRAN authorized resellers. Please contact your ADTRAN reseller to purchase additional products or activate a license key.

To review the current licenses on the NetVanta UC Server

1. From the Administration application, select **Help > License Information**
A list of the current enabled features appears. The current license menu includes the following:

- **Description** — The name of the UC server product feature or attribute.
- **Value** — The current licensed maximum number for a specific product attribute.
- **Notes** — The content is usually the actual number of a specific product attribute that is being used by the system.

The current license key field includes the license key that is currently installed.

To review the purchase key and machine ID on the NetVanta UC Server

1. From the Administration application, select **Help > License Information**
2. Select **Modify License...**

The purchase key and machine identifier are shown. You can copy the contents to the Windows clipboard by selecting the purchase key or machine identifier and press **Ctrl-C**.

To modify the license key on the NetVanta UC Server

1. From the Administration application, select **Help > License Information**
2. Select **Modify License...**
3. Enter the new license key that you received from ADTRAN.

If the license key was delivered in electronic format, simply cut-and-paste the license key from the email body or accompanying attachment containing the license key into the **Enter upgrade field**.

10 Using SIP Telephones

This chapter presents the basics of using SIP telephones, and includes the following topics:

- *Overview on page 155*
- *Customizing your Phone on page 155*
- *Using SIP Phones on page 156*

Overview

The SIP solution provides extensive telephone features including superior voice quality, customer cost-savings, support for real-time communications, and improved efficiency of communications.

SIP-enabled phones, in conjunction with the UC server, offer a wide range of phone features, such as call park/pickup, overhead paging, auto attendant, single button voicemail access, caller ID, conferencing, hunt groups, ring groups, message waiting indicator, music on hold, and more.

The UC server works with many types of SIP-compliant phones. However, phones from different manufacturers are configured differently, and the phone features are accessed in a different way.



This chapter does not replace individual phone installation and configuration manuals. SIP telephone user guides for manufacturers that are part of the ADTRAN Connect or Connect PLUS program are available on the ADTRAN website www.adtran.com/support. Refer to the SIP user guide for information about how to use the phone with the UC server.

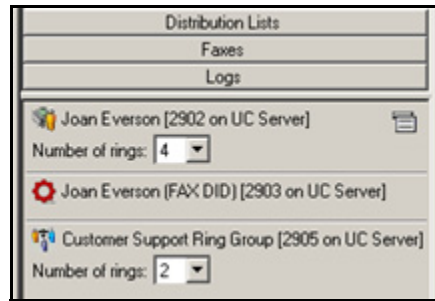
Customizing your Phone

In the UC server you can customize your phone to suit your personal preferences. For example, you can set the number of rings before your phone goes to voicemail. You can also customize your phone according to your particular phone's make and model. For information on customizing your phone in the UC server, refer to the following section.

Changing the number of rings before calls go to voicemail

You can change the number of times a call rings on your extension, before the call is directed to voicemail.

An identity refers to a user extension, an auto attendant, or an email address. If you have identities assigned, they appear in the identities pane, on the bottom-left of the UC client window.



To change the number of rings

- On the identities pane, select the identity that you want to change, and select the **Number of rings** from the drop list.

The default number of rings is 4.

Using SIP Phones

Depending on the manufacturer, SIP phones and the UC server offer a variety of features, such as the following:

- Accessing voicemail messages
- Transferring callers directly to voicemail
- Using the NetVanta UC Server conference bridge
- Paging
- Parking calls and picking up parked calls
- Working with call queues
- Picking up ringing calls
- Changing phones during a call

Accessing voicemail messages

Some telephones provide a single button to access your voicemail. This is useful when accessing messages from your phone. You can press the Messages button and go directly to voicemail to check your messages.

You might have one or several identities (typically extensions) assigned to your telephone. The number of buttons on the phone determines the number of possible identities that can be configured on it.

To use the message button on a phone with one identity

1. Press the **Messages** button on your phone. The phone automatically dials a SIP URL “voicemail.”
2. Enter your telephone Personal Identification Number (PIN) at the prompt to access your personal voicemail.

To use the message button on a phone with more than one identity

1. Press the **Messages** button on your phone.
2. From the Line Select menu, select using the telephone keypad, or the arrow keys on the telephone line for which you want to review your messages. The phone automatically dials a SIP URL “voicemail” from the line that you selected.



As long as there is a user profile associated with the identity, callers are prompted for their telephone PIN.

Transferring callers directly to voicemail

You can transfer a caller directly to voicemail, without ringing the recipient’s telephone. This feature is useful if a caller needs to leave a message for a user, but does not need to speak to the user directly. After a direct transfer, the caller hears the user’s personal assistant greeting, and can follow the prompts accordingly.

To perform a blind transfer to another user’s voicemail

1. During an active call, begin a blind transfer, which puts the caller on hold. The steps to perform a blind transfer vary depending on the type of phone you are using. For more information, refer to the user documentation for your phone.
2. Enter ***86<Destination mailbox number>**.
3. Press **Send** or **Dial**, depending on the phone type you are using.

To perform a supervised transfer to another user’s voicemail

1. During an active call, begin a supervised transfer, which puts the caller on consultation hold. The steps to perform a supervised transfer vary depending on the type of phone you are using. For more information, refer to the user documentation for your phone.
2. Enter ***86<Destination mailbox number>**.
3. Press **Send** or **Dial**, depending on the phone type you are using.
4. To complete the transfer, press the **Transfer** key again.

To leave a message directly in a recipient’s mailbox

1. Enter ***86<Destination mailbox number>**.
2. Press **Send** or **Dial**.

Using the NetVanta UC Server conference bridge

The UC server has the ability to support a limited-sized conference bridge application. The number of participants is limited to the number of licenses available, however additional conference expansion licenses can be purchased. You can use your SIP phone during the conference to access several options,

such as muting or adding participants.



The default conference bridge number is 7050, however this number can be changed. To change the conference bridge number refer to [To edit an existing dial plan on page 44](#)“.

The conference bridge number is associated with a conference and not a user identity.



The conference bridge number must be entered in the list of valid identities, in order to allow inbound callers to be transferred to this number. For more information, refer to [Configuring non-user extensions and identities on page 31](#).



Any lone user in a conference bridge hears the sound of “crickets,” which is confirmation that they are the only person in the conference.

To access the conference bridge

1. From an internal telephone, dial the conference bridge number. The default number is **7050**.



An outside caller can also access the conference through the attendant services, using the dial by extension or dial by name directory.

2. Enter the 4-digit **conference ID**.
3. Follow the prompts to access the conference bridge.



*To access the list of options, you can **press *** while the conference is in progress.*

Conference option list

When you press * during a conference, the following options are available. The available options depend on at which point during the conference you press *. Only relevant options are available.

- **Press 1** to mute yourself. Response: “You are now muted.”
- **Press 2** to unmute yourself. Response: “You are now unmuted.”
- **Press 3** to disable entry and exit notifications. Response: “Entry and exit notifications have been disabled.”

- **Press 4** to mute all other participants. Response: “All others have been muted.”



When you press 4 to mute all participants, this does not apply to new callers who join the conference. To mute new callers, press 4 again.

- **Press 5** to unmute all other participants. Response: “All others have been unmuted.”
- **Press 7** to toggle between closing and reopening the conference to additional participants. Response: “The conference is now closed to additional participants.” Response: “The conference is now reopened to additional participants.”
- **Press 9** to end the conference.
- **Press 0** to report the number of participants. Response: “Including yourself, there are x participants.”
- **Press #** to return to the conference.

Paging

You can send live pages. Live paging allows you to instantly broadcast a page to an analog audio device. You can also send a live or recorded page from a call answering service. For more information, refer to [Page Live element on page 263](#) or [Page Recorded element on page 264](#).

To send a live page

1. From your phone, dial *72 (or *PA).
2. After you hear the tone, begin speaking.
3. When you are finished with the page, hang up or use the end call function on your phone.

Sending a live or recorded page using sound card paging

If you are using sound card paging, then you can send a live page, or you can first record the page, and then broadcast it.



While the instructions below for sending live and recorded pages are still supported, the recommended method is to use the instructions above for sending a live page. To configure sound card paging to use the instructions above, refer to [Configuring Sound Card Paging on page 63](#).

To send a live page

1. From your phone, dial *75.
2. After you hear the tone, begin speaking.
3. When you are finished with the page, hang up or use the end call function on your phone.

To send a recorded page

1. From your phone, dial *77.
2. When prompted, record your message, and then use the options below.

Stop recording	Press #
Playback recording	Press 7
Re-record	Press 6
Broadcast page	Press 5

Parking calls and picking up parked calls

You can park an active call, and the parked call can be retrieved on another phone. This feature is useful if you want to change phones during a call. This feature depends on support from a SIP server.



Call park/pickup must be enabled for use. To enable call park/pickup, refer to [Configuring system call options on page 40](#).

There are two methods of parking calls. In one method, the UC server selects the next available park number. Alternatively, you can select your own park number. Having the UC server select the park number is useful if you are uncertain whether a call is already parked at a particular number.

If more than one identity (for example, more than one extension number) is assigned to your phone, then transfer the call rather than putting the active call on hold.



If a person whose call has been parked then puts the call on hold, the person retrieving the parked call will hear dead air or silence until the other party resumes the call.

To park a call at the next available number

1. Put the active call on hold.



*If your phone has more than one identity assigned to it, then **Transfer** the call instead of putting it on hold.*

2. Dial *99.
3. Note the **hold position** that plays over the telephone. The hold position is used by the person who answers the parked call. The UC server selects the park number for the caller starting at 1, or the next available higher number.
4. Press **DIAL**.
The call is removed from the hold position, and put in the system park position.

To park a call at a selected park number

1. Put the active call on hold.



*If your phone has more than one identity assigned to it, then **Transfer** the call instead of putting it on hold.*

2. Dial ***99**.
3. Enter the number that you want to use as the park number. The hold position is used by the person who answers the parked call. For example, if you enter 1234, then the call is parked at 1234, and can be picked up at 1234.
4. Press **DIAL**. The call is removed from the hold position, and put in the park position.

To answer a parked call

1. Typically, a company announcement informs you that a call is on hold for you.
2. Dial ***98**. The system prompts you for the park hold position.
3. Dial the number of the park hold position. The system disconnects the current call and presents a new call.
4. Answer the new call to speak to the parked caller.



If nobody answers the parked call, the call rings the extension that originally put the call on system park.

Working with call queues



For information on routing calls to queues, refer to [Routing Calls to Queues on page 52](#).

Call queuing allows incoming calls to be distributed to an individual within a specified group of agents. The feature is useful for companies that handle a lot of incoming phone calls, and where the caller is not trying to reach a specific individual, but wants to talk to a member of an agent group, such as a salesperson, as quickly as possible. Call queuing allows callers to remain in a queue while they wait for their call to be answered by the appropriate agent.



The total number of simultaneous calls that can be queued in the system is 64.

The agent is typically associated with a particular area of expertise, for example, sales or support. A support agent who is ready to answer calls can log into the call queue and receive support calls in a

specified order, for example, in order of arrival or priority.

Calls are distributed to the most idle agent. If an agent is logged in, the caller hears an introductory queue greeting. If no agents are logged in, the caller is directed to the overflow destination. The overflow destination then provides an additional greeting.

If callers know the queue number, they can dial directly into a queue.

Logging in and out of a queue

Agents can log in or log out of a queue. Agents who log in to a queue receive calls that are directed to that queue.



The maximum number of agents that can log in to a queue at the same time is 64.

To subscribe to a queue

1. Dial *81<QueueNumber>. Ensure that you enter the correct queue number.
2. Hang up.



To avoid errors when entering queue numbers, use speed dial to login and out of queues without errors.

To unsubscribe from a queue

1. Dial *82<QueueNumber>.
2. Hang up.

To present a call to the queue

1. Dial *80<QueueNumber>.
2. Hang up.

Temporarily stop accepting queued calls

If you are logged into a queue to accept incoming queued calls, there may be times when you wish to temporarily stop accepting queued calls, without logging out of the queue. You can manually set your status to unavailable to prevent accepting further incoming queued calls, without logging out of the queue.

Temporarily setting your status to unavailable does not change your longest idle position in the agent group. When you make yourself available again by logging back into the queue, your idle time is based on the time since your last call, including the time that you were unavailable.

To set your agent status to unavailable

1. Dial *83.
2. Press a number from 1 to 9 to indicate your reason for being unavailable.
3. Hang up.

To set your agent status to available

1. Dial *83.
2. Press **0** at the prompt.
3. Hang up.

Checking your agent status in a call queue

You can check your agent status to verify whether you're currently logged into a call queue, whether your status is set to temporarily unavailable, or whether you're currently logged out of the call queue.

To check your agent status

1. Dial *84<QueueNumber>. Ensure that you enter the correct queue number.
The system will indicate your current status as:
 - Logged in
 - The agent is not available, service code <#>
 - Not Logged in
2. Hang up.

Picking up ringing calls

Ringling call pick up allows you to answer other ringing telephones, which means that you no longer have to run to answer another person's telephone. You can divert a ringing call from another telephone, typically within earshot, to your own telephone to answer.

To use ringing call pick up, you must know the extension of the phone that is ringing.

This feature also allows for night bell support. Many small companies have an audible bell that rings after business hours. Night bell support can also allow employees to contact security after hours. This feature can allow employees, for example on manufacturing floors, who are not stationed at a particular desk to answer any telephone.



*To use ringing call pick up, your phone must be configured to receive incoming calls. Make sure that such features as **call forwarding** and **do not disturb** are not enabled on the phone.*

To answer a call on another user's phone

1. When you hear a phone that you want to answer, dial ***78<Extension>**. Make sure that you dial, on your own phone, *78 plus the extension of the phone that is ringing.
2. If the phone has stopped ringing by the time you dial the *78 code, the system prompts you to enter your pickup ID.



If the pickup ID that you enter is invalid, the system prompts you with "That is not a valid pickup ID," and asks you to enter a valid pickup ID.

Changing phones during a call

Using connected call pickup, you can seamlessly transfer calls from one device to another, such as from a cell phone to your desk phone. This feature is most commonly used with mobile twinning, where calls to one phone number can ring simultaneously at multiple devices, such as an internal extension number and a cell phone number. For example, if you were away from your desk and you answered a call that was transferred or twinned to your cell phone, then when you return to your desk, using the connected call pickup feature, you can pickup that call on your desk phone, without the caller being aware that you changed phones.



For more information on mobile twinning, refer to [Configuring mobile twinning using a ring group identity on page 113](#).

Benefits of using connected call pickup

The benefits of using connected call pickup include using one phone number and multiple devices to be accessible to callers, while minimizing costs by then transferring calls to your internal extension or to another land line phone.

Requirements for using connected call pickup

- To pickup a connected call, the called number must go through the UC server. That is, the call must be transferred, forwarded, or twinned to your alternate phone number. For example:
 - Calls can be made directly from another extension to your alternate phone number
 - Calls can be transferred by internal users to your alternate phone number
 - Calls can be transferred using the Transfer Call options available from your Personal Assistant or Personal Business Assistant to your alternate number
 - Using mobile twinning, calls can ring your alternate phone number and your internal extension simultaneously
- Unless you configured this mode differently, you must know the phone number of the phone that answered the call, that is, the extension number or the external telephone number, in order to pickup the connected call.

- Your phone must be able to receive another call. That is:
 - A line must be available
 - Do Not Disturb (DND) must be off
 - Call forwarding must be off
- The connected call pickup feature must be enabled.



The connected call pickup feature is very powerful in that it allows any user to pickup any connected phone call that goes through the UC server, as long as they know the number of the phone that answered the call, such as the extension number or the cell phone number. This feature is enabled by default. For information on restricting this feature, refer to [Restricting picking up connected calls on page 50](#).

To change phones during a call

1. From the phone that you want to transfer the call to, dial ***27** <called phone number>. The called phone number can be an internal extension number or an external phone number, and refers to the number of the phone that answered the call. If you restrict this feature, you can configure this mode so that a user can only pickup connected calls from their desk phone. If this is the case, then dial ***27** only. To connect the call, the phone that you are dialing from rings.
2. Answer the ringing phone.

When you answer the phone, the call automatically connects to this phone, and is dropped from the phone that you originally answered the call with. The call is transferred from one phone to another silently, so the user may be unaware that you have changed phones.

11 Faxing

This chapter provides information on receiving faxes on trunk lines, sending faxes from other applications, sending cover pages only using Quick Fax, managing faxes, creating and using fax cover pages, queuing faxes, and automatically sending faxes requested by callers.

This chapter covers the following topics:

- [Overview on page 167](#)
- [Sending Faxes on page 168](#)
- [Receiving Faxes on page 191](#)



For information on setting fax specifications for outgoing faxes or when using the Fax-on-demand element, refer to [Setting fax message delivery options on page 26](#). For information on setting dialing options related to faxing, refer to [Setting Outbound Calls and Dialing Options on page 32](#).

Overview

Use the Faxes menu to send and manage fax documents and their cover pages from the UC client. There are several methods of sending faxes including:

- Sending a fax from other applications using the ADTRAN Fax print driver
- Sending a cover page only from the UC client using Quick Fax
- Sending a prepared fax from the UC client
- Automatically sending faxes requested by callers using the Fax-on-demand element in a service.
- Sending faxes to a specific fax number using the Send fax element

The Faxes menu is not used for receiving faxes. Incoming faxes are received and viewed through a user's mailbox, for example, through Microsoft Outlook, Microsoft Exchange Server, Lotus Notes, or through an Integrated messaging client (a Mailboxes navigation bar on the user's UC client) or IMAP4.

The UC server can always receive a fax, provided adequate fax channels are available, whenever a fax tone (CNG tone) is detected.

Other methods of receiving faxes include:

- A dedicated fax number (fax identity). That is, a Personal Fax DID (direct-inward-dial) number.
- A shared voice and fax number (using the same identity)



A shared voice and fax number is not recommended for high numbers of incoming fax calls. If an incoming fax call is answered at the user's extension, the user will have difficulties transferring the call into their service.

You can also use the receive fax element in a service to customize receive fax behavior.

Sending Faxes

You can send a fax document from another application or using the Faxes menu of the UC client. When a fax document is ready to send, the fax is added to the fax queue. Before you send a fax, you can change the user that you want to send the fax from. This is useful if a user manages several mailboxes, or as part of a Fax-on-demand element.



For information on setting fax specifications for outgoing faxes, refer to [Setting fax message delivery options on page 26](#). For information on setting dialing options related to faxing, refer to [Setting Outbound Calls and Dialing Options on page 32](#).



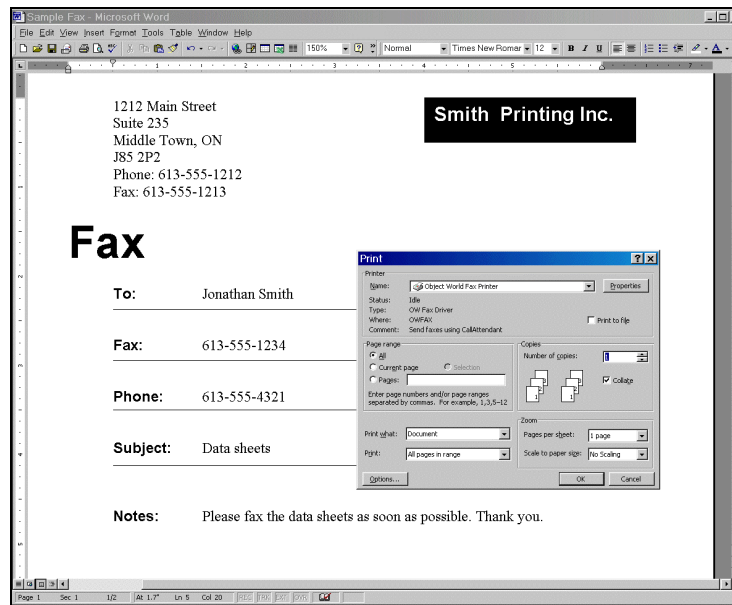
You can send a fax cover page only using Quick Fax. For more information, refer to [Sending cover pages only using Quick Fax on page 190](#).

Sending faxes from other applications using the ADTRAN Fax Printer

You can send faxes from other applications using the ADTRAN Fax Printer. You can also change user profiles to send a fax from another user, or to save a fax to another user's profile.

To send a fax document from another application

1. Open the document that you want to send as a fax.
2. Select **File > Print** to open the Print dialog box.



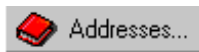
3. From the Printer Name list select the ADTRAN Fax Printer, and select **OK**.

4. If you are prompted, enter the User name and Password for the user from which you want to send the fax, and select **OK**.
5. In the UC client Fax Printer dialog box, select one of the following:
 - **Send now**—sends the fax now, after having made your fax selections
 - **Save to file**—allows you to save the file as a fax to send later



To change the user from which you want to send the fax, refer to [To change users when sending or saving a fax on page 170](#).

6. If you select **Save to file**, refer to [To save a document as a fax on page 169](#).
7. If you select **Send now**, select **OK** to open the Fax Recipients dialog box.
8. Select **Addresses** to select recipients from your Outlook Contacts list. Otherwise, type the recipient names and fax numbers.



9. Select **Add to List** for each recipient that you specify.



*To delete a recipient from the Recipient list, select the recipient and then select **Remove**.*

10. Select **Next** and the Fax Scheduling dialog box opens.
11. Specify when you want to send the fax, select **Next**, and the Select Cover Page dialog box opens.
12. To include a fax cover page select **Yes**, and double-click a folder to view its contents.
13. Select the cover page that you want, select **Next** and the Cover Page Details dialog box opens. Use the arrow if you want to return to the previous folder.



Go back arrow

14. Type the subject and notes that you want to add to the cover page, and then select **Next**.
15. At the Finished menu select **Finish**. The fax is sent according to the instructions.

Saving a document as a fax to send later

To save a document as a fax

1. Open the document that you want to save as a fax.
2. Select **File > Print** to open the Print dialog box.
3. In the Name list, select the **ADTRAN Fax Printer**, and select **OK**.
4. If you are prompted, then enter the user name and password for the user from which you want to send the fax, and select **OK**.

5. If the Server dialog box opens then select **UC Server** from the list, enable the **Save server as default** check box, and select **OK** to open the UC client *Fax Printer* dialog box.
6. Enable the **Save to file** button.
7. Select the **Browse** button to open the *Select Fax Name* dialog box.



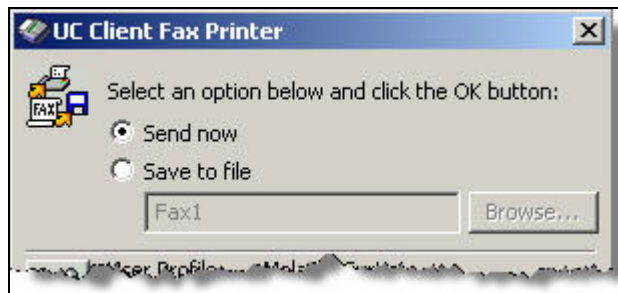
Browse Button

8. In the text box, type a name for the file and then select **Select**.
9. On the Fax Printer dialog box, select **OK**. The fax document is saved in your fax documents folder.

Changing users when sending a fax or saving a fax to send later

When you send a fax from another application, or when you save a document as a fax to send later, you can change the user that you want to send or save the fax from. This is useful if a user has several mailboxes, or as part of the Fax-on-demand element.

If you are faxing from another application, or saving a document as a fax, then from the *Print* dialog once you have selected the **ADTRAN Fax Printer** and selected **Print**, then you are presented with the UC client *Fax Printer* dialog.



To change users when sending or saving a fax

1. From the UC client *Fax Printer* dialog box, select the **Change user** button.



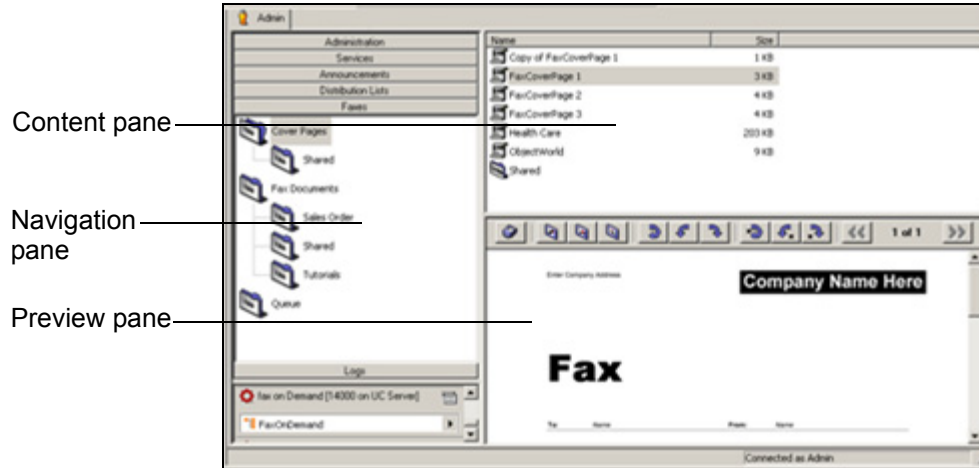
Change user button

2. Type the User name and Password for the user from which you want to send the fax. The user's information appears in the **Current user information** section of the Fax Printer dialog box.
3. Select **OK**.

Sending faxes using the Faxes menu in the NetVanta UC Client

From the UC client, to get the Faxes menu select the Faxes navigation bar.

For information about sharing faxes, refer to *Importing and Exporting on page 311*.



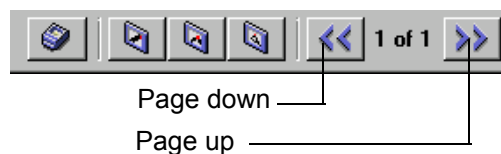
The **Faxes menu** displays a Navigation pane, Content pane, and Preview pane.

The **Navigation pane** displays folders for fax cover pages, fax documents, and the fax queue. You can use these folders to store fax documents and cover pages that are saved to disk. The Queue folder contains the list of scheduled fax transmissions. Each entry includes the recipient's name and fax number, the number of pages to be transmitted, the time and date of transmission, and the cover page that you have associated with it.

The **Content pane** displays the contents of a folder when you select it. These contain all the fax documents that you have prepared to use on trunk-line services containing the Fax-on-demand or Send Fax element, as well as the cover pages that you use to accompany such transmissions. When you select the fax Queue folder in the Navigation pane, a list of all faxes scheduled for transmission appears in the Content pane. Using the Administrator tab, you can see all faxes queued for delivery in the system. You can view the recipient of the fax, the fax status, size and cover page, and cancel or suspend any fax transmissions that you want. However, you can only view change the fax documents that you prepare for transmission yourself.

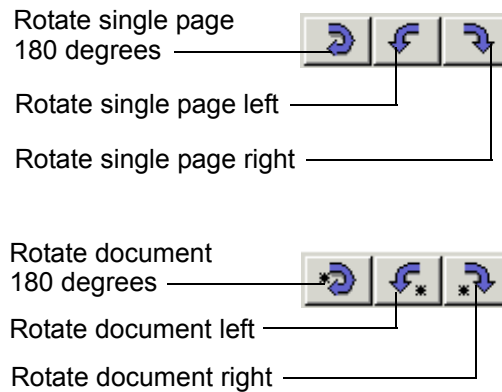
The **Preview pane** provides a partial view of a fax document that you select. As you scroll through the list of available fax documents, you can view them in the Preview pane.

Use the **Page up** and **Page down** commands on the Preview toolbar to view multiple pages of a fax document.



Use the Rotate commands on the Preview toolbar to change the orientation of a fax in the Preview pane.

You can rotate only one page of a fax document, or the entire document.



To view faxes in more detail, you can use the Fax Viewer. For more information about the Fax Viewer, refer to [Managing outgoing faxes on page 173](#).

To send a fax using the Faxes menu

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the Fax Documents folder containing the document that you want to send. The folder's contents are listed in the content pane.



Faxes Folder

3. Select the fax document that you want to send.
4. Select **File > Print** to open the Print dialog box.
5. Follow steps 3 to 18 in [Sending faxes from other applications using the ADTRAN Fax Printer on page 168](#).

To specify fax numbers and account code

1. Select the telephone icon in the Fax Recipients dialog box.



Telephone icon

2. Type the area code and telephone number for the fax that you want to specify.
3. Choose a country code from the Country code list.
4. Enable the **Use Account Code** check box. The administrator must enable account codes from **Tools > Options > Dialing tab** in order for this mode to be available.
5. Type an account code in the text box, and select **OK**.

Sending faxes to a specific fax number using the Send Fax element

The Send Fax element allows you to set up a list of fax documents to be sent to a specific fax number. You can specify the fax number, choose a cover page, and then select the list of documents that you want to send. For more information, refer to [Send Fax element on page 243](#).

Automatically sending faxes requested by callers using a Fax-on-demand element

You can use a Fax-on-demand element to set up a fax-on-demand service on a trunk line. The Fax-on-demand element allows you to select a fax document and an accompanying cover page, and then have the fax transmitted to callers who select the service from a menu. When callers select Fax-on-demand, they are prompted to enter a fax number. The fax that you selected is then transmitted. For information about using the Fax-on-demand element, refer to [Fax-on-demand element on page 247](#).



For information on setting fax specifications when using the Fax-on-demand element, refer to [Setting fax message delivery options on page 26](#). For information on setting dialing options related to Fax-on-demand, refer to [Setting Outbound Calls and Dialing Options on page 32](#).

Managing outgoing faxes

You can preview or print fax documents that you have prepared for transmission. Likewise, you can prepare a document, save it, and make it available for transmission at a later time. Prepared fax documents can be stored in folders for efficient organization and easy retrieval. You can also copy or rename a fax.

A fax transmission report is created and sent with each fax that is sent successfully. A fax transmission report consists of a graphic page image that contains both the text-based transmission report, and a reduced-size image of the first page of the fax. The report includes information such as the date and time the fax was sent, sender's name, fax number, and total number of attempts. Depending on how your system has been configured, you may be able to print the fax transmission report, along with the first page of the fax, by using the **File > Print First Page** command in the Fax Viewer.

For information about specifying fax transmission report options, refer to Setting system-wide options, [Setting fax message delivery options on page 26](#).

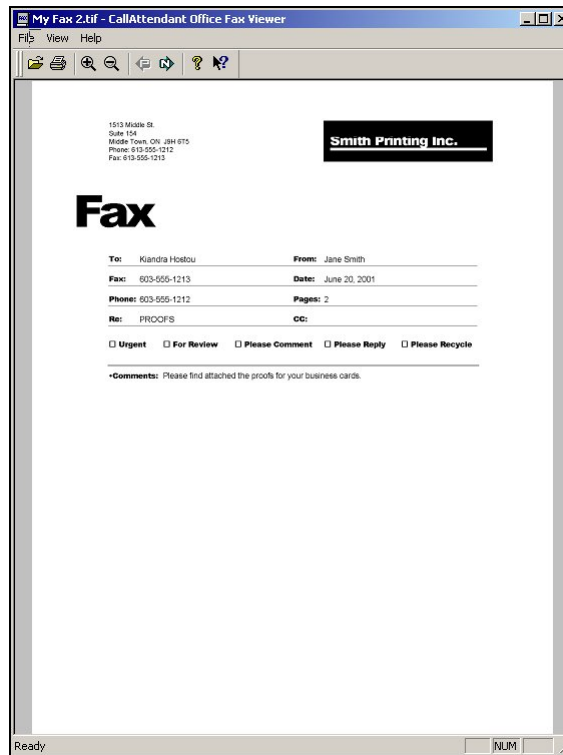
To preview a fax document

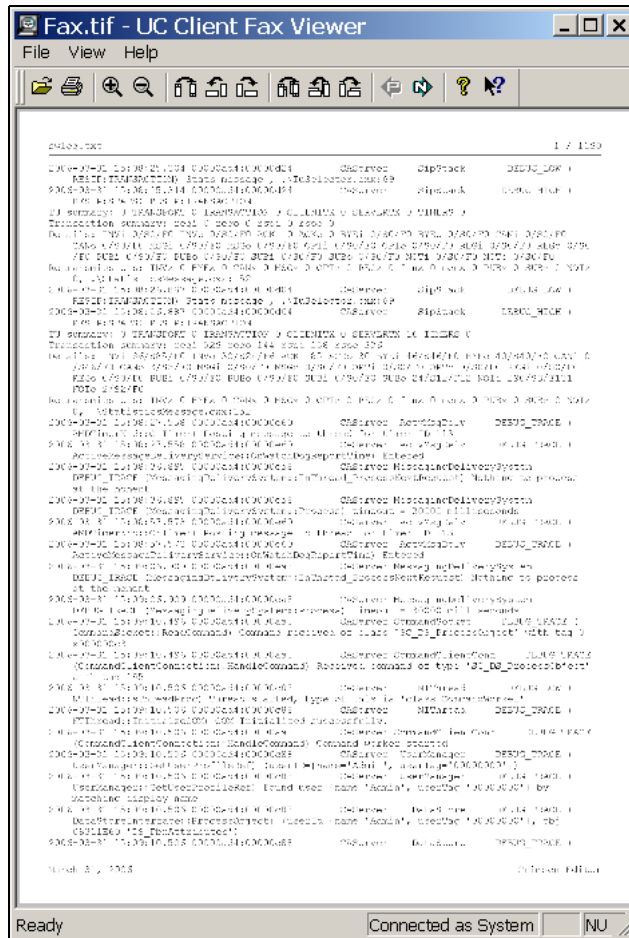
1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the fax documents folder containing the document that you want to preview. The folder's contents are listed in the content pane.



Faxes Folder

3. Right-click the fax document that you want to preview, and select **Open** to display the fax in the Fax Viewer.





Use the zoom control icons to zoom in or out of the fax.



Use the rotate control icons to rotate only one page of a document, or the entire document. Rotate to the left, right, or 180 degrees.



Use the page control icons to scroll through a multiple-page document.

Previous page  Next page



To print the fax document from the Fax Viewer, select the printer icon on the Fax Viewer toolbar.

To print a fax document

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the fax documents folder containing the document that you want to print. The folder's contents are listed in the content pane.



Faxes Folder

3. Right-click the fax document that you want to print, and select **Open** to display the fax in the Fax Viewer.
4. Select **File > Print Setup** to open the Print Setup dialog box.
5. Select a printer, and select **OK**.

To print the first page only of a fax document

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the fax documents folder containing the document that you want to print. The folder's contents are listed in the content pane.
3. Right-click the fax document that you want to print, and select **Open** to display the fax in the Fax Viewer.
4. Select **File > Print First Page** to print the first page only of the fax. Depending on how your system is configured, the fax transmission report may also be printed.



You can configure whether the fax transmission report is printed in addition to the first page of the fax. To configure this mode, refer to [Setting fax message delivery options on page 26](#).

To rename a fax document

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the fax documents folder containing the document that you want to rename. The folder's contents are listed in the content pane.
3. Right-click the fax document that you want to rename, and select **Rename**.
4. Type a new name for the fax document.

To copy a fax document

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the fax documents folder containing the document that you want to copy. The folder's contents are listed in the content pane.
3. Right-click the fax document that you want to copy, and select **Duplicate**. A copy of the fax document appears in the content pane, with the words "Copy of" before its name. You can rename the copied document.

To delete a fax document

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the fax documents folder containing the document that you want to delete. The folder's contents are listed in the content pane.



Faxes Folder

3. Right-click the fax document that you want to delete, and select **Delete**.
4. Select **Yes** to confirm the deletion.

When you create a fax document for transmission at an undetermined time, you can save it to a fax documents folder. That makes it easy to find the fax when you open the Faxes menu and schedule its transmission.

To create a folder for fax documents

1. Right-click anywhere in the content pane.
2. Select **New > Folder**. A new folder is added to the list in the content pane.
3. Type a name for the folder.

To put a fax document in a folder

Select the fax document that you want to store in a folder and drag it to the folder where you want to store it.



Select a folder to view its contents.



On the toolbar, use the Go back arrow to return to the list previously displayed in the content pane.

Viewing the fax queue and status

Select the Queue icon in the navigation pane of the Faxes menu to view all the faxes that you have

scheduled for transmission.

You can suspend fax transmissions indefinitely, resume transmission, or delete an item from the queue.

All faxes that have been prepared by users on the system are visible to you in the list. However, as Administrator you cannot view, forward, or change individual faxes unless you have prepared them yourself. Faxes requested using the Fax-on-demand and Send Fax elements also appear in the fax queue.

When a fax has been successfully transmitted, it is removed from the fax queue. Users receive an email message signaling successful transmission. As Administrator, you do not have a mailbox. Instead, an email message is sent to the general or default mailbox in order to signal successful transmission of fax documents sent by the Administrator. If no general mailbox has been set up, you are not advised of transmission outcomes. The fax transmission includes a history of the attempts and outcome of the fax. A fax transmission report is also available in the call logs.

You can specify the maximum number of transmission attempts. For more information about setting fax options, refer to [Setting System-Wide Options on page 19](#).

If a fax is not successfully transmitted after the maximum number of attempts, users receive an email message to signal the failure.

To specify the general mailbox

1. Select **Tools > Options**.
2. Select the **Operator Definition** tab.
3. In the Default mailbox box, specify a default mailbox.

To specify the maximum number of transmission attempts

1. Select **Tools > Options**.
2. Select the **Fax messages** tab.
3. Type the number of transmission attempts in the **Try** box.

To view the fax queue

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Queue** folder icon. The folder's contents are listed in the content pane.



For each fax in the list, you can see the recipient's name and fax number, the scheduled transmission time and date, and the number of pages including the cover page.

To view an individual fax listed in the fax queue

1. Select the **Queue** folder. Faxes scheduled for transmission are listed in the content pane.
2. Right-click the fax that you want to view, and select **Open**.

To change recipients or fax numbers of a fax listed in the fax queue

1. Select the **Queue** folder. Faxes scheduled for transmission are listed in the content pane. You can change a scheduled fax only if it is in waiting status.
2. Right-click the fax that you want to change, and select **Modify**. The Recipients dialog box opens.
3. Make the changes that you want, and then select **Next**.
4. To remove a recipient from the list, select the recipient in the list and then select **Remove**.
5. To change a recipient's fax number, select the recipient and then make the change.
6. Select **Next** until the Finished menu appears.
7. Select **Finished**.

To reschedule transmission of a fax listed in the fax queue

1. Select the **Queue** folder. Faxes scheduled for transmission are listed in the content pane. You can reschedule a fax only if it is in waiting status.
2. Right-click the fax whose transmission you want to reschedule, and select **Reschedule** to open the Scheduling dialog box.
3. Change the scheduling information that you want and then select **Next**.
4. Select **Next** until the Finished menu appears.
5. Select **Finished**.

To forward a fax listed in the fax queue

1. Select the **Queue** folder.

Faxes scheduled for transmission are listed in the content pane. You can reschedule a fax only if it is in waiting status.



2. Right-click the fax that you want to forward, and select **Forward** to open the Recipients dialog box.
3. Select **Addresses** to select a recipient from your Outlook contacts list. Otherwise, type the name and fax number of the recipient to whom you want to forward the fax.



When you forward a fax document, it appears as a new entry in the fax queue.

4. Select **Add to list**.
5. Select **Next** until the Finished menu appears.
6. Select **Finished**.

To suspend transmission of a fax listed in the fax queue

1. Select the **Queue** folder. Faxes scheduled for transmission are listed in the content pane. You can suspend a fax only if it is in waiting status.
2. Right-click the fax whose transmission you want to suspend indefinitely, and select **Suspend**. The fax transmission is suspended indefinitely. View this information in the status field.

To resume transmission of a fax that had been suspended

1. Select the **Queue** folder. Faxes scheduled for transmission are listed in the content pane.
2. Right-click the fax whose transmission you want to resume, and select **Resume**. The fax transmission is resumed. View this information in the status field.

To delete a fax listed in the fax queue

1. Select the **Queue** folder. Faxes scheduled for transmission are listed in the content pane.



2. Right-click the fax that you want to delete, and select **Delete**.
3. Select **Yes** to confirm the deletion.

Using fax cover pages

This section provides information about how to create and manage fax cover pages. A cover page is sent as the first page of your document and usually includes your name, your company name and fax number, as well as that of the recipient, and some information relating to the subject of the fax.

There is a folder for fax cover pages in the navigation pane of the Faxes menu. When you open the Cover Pages folder, your fax cover pages are listed in the content pane. From there you can create new cover pages and manage those that already exist. For information about sharing fax cover pages, refer to [Importing and Exporting on page 311](#).

You can also send only a fax cover page using Quick Fax.

Creating fax cover pages

To create a new fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon.
3. Right-click anywhere in the content pane.
4. Select **New > Fax Cover Page**. A new untitled cover page appears in the content pane.
5. Type a name for the cover page. You can use the Fax Cover Page Editor to design the appearance and content of the cover page that you created.
6. Select **File > Save** after you finish the design.

To use an existing fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon.
3. Right-click the cover page that you want to use. You can use one of the cover pages that is included, which are called FaxCoverPage 1 or FaxCoverPage 2.
4. Select **Duplicate** to create a copy of the cover page.
5. Right-click the copy of the cover page.
6. Select **Rename**.
7. Type a new name for the fax cover page.
8. Double-click the new fax cover page to open the Fax Cover Page Editor and make the changes that you want.



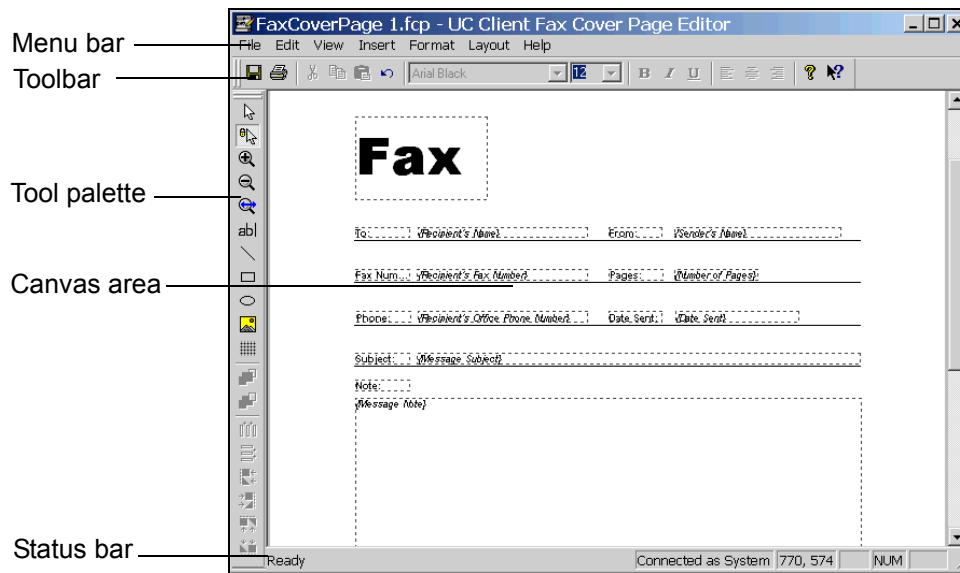
If you do not make a copy of the cover page before you make the changes, the application permanently replaces the original cover page template with your changed copy.

Customizing fax cover pages using the Fax Cover Page Editor

Use the Fax Cover Page Editor to customize the cover pages that you create and to change those that already exist. You can design and manipulate images to give your cover pages a distinctive look, according to your needs and preferences. Your cover page is like a template: you can use it, store it, change it, and reuse it as often as you want. You can access the Fax Cover Page Editor by creating a new fax cover page or by editing an existing fax cover page.

To access the Fax Cover Page Editor

1. From the UC client select the **Faxes** navigation bar to open the Faxes menu .
2. To open the Fax Cover Page Editor, right-click anywhere in the content pane, select **New** in the shortcut menu and select **Fax Cover Page**. Alternatively, to open an existing fax cover page, select a fax cover page, right-click, and select Open.
3. In the content pane, double-click the icon of the new cover page.



The Fax Cover Page Editor consists of a standard Menu bar and Toolbar, a Tool palette, Canvas area, and Status bar. You design your cover page on the canvas using the tools from the tool palette together with the toolbar and menu commands.

The **Menu bar** contains the File, Edit, View, Insert, Format, Layout, and Help menus.

The **Toolbar** contains shortcuts to the most frequently used menu items. Hold the pointer over a button to show the tooltips.

The **Tool palette** contains tools for drawing, editing, and organizing your cover pages.

The **Canvas** is the design area. This is where you customize your cover page.

The **Status bar** shows the pointer's status. For example, if you hold the pointer over a toolbar button, a description of the button's function is displayed on the status bar. The coordinates at the bottom right of the status bar provide precise information about the pointer's position.

Inserting and changing labels and message information

You can include information related to your workplace and your fax message with your cover page. This information can include your workplace name and telephone number, a subject line and short note related to the fax, along with the transmission date, for example. A complete list of information options is provided within this section.

Note that you cannot insert labels with recipient information. This is because recipients are callers who use the Fax-on-demand services that you set up. Such callers are unknown to you. Callers are prompted to enter their fax number when their call encounters the Fax-on-demand element. The fax number is captured and automatically entered on the cover page. The recipient's name may also be captured. This is the only recipient information that appears. The same is true for transmissions initiated by the Send Fax element: the recipient's fax number and name will appear on the fax cover page.

When you create a cover page, you select the type of information you want to include from the list of information options. You do not enter specific sender or message content manually when designing your cover page, because content will vary depending on each Fax-on-demand service that you set up. Instead, you insert labels for the information using the Fax Cover Page Editor. A field is created on the canvas for each of the labels you insert.

To: [Label]

{*Recipient's Name*} [Field]

When a fax is transmitted, the UC server captures the fax number and associated name then inserts the information as content in the field following the appropriate label on the cover page.

When designing the cover page, you can edit a label's font, style and content. For example, you can change the "fax" label so that it appears as "facsimile number." However, you cannot change the actual content that is associated with the field. This is because the content information is captured when a caller initiates a fax request using a service element.

The recipient information labels that you can insert include:

- Name
- Fax number
- Company
- Street Address
- City
- State/Province
- Zip/Postal Code
- Country
- Title
- Department
- Office location
- Home telephone number
- Office telephone number

The sender information labels that you can insert include:

- Name
- Fax number
- Company
- Address
- Title
- Department
- Office location

- Home telephone number
- Office telephone number

The message information labels that you can insert include:

- Note
- Subject
- Date sent
- Number of pages
- Number of attachments

To insert an information label

1. From the **Insert** menu, choose one of the following label types:
 - **Recipient**
 - **Sender**
 - **Message**

In this example, the choice is Recipient.

2. From the submenu, select the label that you want. The label is added to the cover page canvas. It appears in a dotted rectangle. A field is inserted next to the label. The word “recipient’s” identifies the field that appears in a second dotted rectangle.



The label name appears in bold, while the word “recipient’s” is italicized and has a bracket in front of it. The label name is included as content on the cover page, but the word “recipient’s” is not. The application inserts the recipient name associated with a fax number when a caller initiates a fax request.

To reposition a label on the canvas

1. Select anywhere inside the label’s dotted rectangle and then drag it to the desired position.
2. Do the same with the dotted rectangle of the associated field.

To change the font and style of a label or field

1. Select the label or field that you want to change.
2. Do any of the following:
 - Select a font from the font list.



- Select a point size from the size list.



- Select a style by selecting a style button.



You can use the familiar, standard font and text style commands from the toolbar to change the font and style of the cover page, as described above. Alternatively, you can use the menu commands (**Format > Font**) which also allow you to change the script and preview the font that you choose.

To change label content

1. Select the label content that you want to change.
2. Type the changes to the label content.

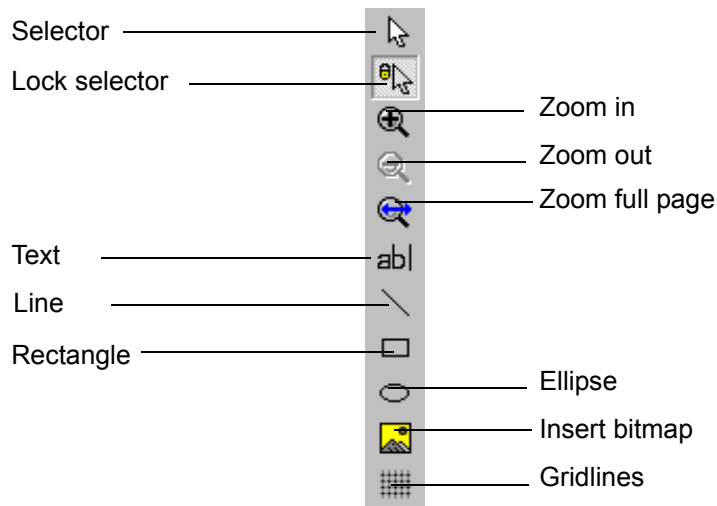


You can change label content, but you cannot change associated field content from within the Fax Cover Page Editor. To change field content related to a specific fax transmission, you need to do so from the fax queue. Select the fax that you want to change and right-click to get a shortcut menu.

Using the tool palette to customize fax cover pages

The tools available on the tool palette of the Fax Cover Page Editor can be divided into two groups: the Text and Drawing group, and the Alignment group.

The tools in the Text and Drawing group allow you to draw shapes and insert text or bitmaps on the canvas. The drawing tools can be selected only from the tool palette. They allow you to select, use shapes, and zoom in on the canvas as you work on your design.



When it is enabled, the Lock selector tool allows you to use the pointer to select and insert a shape, line, or text on the canvas. Once you have inserted the item of your choice, the pointer resumes its standard function.

If you select and insert a shape, line, or text when the Selector tool is enabled, the selector tool does not resume the standard pointer function once you have inserted a shape, line, or text. The Selector tool continues to insert the drawing tool that you chose from the tool palette. It does so until you select the Lock selector tool. This disables the Selector tool.

To insert a shape

1. Select the Lock selector tool.
2. Select the shape that you want to insert.
3. Select on the canvas where you want to insert the shape and then drag down and to the right.
4. Release the mouse button to create the shape.

To reposition a shape on the canvas

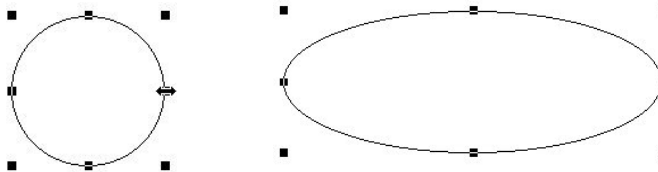
1. Select the shape to select it.
2. Drag the shape to the position that you want.



You can turn on the gridlines. The gridlines help you to position shapes with ease on the canvas.

To change the shape and size of a shape

- Select a handle and drag to change the size or shape of the selected object.



For example, you can change the shape of a circle so that it becomes an ellipse by selecting the handle, as indicated, and dragging to the right.

To delete a shape

1. Select the shape that you want to delete.
2. Press **Delete**.

To draw a line

1. Select the Lock selector tool.
2. Select the Line tool.
3. Select on the canvas where you want the line to begin and drag to draw the line.
4. Release the mouse button to create the line.
5. Select anywhere on the canvas to cancel the selection.

To change or reposition a line

1. Select the line that you want to change to select it.
2. Do either of the following:
 - To shorten or extend the line, select one of the boxes at the end of the line and then drag as desired.
 - To reposition a line on the canvas, select the line and drag it to a new position.
3. Select anywhere on the canvas to cancel the selection.

To delete a line

1. Select the line you want to delete to select it.
2. Press **Delete**.

To insert text

1. Select the Lock selector tool.
2. Select the Text tool.
3. Select where you want to insert the text.
4. Drag down and to the right. A dotted text box appears on the canvas.
5. Type the text that you want to insert. Choose the font, size, and style commands on the toolbar to change the text.

To insert text in a text box

1. Select the lock selector tool.
2. Select the text tool.
3. Select where you want to insert the text box and then drag to the right and down. A dotted text box appears on the canvas.
4. Select **Format > Line, Fill and Color** to open the Line, Fill and Color dialog box.
5. Select the **Draw border/line** check box. The text box lines are now solid.
6. Select line thickness from the Thickness box.
7. Type the text in the text box. Change the font size and style as desired.

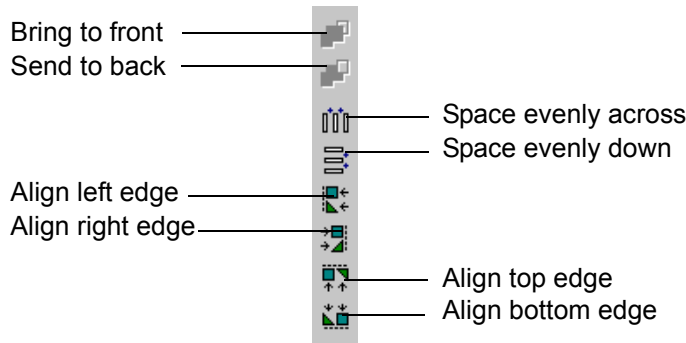
To insert a bitmap

1. Select the Insert bitmap tool to open the Open dialog box.
2. Select the bitmap that you want to insert.
3. Select **Open**. The bitmap is inserted on the canvas.


Using the Alignment tools to customize fax cover pages

The second group of tools in the Fax Cover Page Editor is the Alignment group. These tools are used to space or align selected objects on the canvas. Alignment tool commands can also be found in the Layout

menu.



To move a shape to the front/back in relation to a second shape

1. Select the shape that you want to move to the front/send to the back to select it.
2. On the tool palette, select **Bring to front/Send to back**, as desired.  The shape that you selected is brought forward/sent back.

To space two or more objects evenly across/down the canvas

1. Select the objects that you want to space.
2. On the tool palette, select one of the following:
 - Space evenly across
 - Space evenly down



Spacing tools

The items that you selected are spaced evenly on the canvas.

To align two or more objects on the canvas

1. Select the objects that you want to align.
2. On the tool palette, select the alignment tool that you want to apply.



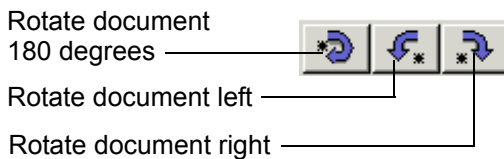
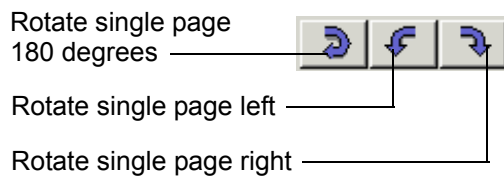
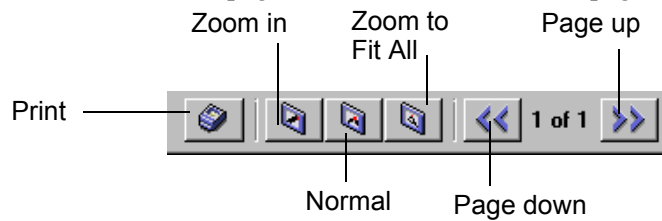
Alignment tools

The items that you selected are aligned by the edges that you specify.

Managing fax cover pages

To preview a fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon. The fax cover pages are listed in the content pane.
3. Select the fax cover page that you want to preview.
4. In the preview pane, select **Download** to get the file from the server. The cover page appears in the preview pane.
5. Use the cover page toolbar to view the cover page.



To rename a fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon. The fax cover pages are listed in the content pane.
3. Right-click the cover page that you want to rename.
4. Select **Rename**.
5. Type a new name for the cover page.
6. Select the icon of the new cover page to make the change.

To copy a fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon. The fax cover pages are listed in the content pane.
3. Right-click the cover page that you want to copy.

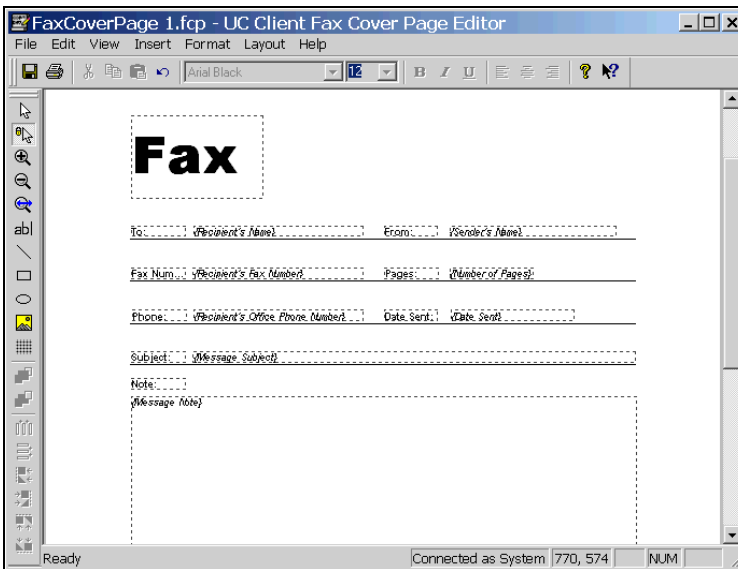
4. Select **Duplicate**. The copy of the cover page appears in the content pane, with “Copy of” before its name. You can rename the cover page if you want to.

To delete a fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon. The fax cover pages are listed in the content pane.
3. Right-click the cover page that you want to delete.
4. Select **Delete**.
5. Select **Yes** to confirm the deletion.

To edit a fax cover page

1. Select the **Faxes** navigation bar to open the Faxes menu.
2. Select the **Cover Pages** icon. The fax cover pages are listed in the content pane.
3. Right-click the cover page that you want to edit.
4. Select **Open**. The fax cover page is displayed in the Fax Cover Page Editor.



Use the toolbar and menu commands to change the fax cover page.

5. Select **File > Save**.

Sending cover pages only using Quick Fax

The Quick Fax feature allows you to send a one-page fax message at any time, to as many recipients as you choose. When you use Quick Fax, you send only a cover page. Your messages appear in the subject area.

To send a cover page only using Quick Fax

1. On the toolbar, select the **Quick Fax** icon to open the Fax Recipients dialog box.



Quick Fax icon

2. Select **Addresses** to select recipients from your Outlook Contacts list. If you are not choosing a recipient from your Outlook Contacts list, type the names and fax numbers.
3. Select **Add to List** for each recipient that you specify.



To delete a recipient from the Recipient list, select the recipient and then select Remove.

4. Select **Next** to open the Scheduling dialog box.
5. Specify when you want to send the fax.



“When phone rates are discounted” means at midnight of the same day.

6. Select **Next** to open the Select Cover Page dialog box.
7. Select a cover page to use as your Quick Fax document.
8. Select **Next** to open the Cover Page Details dialog box.
9. Type the subject of your Quick Fax and then type a message in the Notes section.
10. Select **Next** and the Finished menu appears.
11. Select **Finish**. The Quick Fax message is sent according to the instructions.

Receiving Faxes

There are different methods of receiving incoming faxes.

- The default behavior for incoming faxes is that the UC server detects the standard CNG tone that signals a fax, the UC server receives the fax, and then stores it in a general mailbox. Using email, an office administrator can then distribute received faxes to user mailboxes. No configuration is required for this method to work, however, no notifications are sent to the user that they have a received a fax.
- A company can have a corporate fax attendant identity using a direct-inward-dial (DID) number, which receives all incoming faxes.

- A user can have a shared voice and fax number, using the same identity.



A shared voice and fax number is not recommended for high numbers of incoming fax calls. If an incoming fax call is answered at the user's extension, the user will have difficulties transferring the call into their service.

Incoming faxes are received and viewed through a user's mailbox, for example, through Microsoft Outlook, Microsoft Exchange, Lotus Notes, or through an Integrated messaging client (a Mailboxes navigation bar on the user's UC client) or IMAP4.

Receiving Trunk-Line Faxes

Trunk lines can receive fax transmissions in various ways. Receiving faxes is automatic when the application detects the standard CNG tone that signals a fax, receives the fax message, and then stores it in a general mailbox. Using emails, an office administrator then distributes received faxes to user mailboxes. You can set up this mailbox on the Options tab for Operator Definition. For information about using the Options tab, refer to [Setting System-Wide Options on page 19](#).

When you specify user information, you can assign a DID trunk to a user so that faxes arriving on that trunk go to that user's mailbox. For information about assigning a DID trunk to a new user, refer to [Creating User Profiles on page 85](#). You can also create a fax attendant identity for an existing user using the New Identity Wizard, and selecting **Receive Fax**. For more information, refer to [Using the New Identity Wizard on page 114](#). Whether you assign a fax identity to a new or an existing user, the result is the same. Incoming faxes are received in the user's mailbox, the Message Waiting Indicator (MWI) flashes on their phone, and a voicemail message indicates that they have received a fax.

Fax reception can also be initiated by a Receive Fax element when you have included one in a trunk-line service. In this case, a fax arriving on a trunk line is subject to the trunk's active service.

- You can also use the Receive Fax element in a voice service in conjunction with a Select Extension element so that the fax can be delivered correctly. Thus, when the call encounters the Select Extension element of the call flow, callers are prompted to enter the extension of the user to whom they want to send the fax. From there, the fax message encounters the Receive Fax element. When you have set this element to the "Current Extension," the fax message is routed to the extension that the caller specified and it is stored in the user's mailbox.
- Alternatively, you can create a dedicated "Receive Fax" service to customize receive fax behavior. For more information, refer to [Receive Fax element on page 244](#).

Notifying users of an incoming fax on a shared voice and fax identity

Personal Assistant (PA) users will be notified of an incoming fax using the same notification settings that are configured for Active Message Delivery, Email Notification, or Pager Notification. The settings of each of these options apply equally for voice and fax messages.

Personal Business Assistant (PBA) users can have an identity specifically dedicated to receiving faxes, using a direct-inward-dial (DID) number, and a custom service that defines the behavior when a fax is received.

Alternatively, PBA users can receive faxes on a shared voice and fax identity, but they are not automatically notified of incoming faxes. In order to receive notifications that a fax has been received, a Receive Fax service must be created.

The user or the administrator can create a Receive Fax service using the Receive Fax element. For more information on creating a Receive Fax service, refer to [Receive Fax element on page 244](#).

12 Creating Distribution Lists

Overview

You can use the Distribution Lists menu to create and manage system distribution lists. Distribution lists allow users to send voice and fax messages to groups of users. Users can create their own distribution lists but cannot edit system distribution lists.

Two folders are included with the Administrator application, Everyone and System. The Everyone folder contains all the users that are configured for the system. You cannot add a new folder to the Everyone folder.

Using System Distribution Lists

You can use the System folder to add new distribution lists to the system, and then add, edit, and delete users from the distribution lists. You can record an audio identifier for a system distribution list.

To record an audio identifier for a distribution list

1. In the content pane, right-click the system distribution list for which you want to record an audio identifier, and select **Record Identifier**.
2. In the **Record Identifier for** box, select the **Record** button.
3. Make sure that the audio device you want is selected.
4. From the Insert new audio list, select **replace all**.
5. Select **record**. If you are using your telephone, wait for the beep before you begin to record.
6. Select **stop** when you are finished. If you used your telephone microphone, you can hang up the phone or press the Release button on the phone to end the recording.
7. Select **play** to hear the recording.
8. Select **OK** to keep the recording.

Using Personal Distribution Lists

Use the Personal folder to create distribution lists and then add, edit, and delete users from the distribution lists. You can also record an audio identifier for a personal distribution list.

To create a new distribution list folder

1. Select the **Distribution Lists** navigation bar to open the distribution lists menu.
2. Right-click the **System** folder.
3. In the right content pane right-click, and select **New > Distribution List**.

4. Type a Name for the distribution list folder.
5. Type a unique number in the List Identifier box.
6. If you want to record an audio identifier for the distribution list, select **Record** and follow steps 2 to 7 of [To record an audio identifier for a distribution list on page 195](#).
7. Select **OK**.

To add a user to a distribution list

1. Select the folder that contains the user that you want to add to the list.
2. Drag the user name to the distribution list folder that you want to copy it to.

To delete a user from a distribution list

1. Select the folder that contains the user that you want to delete from the list.
2. Right-click the name of the user that you want to delete, and select **Delete**.

To change distribution list information

1. Right-click the system distribution list that you want to change, and select **Open**.
2. In the General Information section, do either of the following:
 - Type a new name in the Name box.
 - Type a new number in the List Identifier box.
3. If you want to change the audio identifier, select **Record** and follow steps 2 to 7 of [To record an audio identifier for a distribution list on page 195](#).
4. Select **OK**.

To rename a system distribution list

1. Right-click the distribution list that you want to rename, and select **Rename**.
2. Type a name for the distribution list, and select **OK**.

To copy a system distribution list

1. Right-click the distribution list that you want to copy.
2. Select **Duplicate**.

13 Using Advanced Features

This chapter explains how to use advanced features, that require some custom configuration in order to use. This chapter covers the following topics:

- [Using the Outbound Dialer on page 197](#)
- [Customizing System Announcements on the Server on page 212](#)

Using the Outbound Dialer

The Outbound Dialer can be used to make outgoing calls, and to run a service when a call connects. You can customize the service to meet your organization's needs, but typically it might play an announcement to notify the person of some information. It could also request that the person enter a PIN or press a number to indicate that they have received the message, or to input a response into a database.



The Outbound Dialer is an advanced feature that requires some programming to customize for your business requirements.

The Outbound Dialer requires the following to run:

- A customized application that you create, that may access a company database, and that initiates the OutDial.exe
- The OutDial.exe provided by the UC Server that makes the call request to the UC Server
- A customized UC Server service that runs when a call connects

You can create the customized application, script, or .NET application to meet your particular business requirements. Ideally, the application would gather information from a database, such as the names and the phone numbers of the people to call. Your custom application must then initiate the OutDial executable, which makes a call request to the UC Server to initiate the calls. When a call is made successfully, then the service you selected is run.



To create the customized application requires scripting knowledge. Knowledge of databases would also be beneficial.

Outbound Dialer Requirements

For the Outbound Dialer to function optimally, there are some requirements that must be met, and some

options that can be configured.



For best results, the Outbound Dialer should be used in combination with a database. For example, your custom application might access a database to gather the names and telephone numbers of the people to call. While a database is not required to use the Outbound Dialer, not accessing a database limits the capabilities of this feature.

The following requirements must be met in order to use the Outbound Dialer.

- The Outbound Dialer is a licensed option. Only UC Server configurations with the Outbound Dialing Support license have this option available.
- The Outbound Dialer should only be used on trunks with true answer supervision that can accurately determine when the called party answers the phone, such as digital trunks. Using the Outbound Dialer with analog gateways may not function reliably.
- By default only Admin users can use the Outbound Dialer. If you have created a custom authorization store, then PBA users can also use the Outbound Dialer if they have the right to **SubmitOutboundCallRequest**.
- You must create a custom application, script, or .NET application, according to your particular business requirements. For maximum functionality, the application should access a database, and the application must also initiate the OutDial.exe, which logs into the UC Server to make the calls.
- You must create a service to run when a call connects.
- You must configure a port for outbound dialing.

To verify the Outbound dialing support license

1. From the Administrator application, select **Help > License Information**. A list of the current enabled features appears.
2. Ensure that Outbound dialing support has the value **Yes**. If Outbound dialing support has the value **No**, then you do not have a license to use the Outbound Dialer. To change your current license, contact your system integrator or reseller.

Try the Outbound Dialer

The Outbound Dialer requires some customizing to use as a business solution, however we've included a simple example for you to try in order to see how the Outbound Dialer works. Entering the commands below into a Command Prompt causes the OutDial.exe to run. OutDial then logs into the UC Server using the authentication provided, calls the phone number that you provided, and runs a sample service included with the UC Server.



To try the Outbound Dialer example requires having the Outbound Dialer license. To verify whether your configuration has this license available, refer to [To verify the Outbound dialing support license on page 198](#).

To try the Outbound Dialer

1. Open a Command Prompt window by selecting **Start > Programs > Accessories > Command Prompt**.
2. Use the change directory command, **cd**, to change the directory path to the UC Server Bin folder. If the UC Server was installed in the default location, then the command and the path would be:

```
cd \Program Files\ADTRAN\UC Server\Bin
```
3. Enter the outdial command, followed by the parameters listed below, substituting your own values for the information in brackets. The command and the parameters have been placed on separate lines for ease of reading only. The parameters can be listed in any order.

outdial

-h <host>

-u <user name>

-pw <password>

-ph <phone number>

-s \Shared\Default Trunk Service

The **outdial** command initiates Outdial.exe.

The **-h** parameter connects Outdial to the host server. The host value can be the host name of the server or an IP address.

The **-u** and **-pw** parameters refer to the user name and password for authentication, so that Outdial can login into the UC Server to make the call. If you are using single sign-on, then you can omit these parameters.

The **-ph** parameter refers to your phone number, or the phone number that you want Outdial to call. This can be the full phone number or an extension number.

The **-s** parameter refers to the service. The Outdial.exe automatically accesses the Services folder. Within the Services folder we've entered the path to the Shared folder, and to the Default Trunk Service, which is a sample service provided with the UC Server. We've used quotation marks around the path because of the spaces in the service name, so that the Outdial.exe includes the entire path for the **-s**, service parameter.

Outbound Dialer Applications

To use the Outbound Dialer, you need to create a custom application, script, or .NET application that satisfies your particular business requirements. Outbound dialing can be used for a variety of purposes, including the following:

- School Absentee Notifications
- Appointment Reminders
- Emergency Response Notifications
- Cancellation announcements
- Information calls
- Sale announcements
- Product in-stock notifications
- Overdue book reminders
- Wake up calls

- Surveys

School Absentee Notifications

You can use the Outbound Dialer to send school absentee Notifications. In this case, you may want the custom application to access a database and generate a list of absent students and their parents' contact numbers. The custom application then must initiate the OutDial.exe, which logs into the UC Server.

When the server runs the call request, a call is attempted at each phone number listed until a successful connection is made, or until the maximum number of attempts have been made. When a call is made successfully, then the specified service is run. You can customize the service to meet your organization's needs, but typically it might play an announcement to notify a parent or guardian that a student has been marked absent that day. For example, when a call is answered, the caller might hear:

"John Smith is absent from school today."

The service could request that the person press a number to indicate that they have received the message, and this would update the database to indicate that a response was received. Note that if a home number was called, it could be the student who answers the phone.

If the person hangs up or if the call goes to voicemail, then no response is given. The custom application could scan the notification records to determine if any calls were not acknowledged, and if so, new call requests could be made, up to a specified number of attempts.

You could create a service that asks the person to press a number to indicate a reason for the absence, for example:

"Press 1 if your child is ill."

The options available, such as illness, appointment, or unauthorized absence, would be stored in a database, as would the response received.

To prevent students from responding to the call, you could assign a PIN to each parent and ask them to enter it. The number entered would then be compared to the PIN assigned to the parent, to determine whether they match.

Appointment Reminders

You can use the Outbound Dialer to send appointment reminders. In this case, you might want your custom application to extract from a database a list of patients, their contact numbers, and their appointment dates and times. The custom application then must initiate the OutDial.exe to login into the UC Server.

When the server runs the call request, a call is attempted at each phone number listed until a successful connection is made, or until the maximum number of attempts have been made. When a call is made successfully, then the service is run.

You can customize the service to meet your organization's needs, but in this example it might play an announcement to notify the person about their appointment. For example, when a call is answered, the caller might hear:

"John Smith has an appointment at Park Hills Medical Clinic on Wednesday September 7th at 2:00."

Emergency Response Notifications

You can use the Outbound Dialer to send emergency response notifications. Using the example of fire fighters, when a fire is reported, you might want the custom application to extract from a database a list of volunteer fire fighters and their contact numbers. The custom application then must initiate the OutDial.exe, which logs in to the UC Server, to call each fire fighter on the list.

When the server runs the call request, a call is attempted at each phone number listed until a successful connection is made, or until the maximum number of attempts have been made. When the call connects successfully, then the service you have selected is run.

You can customize the service to meet your organization's needs, but in this example it might play an announcement to notify the person about a fire.

You could create a service that requests that the person press a number to indicate that they have received the message.

- If the correct response was received, the database would be updated to show the message was acknowledged.
- If the person hangs up, the call goes to voicemail, or if the number requested wasn't pressed, for example if someone else answers the phone, then the database could be updated to show that the notification was not acknowledged.

The custom application could scan the database notification records to determine if any calls were not acknowledged, and if so, new call requests could be made, up to a specified number of attempts.

Configuring the Outbound Dialer

To use the outbound dialer, some configuration is required. There are also options that do not require configuration, but which can be configured to optimize the outbound dialer.

To use the outbound dialer, you must configure a port for outbound dialing to process the call requests.

If you want, you can increase the number of outgoing calls that can be made simultaneously. Refer to [To specify the maximum number of concurrent calls on page 20](#) for more information.

Server activities regarding outbound dialing are automatically noted in the Sent Items logs, and can be viewed in the UC Server Client. The information that is logged includes the user making the call request, the result, the called number for successful calls only, the date and time, the number of attempts, and any error conditions or failures to complete a call. If you want, you can create custom log entries for services that are run.



The logs are placed in the Sent Items log category.

To configure a port for outbound dialing

1. In the Administration navigation pane, select **Ports**.
2. Right-click on the port entry, and select **Open**.
3. Under General information, ensure that **Enable use of port for** is enabled.
4. Under Enable use of port for, ensure that **Outbound dialing** is enabled, and select **OK**. When a call is made using the Outbound Dialer, the call displays as *Outbound Call* in the Ports content pane.

To view Outbound Dialer logs

1. Select the Logs navigation bar to open the Logs menu.
2. Select the Sent Items menu to view the Outbound Dialer call logs. The Outbound Dialer call logs are listed in the content pane. The information for a specific log is shown in the preview or information pane of the Logs menu.

Creating a Custom Application for Outbound Dialing



To create the custom application requires scripting knowledge, and knowledge of databases would also be beneficial.

The Outbound Dialer can be used for a variety of purposes, whether to convey information only, or to input responses into a database. The custom application you create, and the service created to run when the call connects, can be as simple or as complex as you wish. How you create the custom application, script, or .NET application, and what you decide to include, depends on your particular business requirements. Ideally, the application would gather information from a database, such as the names and the phone numbers of the people to call. Your custom application must then initiate the OutDial.exe, which makes a call request to the UC Server to initiate the calls.



For best results, the Outbound Dialer should be used in combination with a database. While a database is not required, not having access to a database seriously limits the capabilities of this feature.

Locating the Outbound Dialer

The OutDial.exe is available in the following folder, if it was installed in the default location:

\Program Files\ADTRAN\UC Server\Bin

Using Outbound Dialer Parameters

The OutDial.exe is a command line utility that uses various required and optional parameters. The command line syntax is OutDial followed by one or more parameters. Each parameter is composed of a name and a value, with at least one space in between them. For example:

OutDial parameter_name1 value1 ...

Some parameters are required, while others are optional. An example of a command is shown below, and the optional parameters are surrounded by square [] brackets. The parameters are shown on separate lines for ease of reading only.

OutDial

[-u <username> -pw <password>]

-h <host>

-ph <phone number list>

[-p <prompt delay>]

[-d <delay between attempts>]

[-r <maximum attempts>]

-s <service>

[-pm <parameters>]

[-? |-help]



The default prompt delay between when the call connects and when the service runs, is 1 second. The default delay between call attempts is 3 minutes, and the default maximum number of call attempts is 3.

Parameters can be specified in any order. For example, all of the following examples are valid, as they have the required host, phone number, and service parameters, and all would have the same result.

```
OutDial -h somewhere.com -ph 6135551212 -s myService
```

```
OutDial -ph 6135551212 -s myService -h somewhere.com
```

```
OutDial -s myService -ph 6135551212 -h somewhere.com
```

The command line syntax follows normal Windows command line rules. For example, double quotation marks ("...") are required around any values that include a space () character in those values, such as might be found in the username, host, phone number list, or service parameters. For example, **Admin** would not require double quotation marks, but "John Smith" requires double quotation marks because of the space between the first and the last name.

The following is a list of required parameters:

Table 13–1: Required Parameters

Parameter	Value	Description
-h	IP Address/Host Name	The IP address or host name where the server is located.
-ph	Phone Number List	This is a comma separated list of phone numbers to be dialed. Each phone number can be specified as a sequence of digits, for example, 5551212, or using standard formatting characters, for example, 555-1212, "(613) 555-1212", or "+1 (613) 555-1212". Numbers that have no formatting characters are dialed as is. Other numbers have the standard dialing rules applied, as defined in the Admin options (Tools > Options > Dialing). For example, the default area code can be added. Account codes can also be used if they are configured on the user profile associated with the request.
-s	Service Name	This refers to the UC Server service to run when a call is completed. The value must be the name of a service that belongs to the owned user profile for the authentication that the utility logs in as. The service name should be a full path name including any folders the user has defined. The path is assumed to start from the Services folder, and thus only any subfolders and the service name need to be included in the path. The service path should be from the user's UC Server Client. For example: OutService Shared/OutService2 "My Folder\Some Service" Note: Forward or reverse slashes may be used.

The following is a list of optional parameters:

Table 13–2: Optional Parameters

Parameter	Value	Description
-u	Login Name	Valid UC Server login name. If this parameter is specified, the -pw parameter (login password) must also be specified. Specifying -u and not specifying -pw causes an error. Both values, -u and -pw , are used to login to the server. If both parameters are omitted, the utility attempts to connect using single sign-on.
-pw	Login Password	Valid UC Server login password. If this parameter is specified, the -u parameter (login name), must also be specified. Specifying -pw and not specifying -u causes an error. Both values, -u and -pw , are used to login to the server. If both parameters are omitted, the utility attempts to connect using single sign-on.

Parameter	Value	Description
-sd	Start Delay	The delay before the outdial request is executed. If this mode is not specified then the outdial request is executed immediately (or as soon as possible, depending on the number of requests in the queue). The default value is zero.
-sdu	Start Delay Units	The units used for the delay. This may be minute, minutes, hour, hours, day, or days. The default is minutes. The start delay is calculated from the time the request is entered. The request will not be processed before the time delay has elapsed, however it is possible that it may be processed somewhat later if the system is heavily loaded, for example, with many Active Message Delivery (AMD) requests.
-p	Pause in seconds before starting service	The number of seconds to wait after the call has connected, before starting the service. If the parameter is not specified, the value of the outgoing call prompt delay (from the Admin options, Tools > Options > Message Delivery) is used. On trunks with no true answer supervision, such as analog trunks, the UC Server can determine when a call is connected, but it cannot determine precisely when the call is answered. This delay can be used to ensure that the service is probably started after the call is answered. Note that using analog trunks with the Outbound Dialer is not recommended.
-d	Number of minutes delay before next attempt	When the server fails to connect to a specific phone number, the request is queued again. The -d option indicates the number of minutes delay before the request is attempted again. If this mode is omitted, the server uses the Active message delivery delay mode (Tools > Options > Message Delivery) for the user for whom the request is being processed.
-r	Number of retry attempts	The server tries each number in turn until a call connects. If all the numbers fail, the server tries the first number again and so on. This mode specifies how many times each number is called before the request is abandoned. For example, if the value of this mode is 2, then each number is attempted a maximum of two times. If the mode is not specified, then the server uses the Active message delivery retry mode (Tools > Options > Message Delivery) for the user for whom the request is being processed.
-pm	Additional Parameters List	This is a comma separated list of additional parameters that are passed to the service when it executes. These values can be accessed within a service definition as special Text values whose names are Parameter1, Parameter2, and so on. The first value in the list becomes the value of Parameter1 and so forth.

Parameter	Value	Description
-? or -help	List Parameters	This parameter requests the utility to list a description of valid parameters and their meanings.

Creating Outbound Dialing Services

The Outbound Dialer can be used from a command prompt to make outgoing calls, and to run a service when a call connects. You can also create a service that uses the Queue Outdial element to launch an outdial request after a specified period of time. The information in this section explains how to create an outbound dialing service to run when an outbound dialing request is made from a command prompt. For information on creating a service to *initiate* an outbound dialing request, refer to [Queue Outdial element on page 267](#).

The type of service you create to run when an outbound call connects can be as simple or as complex as you wish. What elements you choose to include in the service depend on your business requirements.

At a minimum, you may want to use the Play Announcement element to play a message for the person answering the call. For more information, refer to [Play Announcement element on page 229](#). You may also want to use some database elements so that the person answering can input a response into a database. For more information, refer to [Using Database Elements on page 300](#). For example, you may want the person to enter a PIN if it's important to ensure that a particular person receives the message. Alternatively, you may want the person to press a number to indicate a response, for example, "Press 1 to indicate that you have heard the message. Press 2 to repeat the message..."

You could also use the [Generate Tones element on page 266](#) for an outbound dialing service to enter DTMF tones, also known as touch-tone dialing. For example, the Generate Tones element could be used to enter DTMF digits to dial an extension number for a specific caller in an automated attendant, to access an external paging system, or to enter an account code.

You can use the [Queue Outdial element on page 267](#) to determine if a call is answered by an answering machine, so that the behavior of the service can be varied accordingly.

You may want to use additional parameters to customize the outbound dialer service. In the Service Editor, parameters are not visible by default so you need to modify the view to enable parameters.

Parameter	Value	Description
-pm	Additional Parameters List	This is a comma separated list of additional parameters that are passed to the service when it executes. These values can be accessed within a service definition as special Text values whose names are Parameter1, Parameter2, and so on. The first value in the list becomes the value of Parameter1 and so forth.

Parameters can be used with the following elements to create custom outbound dialing services:

- The **Equals** field in the Data Source Selector element (any element that deals with a database)

- The Assisted Transfer, Transfer Call, Deliver Messages, Notify Pager, and Send Fax elements can accept a parameter as a number to call.
- The Send email element can accept parameters for email addresses.
- The Text To Speech element can read parameters to the person called.
- The Compare Data element can compare a parameter against several values and route the service based on the outcome.
- The Create Log Entry element can write parameters into the log files.

Note that parameters can be used to obtain a specific record from a database.

To use parameters in defining a service for an Outbound Dialer request

1. From the Services navigation bar, right-click in the content pane, select **New** and **Service**.
2. Select the service, right-click and select **Open** to open the Service Editor.
3. In the Service Editor, select **View > Parameters** to enable the Parameters. Once Parameters are enabled, then parameter names are available for selection.

When an Outbound Dialer call connects, and the service selected runs, it initializes the values of the parameters based on the parameter values in the call request. This uses the optional `-pm` parameter, which refers to any additional parameters. Specific values for the parameters can be provided, and system values are filled in as usual. The values of Called-ID and Called-ID Number are filled in with the called number. The Called-ID Name is blank, however if the service requires the name of the person being called, then it can be passed as a parameter in the request.

If an error results from running the service, then the element exits from the Otherwise pin.



If values are not provided for one or more parameters, then their default value is blank.

The server knows when a service has started running, however it does not know whether the service was run successfully, for example, whether the person listened to the message before hanging up. Additional parameters can be used to determine whether a service was successful. For example, the person answering the call might be required to press a number to indicate that they have received the message.

Processing Outbound Call Requests

The custom application initiates the OutDial.exe, which is a command line utility. The OutDial.exe logs in to the UC Server using the authentication provided, or using single sign-on. The OutDial.exe initiates a call request to the UC Server, according to the parameters specified. The utility sends a request to the server identifying the service, the phone numbers to call, and any additional parameters provided. The call request is then placed in the Outbound Call Queue.

When a call request is added to the call queue, then an Outbound Call Processor is created for it. The Outbound Call Processor calls the first number in the list.

- If the call is answered, then the service runs.

- If the call is not completed, then the next number is called.

This process continues up to a specified number of attempts before the request is deleted and a log entry is created.

Before the calls are made, there are a number of intervening steps. First, the command line parameters must be verified. When the OutDial.exe is initiated, it scans the command line parameters to ensure that the required parameters are included, and that the parameters are properly formatted. It verifies that both the -u <username> and -pw <password> parameters are provided for authentication, or that both are omitted for single sign-on. It also verifies that the phone numbers in the list are properly formatted. If any errors occur, they are noted in a written error message, and the OutDial execution ends.

If the parameters pass the validation, then the utility attempts to connect to the server and login. The connection uses the -h parameter. If the connection succeeds, the login is done using the provided username and password, or using single sign-on. If any errors occur, they are noted in a written error message, and the OutDial execution ends.

If the login is successful, then the utility sends the call request to the server and waits for confirmation. The server verifies the following information before validating and queuing the request:

- That the service specified exists
- That the user profile for the login authentication is a system user
- That the user is authorized to perform this operation

If any errors occur, for example the utility times out waiting for confirmation, or the server returns an error, then these are noted in a written error message, and the OutDial execution ends. If the server receives a request that matches an existing queued request, then the request is rejected as a duplicate. Two requests match if they are for the same user, service, parameters, and phone numbers. If there are no errors, then the utility writes a message summarizing the request, and the OutDial execution ends.

When the OutDial execution ends, the utility returns a numeric return code that could be used by a shell or batch command processor. If the execution is successful, the utility returns 0. Unique non-zero return codes are returned for the situations below, and any other specific error returns from the server:

- Failed to connect to server
- Failed to login
- No response from request to server
- Duplicate request
- Invalid parameters

If the outbound call request passes the server's validation, then the call request is processed, and the following actions occur:

- An attempt is made to call the first phone number in the list.
- If the call succeeds, then the service is executed. The execution of the service may be delayed to ensure the call is answered, using the -p parameter, or the Outgoing call control, prompt delay (Tools > Options > Message Delivery).

- If the call is not completed, the request is re-queued for the number of minutes indicated in the -d parameter, or in the user's Active message delivery Delay mode, (Tools > Options > Message Delivery). The next phone number on the list is then attempted. Calls that were not initially successful are retried after the period of time has elapsed. The call request is attempted until the maximum number of attempts have been made, as per the -r parameter, or the Active message delivery Retry mode.
- If a call request is not completed successfully after the appropriate number of call attempts have been made, then a log entry is written, and can be viewed in the UC Server Client. The request is then removed from the queue.
- When a server receives an outbound call request, this request is copied to disk as well as being entered in the processing queue. When a request is removed from the queue, because it has been completed, or because it has not been able to execute the service on any of the phone numbers given, the copy of the request on the disk is removed. When a server starts up, it reads any requests available and inserts them into the queue for processing. This ensures that if a server is restarted, any existing call requests are continued.

A call request is considered processed if a call connects, and the service begins to run. This means that the request is not re-queued, even if errors are subsequently detected, for example, the person hangs up before the service has finished running.



Server activities related to outbound dialing are noted in the logs. In particular, any error conditions, failure to complete a call, and successful initiation of the service should be noted in the logs.

Outbound Dialer Return Codes

Once the OutDial.exe has successfully logged in to the server, then it sends the call request to the server and waits for confirmation. The server verifies the information given before validating the request and queuing the request. When the OutDial execution ends, the utility returns a numeric return code that could be used by a shell or batch command processor. If the execution is successful, the utility returns 0. Unique non-zero return codes are returned for the situations below, and any other specific error returns from the server. The following is a list of the UC Server return codes, and contains all the errors that can be reported by OutDial.



Server activities related to the Outbound Dialer are noted in the logs. In particular, any error conditions, failure to complete a call, and successful initiation of the service should be noted in the logs.

Table 13–3: Outbound Dialer Return Codes

Return Code	Description
0	Success
1	Unknown failure
2	Unsupported command
3	Communication error
4	License key error
5	Invalid number to dial
6	Specified service not found
7	Subsystem not available
8	Service fails validation
9	Failed to activate service
10	Failed to open line
11	System busy
12	Permission denied
13	Nothing to do
14	Invalid object
15	Invalid parameters
16	Invalid path
17	Not logged in
18	Data store failure
19	File already exists
20	Mailbox list is empty
21	Object does not exist
22	User not available to the server
23	Invalid password
24	Client using an out-of-date version
25	PBX program error
26	Not licensed or license limit exceeded
27	Cannot delete active service
28	Still in use
29	Authentication has been disabled

Querying Outbound Call Requests

You can use the OutDial.exe to query any outstanding outbound call requests, using the command line shown below. The utility then connects to the server and logs in as usual. A request is sent to the server that returns a summary of each outbound call request that has not yet been processed, that is, the request is still in the queue for the given user. The utility returns a message describing each request, including the time it was submitted, the service, the parameters, and the number of attempts so far.

To query an outbound call request

1. Locate the OutDial.exe. It is available in the following folder, if the UC Server was installed in the default location:

```
\Program Files\ADTRAN\UC Server\Bin
```

2. From the OutDial command line utility, enter the following command line:

```
OutDial -q[-u <username> -pw <password>] -h <host> [-? | -help]
```

The username, password, host, and help parameters function as they do when running the OutDial utility. In the command above, the **-q** queries the queued outbound call requests for that user. The optional parameters appear in the square [] brackets. The parameters may be used in any order.

Canceling Outbound Call Requests

You can use the OutDial.exe to cancel outbound call requests, using the command line shown below. The utility then connects to the server and logs in as usual. The server looks for a queued call request that matches the specified parameters, that is, the phone number list, service, and parameters must match exactly. If a matching request is found, then it's removed from the queue, and a log entry is created indicating the reason. If an entry is found that has already started, where dialing is in progress, then the request cannot be canceled and an error code is returned to the client.

To cancel an outbound call request

1. Locate the OutDial.exe. It is available in the following folder, if the UC Server was installed in the default location:

```
\Program Files\ADTRAN\UC Server\Bin
```

2. From the OutDial command line utility, enter the following command line:

```
OutDial -c
```

```
[-u <username> -pw <password>]
```

```
-h <host>
```

```
-ph <phone number list>
```

```
-s <service>
```

```
[-pm <parameters>]
```

```
[-? | -help]
```

The parameters listed function as they do when running the OutDial utility. In the command above, the **-c** cancels the outbound call request. The parameters may be used in any order, and the optional parameters are shown in square [] brackets. The parameters are shown on separate lines for ease of reading only.

Customizing System Announcements on the Server

The UC Server is configured with system announcements, which are professionally recorded audio files used within the application, and play to the user over the telephone. System announcements are general system prompts that are used throughout the application, for example, for end-of-menu prompts.

However, if you want to change the system announcements, you can customize the announcements file on the server. The changes that you make are maintained during UC Server software upgrades.

There are two files on the server that the UC Server uses as defaults for system announcements: Male Voice and Female Voice. You can customize one or both of these files. If you want the files that you add to the custom directories to play in place of the default system announcements, make sure that the file has the same name. For example, the default **MM_DefaultGreeting.wav** male voice file is replaced by the **MM_DefaultGreeting.wav** file, if you copy it to the “CustomAnnouncements” male voice folder.



Customizing system announcements is an advanced feature. It is recommended that you do not carry out this procedure unless you receive further instruction from ADTRAN.



If the format of the audio file is incorrect, the new file is not heard. Make sure that the audio of the new file is recorded in G.711 mu-law format, 8-bit samples, 8 kHz, and Mono.

To customize server male voice system announcements

1. Navigate to the following file:
Program Files\ADTRAN\UC Server\Data\System\CustomAnnouncements\Male Voice
2. Change the male voice system announcement that you want.

To customize server female voice system announcements

1. Navigate to the following file:
Program Files\ADTRAN\UC Server\Data\System\CustomAnnouncements\Female Voice
2. Change the female voice system announcement that you want.

14 Creating Custom Call Answering Services

This chapter describes call answering services, managing services, and building services using the Service Editor. This chapter covers the following topics:

- [Overview on page 213](#)
- [Managing Services on page 214](#)
- [Using the Service Editor on page 217](#)
- [Working with Elements to Build a Service on page 219](#)
- [Customizing Services using Parameters on page 222](#)
- [Validating Services on page 223](#)
- [Troubleshooting Services on page 223](#)

Overview

A service is like a map that the UC server follows when it answers a call. Services are made up of elements that are linked together to create a call flow. You can design simple services made up of two or three elements, or you can create complex call flows made up of a dozen elements or more. Use the Service Editor to build a service.

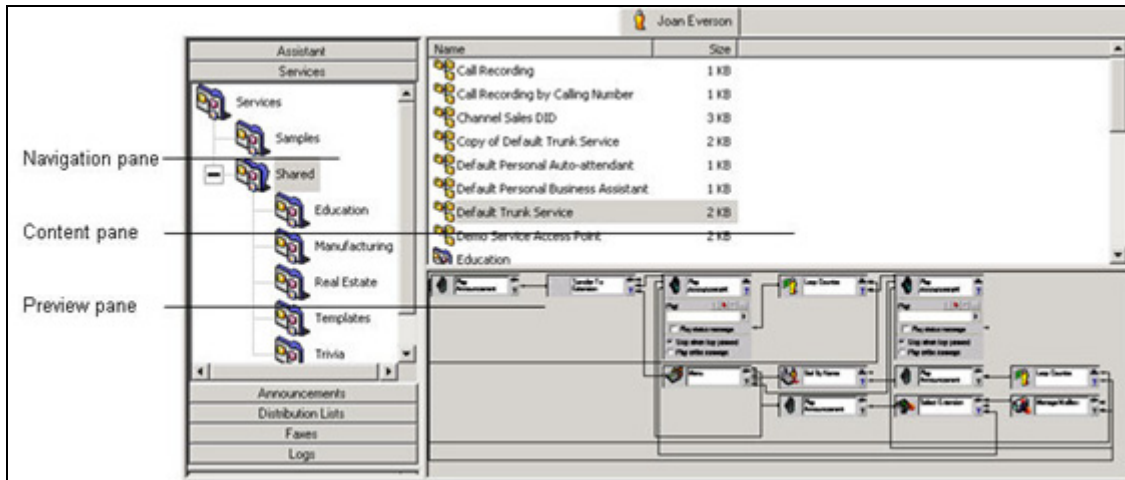
You can build a service using the Service Editor and then assign it to an identity so that incoming calls follow the path that you specify. You can create a new service for the purposes of setting up a 1-800 line, a sales information line, or a line associated with a menu of options that callers can select to obtain specific information. Likewise, you can set up a fax-back service and activate it on an identity.

For information about designing and changing services, refer to [Using Elements to Create Call Answering Services on page 225](#). Services can be renamed, copied, organized in folders, and deleted. You can also share services using the Import/Export feature. For information about the Import/Export feature, refer to [Importing and Exporting on page 311](#).

Accessing Services

To create a service or change information associated with it, from the Administrator application:

- Select the **Services** navigation bar to open the Services menu. Services are displayed in the content pane.



Managing Services

You can create a service, and then you can preview, activate, rename, copy, or delete that service. You can create and manage the folders in which you save your services.

NOTE *ADTRAN has provided some samples services to get you started. Before creating a new service, preview the samples provided to see if a sample service exists that is similar to what you are looking so. If you find a similar service, copy it, rename it, and then modify it suit your needs.*

NOTE *Many of the sample services provided by ADTRAN use database integration. If your UC server version is not licensed for database integration then the database elements used in the sample services will not work as intended.*

To create a service

1. Right-click anywhere in the content pane.
2. Select **New > Service** to add a new service to the list in the content pane. The service icon denotes the new service and the default name is “Service.”



Service icon

3. Type a name for the service.

NOTE *You can also right-click the service and then select **Rename**. Type the new name again, and then select the icon next to the service name to make the change.*

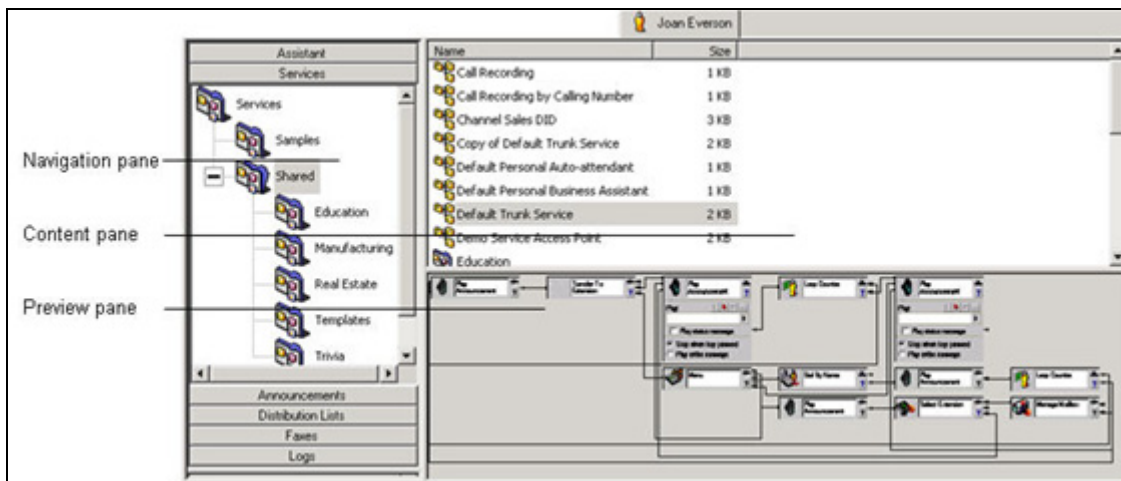
Working with services

When you select the Services navigation bar, the services you create are listed by name in the content pane. You can use the Service Editor to view a service and change it. If you want only an overview of a service, you can preview it in the Services menu.

To preview a service

- In the content pane, select the service that you want to preview.

If the service is in the cache, it appears immediately in the preview pane. Otherwise, select the **Download** button in the preview pane to get the preview.



To activate a service on an identity

1. Right-click the service that you want to activate, and select **Activate Service**.
2. In the **Activate on** section, enable the check box of the identity, or identities, to which you want to assign the service. The **Already activated on** section allows you to see on which user identities the service is already active.
3. Select **OK**.



The service must already be created and validated before activating it.



Once you have activated a service on an extension, you cannot “deactivate” that service. To change the service, activate another service on the extension.

To rename a service

1. Right-click the service that you want to rename.
2. Select **Rename**.

3. Type the new name for the service.
4. Select the icon next to the service name to make the change.

To copy a service

1. Right-click the service that you want to copy.
2. Select **Duplicate**. The copy of the service appears in the content pane, with the words “Copy of” before its name.
3. Rename the copy of the service by following all the steps in [To rename a service on page 215](#).

To delete a service

1. Right-click the service that you want to delete.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.



You cannot delete a service that is active on an identity.

Creating and changing service folders

You can create and manage folders in much the same way as services, and you can use folders to organize your services. For example, you can create a folder for each of your identities and then store associated services in it.



Only one service can be active on an identity at any time.

When you want to change the active service on an identity, you can quickly do so if related services are kept in the same folder. You might have built some services that you are not currently using, for example, those designed for statutory holidays. Store these together in a folder called “Holidays,” and then retrieve them when you need them.

To create a folder

1. Right-click anywhere in the content pane.
2. Select **New > Folder** to add a new folder to the list in the content pane. The folder icon denotes the new folder and the default name is “New Folder.”



Folder icon

3. Type a name for the folder.



*You can also right-click the folder and then select **Rename**. Type the new name again, and then select the icon next to the folder name to make the change.*

To rename a folder

1. Right-click the folder that you want to rename.
2. Select **Rename**.
3. Type the new name for the folder.
4. Select the icon next to the folder name to make the change.

To copy a folder

1. In the content pane, right-click the folder that you want to copy.
2. Select **Duplicate**. The copy of the folder appears in the content pane, with the words “Copy of” before its name.
3. Rename the copy of the folder.
4. Select the icon next to the folder name to make the change.

To delete a folder

1. Right-click the folder that you want to delete.
2. Select **Delete**.
3. Select **Yes** to confirm the deletion.

To store a service in a folder

- Drag the service to the folder where you want to store it.

Using the Service Editor

To access the Service Editor

From the Administrator application:

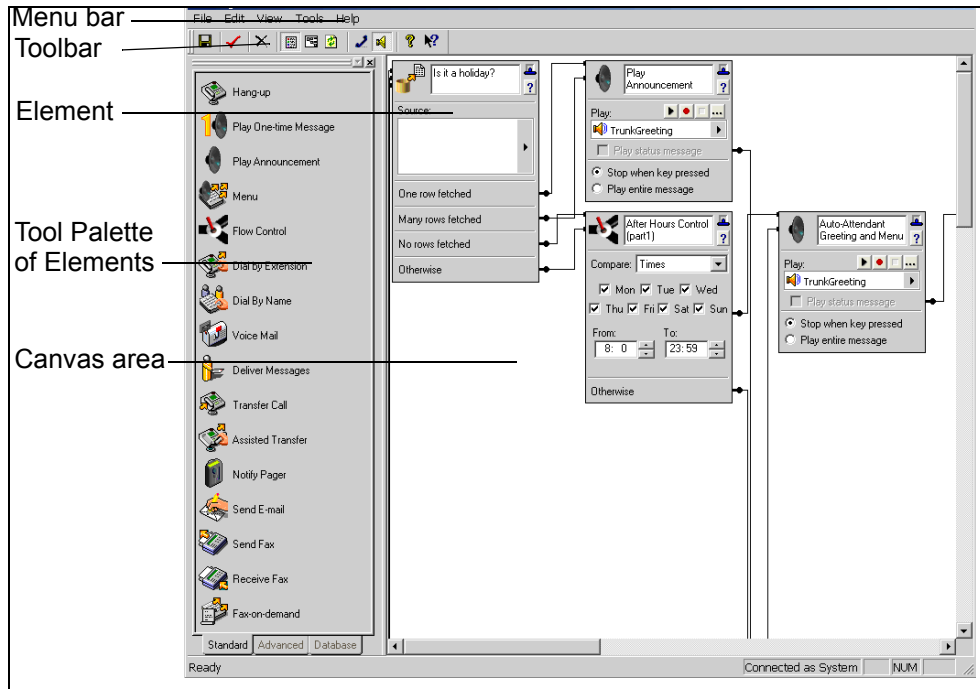
1. Select the **Services** navigation bar to open the Services menu. Services are displayed in the content pane.

- To open the Service Editor, right-click anywhere in the content pane. In the shortcut menu, select **New > Service**. The newly created service is listed in the content pane. Double-click the service and the Service Editor opens.



Alternatively, you can open the Service Editor by double-clicking an existing service.

The Service Editor



The Service Editor consists of a standard Menu Bar and Toolbar, a Tool Palette of Elements, and a Canvas area. You design a service on the canvas using a combination of elements that you choose from the tool palette.

The **Menu Bar** contains the File, Edit, View, Tools, and Help menus. Use these menus to manage the service design.

The **Toolbar** contains visual shortcuts to the most frequently used menu items. Hold the pointer over a button to show the tool tips.

The **Tool Palette** contains all available elements. It has Standard, Advanced, and Database element tabs. You can dock the Tool Palette to either side of the window.



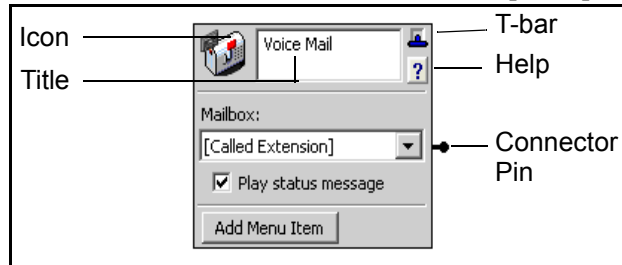
To access database information, the system must be licensed for database integration. If you are not licensed for database integration, the Database tab in the Service Editor is unavailable.

The **canvas** is the design area. All elements and links are shown here. Each element performs a separate

function within the call flow. Connections between elements are referred to as links, and they define the call flow.

To view an element

- Drag an element from the tool palette to the canvas. The element is automatically positioned in the top-left corner. Each Element has an icon, title area, T-bar control, help button, and a connector pin associated with it. Some elements have more than one connector pin depending on their function.

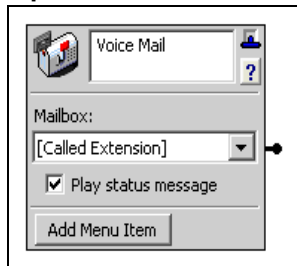


The **Icon** is a visual representation of the element type.

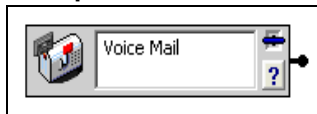
In the **Title** area you can type a description of the element's function.

The **T-bar** control displays the properties associated with the element. To save canvas space, select the T-bar control to collapse the element as needed. To restore information about a collapsed element, select its T-bar control. The three modes of display are illustrated below.

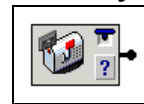
Open element



Collapsed element



Icon only



Select the **Help** button to access the appropriate help page.

The **Connector Pin** on the element is used to make a connection to the next element in the call flow. An element might have more than one connector pin, depending on its function. If you do not connect at least one pin to another element, the call is terminated.

Working with Elements to Build a Service

To build a call flow for incoming calls, you need to create a service and then open the Service Editor. To begin the call flow, drag elements one by one from the tool palette and link them using the connector pins. Repeat this procedure with as many elements as you need until the service is complete.

To create a service

1. Open the Services menu.
2. Right-click anywhere in the content pane.
3. Select **New > Service** to add a new service to the list in the content pane. The service icon denotes the new service and the default name is “Service.”



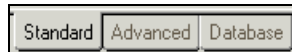
Service icon

4. Type a name for the service. If you are unable to type the name, right-click the service and then select **Rename**. Type the new name, and then select the icon next to the service name to make the change.

To initiate a call flow for a new service

1. Double-click the new service to start the Service Editor.
2. Drag an element from the tool palette to the canvas to start the call flow. The element is automatically positioned in the top-left corner.

The Service Editor's default menu displays the Standard elements tab. To view advanced or database elements, select the Advanced or Database tab located at the bottom-left corner of the menu.



The Database tab is available only if the system is licensed for database integration.

To add an element to the call flow

1. Right-click the connector pin of the first element that you put on the canvas, and then select **Insert**.
2. In the shortcut menu of standard elements, select the element that you want to add. The element is linked to the first one that you put on the canvas.



To access the shortcut menu of advanced or database elements, select Advanced or Database on the standard elements shortcut menu. The Database tab is available only if your system is licensed for database integration.



You can also drag the element that you want from the tool palette to the canvas. Position the pointer over the connector pin of the existing element until a red box appears around the connector pin, and then drop the new element onto the canvas.

Build a service by adding elements until the call flow is complete.

Using and deleting links

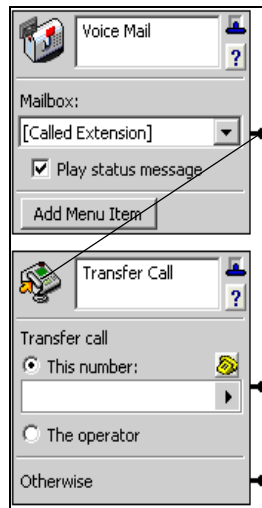
If you link two elements and then change your mind, you can delete the link and reuse the element in the

service.

To delete a link between two elements

1. Right-click the connector pin linking the two elements that you want to separate.
2. Select **Delete link** to delete the link.

When you delete a link between two elements, the elements are no longer connected on the canvas. At first glance, one of the elements seems to have been deleted because it disappears from the place it occupied in the call flow. The element is repositioned at the bottom-left corner of the canvas. From there, you can connect it elsewhere in the call flow by linking it with another existing element.



To link existing elements

1. Select and drag from the connector pin to the icon of the element that you want to link.
2. Release the mouse button to link the elements. If you link an element and then decide that you no longer want to include it in the call flow, you can delete the element from the canvas.

To delete an element from the canvas

1. Select the icon of the element that you want to delete to highlight the element.



Delete

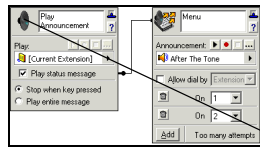
2. Select Delete on the toolbar to delete the element from the canvas, or press the Delete key.

Creating loop-back links

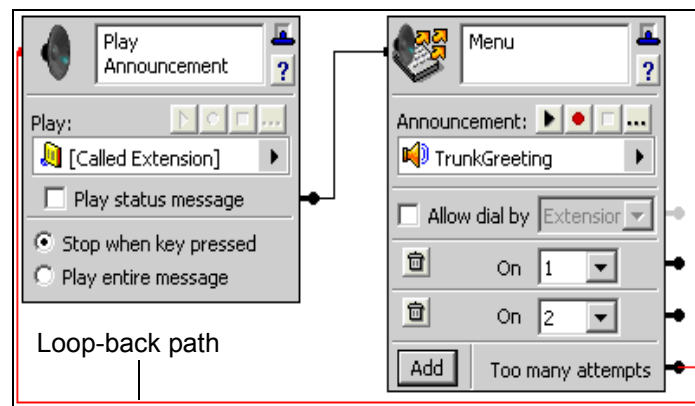
A loop-back link returns a call to a preceding element in the call flow. Loop-back links are useful for repeating a set of menu options or for restating instructions to a caller who has failed to make a selection on the phone keypad following a prompt.

To create a loop-back link

1. Select the connector pin and drag to the element icon that you want to loop back to.



2. Release the mouse to create the loop-back link.
3. Point to the connector link. The loop-back path is highlighted in red.



Customizing Services using Parameters

You can use additional parameters in order to customize services. Parameters can be used in most elements that accept input, such as email addresses, log entries, transfer numbers, or database comparisons. Parameters can also be used in the **Equals** field in the Data Source Selector. For more information, refer to [Using the Data Source Selector on page 297](#).

Parameters can be used with the following elements:

- The Assisted Transfer, Transfer Call, Deliver Messages, Notify Pager, and Send Fax elements.
- The Send Email element.
- The Text To Speech element.
- The Compare Data element.
- The Create Log Entry element.
- The Select Extension.

Note that parameters can be used to obtain a specific record from a database.

Parameter	Value	Description
-pm	Additional Parameters List	This is a comma separated list of additional parameters that are passed to the service when it executes. These values can be accessed within a service definition as special Text values whose names are Parameter1, Parameter2, and so on. The first value in the list becomes the value of Parameter1 and so forth.

In the service editor, parameters are not visible by default. Once enabled, you can select parameters for the elements that use them.

To view parameters

1. From the Services navigation bar, right-click in the content pane, select **New > Service**.
2. Select the service, right-click and select **Open** to open the Service Editor.
3. In the Service Editor, select **View > Parameters** to enable the Parameters. Once Parameters are enabled, then parameter values are available for selection.

Validating Services

After you finish building your service, you can validate it. The Service Editor inspects the service for logical errors or unresolved aspects of the call flow. If there is a problem with your service design, a message box appears with information about the changes that you need to make.

To validate a service

1. On the toolbar, select the check mark.



Check mark

2. If the Service Editor message informs you that the service is valid, select OK. Otherwise, follow the Service Editor's suggestions about how to change your service to make it valid.

Troubleshooting Services

If you are having trouble running services, note that particular issues may be noted in the log files. The following is a list of potential service warnings that may be noted in the System Errors and Warning logs. To troubleshoot a warning, locate the log entry below, and note the possible causes of and solutions for the

warning.



Not all errors and warnings necessarily require action. For more information on logs, refer to [Using the NetVanta UC Server Logs on page 320](#).

To view service warning logs

1. Select the **Logs** navigation bar to open the Logs menu. All logs are presented in the Content menu.
2. If the logs are not already sorted into categories, then double-click the **Logs** folder.
3. Select **System Errors and Warnings** to view the logs in that category.
4. Verify whether either of the following entries appear.

Log Entry

Possible Causes and Solutions

Failed to update %s's database with the received fax. The fax will be saved instead to the default mailbox.

There was a problem communicating with the database defined in the Receive Fax element.
Check your database connection parameters, and ensure that the database is available.

The data source, %s, referenced in the element, %s, is unknown.

ServiceStateMachine when an element refers to a data source that no longer exists. Verify the following:
 1.The ODBC data source, if applicable, is correctly configured
 2.The fields in the reference data source may have been changed or deleted. Verify the data source configuration.

15 Using Elements to Create Call Answering Services

This chapter presents information about the individual elements that you use to build a service. This chapter covers the following topics:

- [Overview on page 225](#)
- [Using Standard Elements on page 228](#)
- [Using Advanced Elements on page 248](#)



If your system is licensed for database integration then you will also have a Database elements tab available. For more information, refer to [Using Database Elements on page 300](#).

Overview

Elements are the building blocks of the services that you create for identities. By combining elements on the canvas of the Service Editor, you design a call flow that routes incoming calls.

Elements are on the tool palette of the Service Editor. They are divided into three groups: standard, advanced, and database. Each group has its own tab.



To access database information, the system must be licensed for database integration. If you are not licensed for database integration, the Database tab is unavailable. For more information, refer to [Using Database Elements on page 300](#).

User context and service elements

When you start a service, it begins by operating in the context of the user who owns the identity (for example, the extension) that received the call. The different service elements access mailboxes and objects that are related to this user context. You can change the user context by modifying the service.

The service user context is characterized by an authentication and a user profile (that is accessible from the authentication).

How elements affect user context

A service is associated with an identity. When a service starts, the identity is used to locate a user profile (the one that owns the identity). The user context is then established with the user profile that owns the identity and the authentication that owns that profile.

Elements cannot change the user context. In particular, Transfer Call, Dial by Name, Dial by Extension, and Select Extension, do not change the user context. Select Extension causes the selected extension to be saved in the current call object so that it can be used by other elements.

Synchronizing contacts

The Synchronize contacts mode is only available to users who have a client-side contact database, where contacts are stored on a user's computer, and not on a server like Microsoft Exchange. Examples of client-side contact databases include Microsoft Outlook (running in **Internet Only Mode**), Outlook Express, and Windows Address Book.

The UC server cannot access contacts that are stored in these client-side contact databases, so in order to perform real-time contact matching this contact information must be copied and stored on the UC server. Client-side contacts are synchronized with the information stored on the UC server every time a user starts the UC client and when the UC client is shut down. If a user makes changes to their client-side contact database while the UC client is open, and they want the UC server to recognize these changes without having to shut-down and restart the UC client, the user can force a contact synchronization by using the Tools > Synchronize contacts menu mode.

To synchronize contacts from a user's computer to the NetVanta UC Server

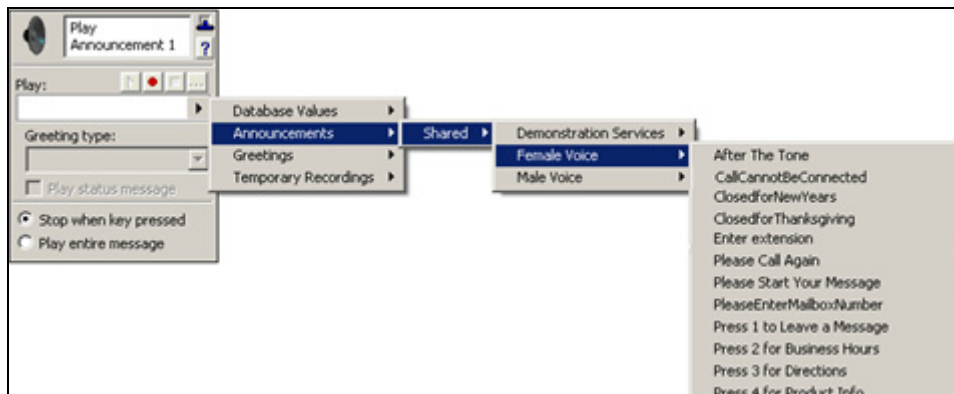
Select the **Tools** menu > **Synchronize contacts**.

Selecting values within elements using the value selector

Use the Value Selector feature, which is available on some elements, to specify the following types of values:

- Literal values** For example, a number or piece of text
- References to data files** For example, a fax, announcement, or greeting
- References to database values** References that have been fetched from a database
- System values** For example, caller ID
- Strings of digits** Gathered by a Gather Digits element

For example, the Play Announcement element allows you to play a dynamic value, announcement, greeting, or temporary recording to a caller that encounters the element in a service.



The values that are available using the Value Selector depend on how your service and system are set up.

For example, to use database values, a database element must be included in the service. To specify Gather Digits values, a Gather Digits element must be included in the service.



To use database values and elements, your system must be licensed for database integration.

Allowing callers to make temporary recordings within a service

Some elements allow you to select audio files such as announcements, greetings, database values, or temporary recordings. Using the Record Announcement element, a caller can record audio into a temporary recording, and elements like Play Announcement can be used to play the audio.

Temporary recordings exist only for the duration of the service. That is, when a service starts, no temporary recordings exist. If a caller makes a temporary recording, the recording is stored in a temporary file that can be played by another element. All temporary recordings are discarded when the service ends. Up to ten temporary recordings can be created.

Temporary recordings can be referenced in the following elements:

- Play Announcement element
- Menu element
- Send Email element (custom)
- Record Announcement element
- Page Recorded element
- Add Data element
- Update Data element



















In order to use the Add Data and Update Data elements, your system must be licensed for database integration.

Temporary recordings can be used with the Page Recorded element for a caller to review and confirm a page announcement before playing it over a paging system. Temporary recordings can be used with a Send Email element, or Add Data and Update Data elements to record audio for form filling applications, such as requesting quotes, or answering general customer service questions.

Using Standard Elements

The standard elements tab is shown to the left. A brief description is provided for each element available.

Table 15–1: Standard elements

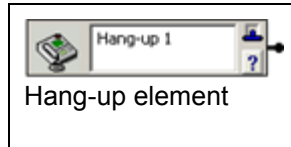
 Hang-up	Hang-up element on page 229	Terminates the call
 Play One-time Message	Play One-time Message element on page 229	Allows you to leave a personalized message for a specific caller
 Play Announcement	Play Announcement element on page 229	Plays a prerecorded announcement, greeting, temporary recording, or audio database file to callers
 Menu	Menu element on page 232	Allows callers to make a selection using the telephone keypad
 Flow Control	Flow control element on page 233	Controls the call flow according to phone number or time and day restrictions.
 Dial by Extension	Dial by Extension element on page 235	Allows callers to select an extension and be transferred to it
 Dial by Name	Dial by Name element on page 235	Enables callers to dial and be transferred to a user's extension by entering the user's name
 Voice Mail	Voicemail element on page 236	Provides basic voicemail service
 Deliver Messages	Deliver Messages element on page 237	Delivers voice or fax messages from a selected mailbox to a specified number
 Transfer Call	Transfer Call element on page 238	Transfers the call to another number or extension
 Assisted Transfer	Assisted Transfer element on page 239	Allows call transfer recipients to accept or deny calls based on the caller's recorded name
 Notify Pager	Notify Pager element on page 240	Enables you to send a message to a pager number
 Send E-mail	Send Email element on page 241	Allows you to be notified by email when a caller moves successfully through the Send Email element
 Send Fax	Send Fax element on page 243	Allows you to send a list of faxes
 Receive Fax	Receive Fax element on page 244	Receives a fax and places it in a mailbox
 Fax-on-demand	Fax-on-demand element on page 247	Captures a caller's number and faxes the selected document to a caller



The Dial by Extension and Dial by Name elements are available only to Administrators, and not to Personal Business Assistant (PBA) users.

Hang-up element

The Hang-up element terminates the current incoming call. If a call is present, this element hangs up. The same result occurs if a call flow leads to an unconnected connector pin.

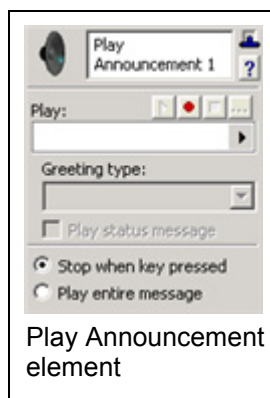


Play One-time Message element



Including the Play One-time Message element in a service allows you to leave a personalized message for a specific caller. Once a message has played, the call flow follows the “Message played” connection. Calls for which there is no customized message follow the “No message for current caller” connection. Enable the **Stop** when key pressed button to stop the message when the caller presses a key. Enable the **Play entire message** button to make sure that the caller listens to the message in its entirety.

Play Announcement element



The Play Announcement element plays prerecorded audio to callers including audio files stored in a database, announcements, greetings, or temporary recordings. You can select one of the prerecorded announcements included with the UC server, or one that you recorded. Alternatively, you can record one as

you build the call flow.



Make sure that there is a sound card installed on the computer that you are using.

Play

Database Values

Supports playing audio stored in a data source. The audio must first be fetched using a [Fetch Data element on page 301](#), or a [Dial for Data element on page 303](#). Database values come from a data source element and are not directly affected by user context.

Note: Your system must be licensed for database integration in order to use this mode. In the Service Editor, if you have the Database tab of elements available, then your system is licensed for database integration.

Announcements

Announcements come from the list that belongs to the user profile in the user context. For more information, refer to [User context and service elements on page 225](#).

Greetings

You can select [Called Extension], , or a specific user's greeting. The [Selected Extension] mode is available to Administrators only. For more information, refer to the [Select Extension element on page 249](#). If you select Greetings, then select a Greeting type, as described below. If the Greeting selected is [Called Extension], it is advisable to include a Play Announcement element before it is in the call flow, because the Administrator application has no greetings.

Temporary Recordings

Temporary recordings can be used to record audio temporarily, after a service has started. Use the [Record Announcement element on page 252](#) to record the audio for a specific temporary recording. The recordings are then stored in a temporary file, which can be played by another element that plays audio such as the Play Announcement element. All temporary recordings are discarded when the service ends. Up to 10 temporary recordings can be made. For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

Greeting type

The Greeting type is only enabled if **Greetings** was selected for the **Play** mode.

Active Greeting

This is the default greeting.

Internal Greeting

Internal greetings are intended for callers calling from extensions at your workplace.

External Greeting

External greetings are intended for callers calling from outside your workplace.

Out-of-Office Greeting Out-of-Office greetings are typically enabled when you are out of the office for an extended period of time.

Mailbox Identifier The mailbox identifier consists of your name. This greeting is typically used when callers are transferred to your extension.

The **Play status message** mode pre-empts the user's greeting message and cannot be interrupted.

- Enable the **Stop when key pressed** button to stop the message when the caller presses a key.
- Enable the **Play entire message** button to make sure that the caller listens to the message in its entirety.

To record an announcement for a Play Announcement element

1. On the Play Announcement element, select the record button.
2. In the Select Announcement dialog box, type a name for the announcement that you want to record.
3. Select **Select** to start the Audio Editor.
4. Select the record button and the Record Audio dialog box opens.
5. Select **at the beginning** from the **Insert new audio** list.
6. Select the icon for the computer audio device. The icon is enclosed by a dotted line.
7. Select the record button and then record the message. As you record, the clock runs and the red record button turns grey.
8. Select the stop button when you are finished.
9. Select **OK** to keep the announcement.
10. Select **File > Update**.

To play a greeting from a mailbox using a Play Announcement element

1. On the Play Announcement element, select the Value Selector arrow on the Play box.
2. Select **Greetings > Called Extension**. If you specify [Called Extension], the greeting is fetched from the user profile in the user context.

To enable the Play status message mode for greetings

- On the Play Announcement element, enable the **Play status message button**.



*Make sure that the **Play status message** mode is enabled on a corresponding voicemail element (refer to [Voicemail element on page 236](#)).*

To play temporary recordings

On the Play Announcement element, select the Value Selector arrow, select **Temporary Recordings**, and select one of the ten temporary recordings available. You must use the Record Announcement element in the service in order for a caller to record the audio to a temporary file.

Menu element



The Menu element allows you to make an announcement and then direct the call flow based on the DTMF input received when a caller presses a number on the telephone keypad.

This element can play an announcement, database value, greeting, or temporary recording before it takes the specified menu action. It retrieves announcements from the set of announcements that belong to the user profile in the user context. Database values come from a data source element and are not affected directly by user context. The service retrieves greetings from an explicit user, if such a user is specified. Temporary recordings are created using the Record Announcement element, to record the audio for a specific temporary recording after a service has started, which is discarded after the service ends. For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

Callers are allowed to try only three times to select from the menu. If they make another attempt, they follow the “Too many attempts” connector pin.

When callers press a selection key, they move through the connector pin to the next element, depending on their choice. When you are finished setting selection keys, you can connect an element for each selection key that you set.

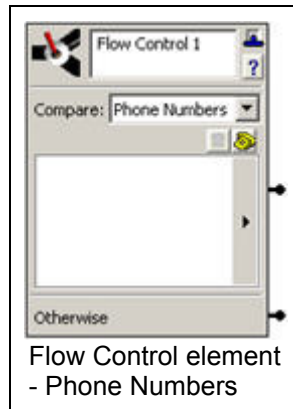
To set selection keys on a Menu element

1. In the first **On** list box, select the number that you want. The number one appears by default.
2. Select **Add** to add an On list box for setting additional keys. The UC server enters the key in numerical sequence. You can also select a key from the list. You can set a maximum of 12 keys. Keys range from one to nine, then include zero, *, and #.
3. To delete a selection key on a Menu element, select the trashcan icon to the left of the key that you want to delete. Remember to change the announcement associated with the menu so that it reflects the changes.



For information about recording an announcement for the Menu element, refer to [To record an announcement for a Play Announcement element on page 231](#).

Flow control element



The Flow Control element acts as a decision tree, and allows for two different comparison methods: matching based on phone numbers or on time of day and date. The call will be routed to the next element based on a successful or failed match (a failed match routes the call through the Otherwise connector pin). You can select either Phone Numbers or Times, which determines the comparison element.

For example, you could use the Flow Control element with a Receive Fax element to call your cell phone, send a pager notification, or send an email to your personal email address, to notify you when you receive a fax from a particular number after business hours.

Routing incoming calls based on phone numbers

The UC server compares the incoming caller ID to those telephone numbers included in the Flow Control element list. Telephone numbers in the list can be entered manually, or they can be pulled from your contact address book by using the value selector.

For more information, refer to [Selecting values within elements using the value selector on page 226](#). For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#).

The UC server compares the numbers of incoming calls with those included in the list.

- If the numbers match, the call is routed to the subsequent element that you specify.
- If the numbers do not match, the call moves to the element connected to the Otherwise connector pin.

To set call flow according to an unknown phone number category

1. On the Flow Control element in the Compare list, select **Phone Numbers**.
2. Select the Value Selector arrow.
3. Select **Contacts > Unknown**.
4. Select one of the following categories:
 - Unknown Number
 - Private/Blocked Number
 - Out-of-area Number

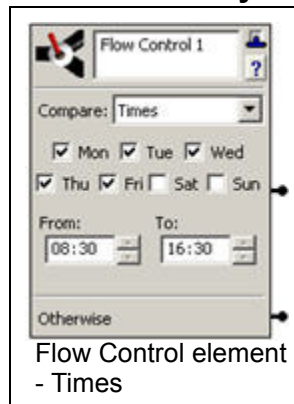
To enter a phone number manually

1. On the Flow Control element in the Compare list, select **Phone Numbers**.
2. Select the telephone icon to open the Enter Phone Number dialog box.
3. Type the phone number that you want in the Telephone number box, and select **OK**.



You can also enter a phone number by right-clicking the blank box and selecting New.

Routing incoming calls based on times and days



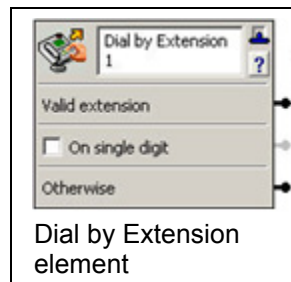
You can specify days of the week and a time range. For time restrictions, if either the From or To box is empty, a 24-hour period is assumed.

- Calls received within the specified time range are directed through the connector pin for Times.
- Calls received outside the specified time range move to the element connected to the Otherwise connector pin.

To set call flow according to time restrictions

1. On the Flow Control element from the Compare list, select **Times**.
2. Enable the check boxes for the days of the week when the restrictions apply.
3. From the **From** list, select the hour and minutes when the time restriction begins. Use a 24-hour clock.
4. From the **To** list, select the hour and minutes when the time restriction ends. Use a 24-hour clock.

Dial by Extension element



The Dial by Extension element accepts DTMF tones that a caller enters for an extension. If the tones match a valid extension, the call is transferred to the specified extension, and the Selected Extension and Selected User Name system values are changed to the extension and the user name owning that extension. Since the call has been transferred, no call is present after leaving the “Valid extension” connection. The valid extension is a user account as well as any extensions defined in the extensions tab of the options menu.

If you enable the On single digit check box, you can connect an Advanced Menu element to provide single-digit entry processing for multi-function entry. If a caller enters an invalid extension, the call leaves the element by the Otherwise connector pin.

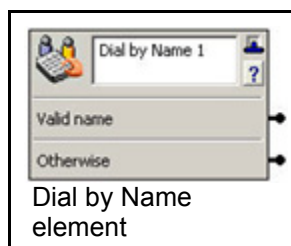


In order to transfer callers using the the Dial by Extension element, you must define the list of identities which the calls can be transferred to. For more information, refer to [Configuring non-user extensions and identities on page 31](#).



The Dial by Extension element is not available for PBA users.

Dial by Name element



The Dial by Name element allows callers to dial a user by entering the digits that correspond to the first few letters of either the person’s first or last name. The name is compared to the list of users defined by the contact list. The alphabetic characters correspond to the numerical digits on the caller’s telephone keypad. For information on selecting the order in which callers enter a name, refer to [Selecting the order in which callers enter a name using the dial by name mode on page 31](#).



The letters Q and Z appear on different keys on different phones. The Dial by Name element works whether the letter Q corresponds to the 0 or the 7 key, and whether the letter Z corresponds to the 0 or the 9 key.

Callers hear a standard announcement prompting them to spell either the first or last name. A caller can attempt to make an exact match of a name by pressing the # key after entering the digits. This is useful for matching shorter names such as Li. If a caller stops entering digits, and if the digits entered match five or fewer names, then the caller is presented with a list of names, and is prompted to make a selection. If more than five names match the digits entered, then the caller is prompted with *unable to match the dialed name*, and is prompted to try again.

If the name entered matches a UC server user, the call exits the element by the Valid name connector pin, and changes the Selected Extension and Selected User Name system values to the UC server user's extension and name.

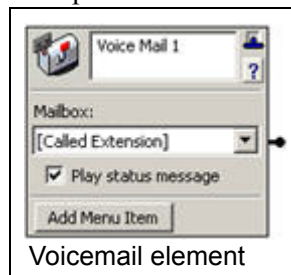
If the digits that the caller enters do not match a valid user or the caller does not enter any digits, the call moves to the element connected to the Otherwise connector pin. The caller is allowed three attempts to enter a valid user before the call exits the element.



The Dial by Name element is not available for PBA users.

Voicemail element

The voicemail element provides basic voicemail service. It plays the mailbox greeting and can take a message from a caller. If the caller presses the access number for mailbox management, the voicemail element allows mailbox management for the specified mailbox.



If the Mailbox selection is Called Extension, the mailbox is the one selected using a Select Extension element. In this case, the Select Extension element must come before the voicemail element in the call flow. If the Mailbox selection is Selected Extension, it refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as Called Extension.

The **Play status message** mode is enabled by default. The status message pre-empts the user's greeting message and cannot be interrupted. Make sure the **Play status message** mode is also enabled on a Play Announcement element, if required.

As Administrator, you can specify the mailbox where messages are stored.

A number of system-wide options apply to the voicemail element. These can be found on the voicemail tab under Options on the Tools menu. For information about administrator options for the voicemail element, refer to [Setting voicemail options on page 24](#).

Deliver Messages element

The Deliver Messages element sends voice and fax messages from a selected mailbox to the phone numbers that you specify. Called Extension refers to the identity originally called. Selected Extension refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as the Called Extension.



The UC server calls the sequence of numbers, in the order that you specify, until the call is answered. The call recipient must enter the password associated with the selected mailbox to retrieve messages over the phone. You can limit deliveries to new messages or allow full mailbox management.

Specify call numbers by using the telephone icon, or use the Value Selector. For more information, refer to [Selecting values within elements using the value selector on page 226](#). For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#).

The Mailbox list allows you to specify the mailbox from which the messages are sent.



Message delivery is provided only for messages left by incoming callers. It does not apply when a caller forwards, replies to, or sends a message.

To specify phone numbers for the Deliver Messages element

1. Select the telephone icon.
2. In the Enter Phone Number dialog box, type a phone number in the Telephone number box.



*You can also enter a phone number by right-clicking inside the **Call** box and selecting **New**.*

3. If you want to specify an account code, enable the **Use account code** check box and type an account code in the box.
4. Select **OK**. The target number appears in the Call Number(s) list. The UC server calls the target number when it attempts to deliver messages.

- Repeat steps 1 to 3 to add more target numbers to the list. The UC server calls each number in the list, according to the order you specify, until the call is answered.



Refer to [Selecting values within elements using the value selector on page 226](#) to select a dynamic value for the element.

To change the sequence of target phone numbers listed for Deliver Messages

- Select the number whose ranking you want to change.
- Select an arrow to move the number up or down in the list.



Up and down arrows

To delete a target phone number from the list for Deliver Messages

- Select the telephone number that you want to delete.
- Select the trash icon.



Trash icon

Transfer Call element



The Transfer Call element transfers calls to another number.

If the transfer fails, the caller leaves the element through the Otherwise connector pin. The UC server automatically adds a voicemail element to the Otherwise connector pin if you do not add an element.

The telephone number that you specify is subject to the outgoing call restrictions that are specified.

To specify a transfer target for the Transfer Call element

- In the **Transfer call to** section, select one of the following:
 - This number**—if you want to specify a number to transfer callers to
 - The operator**—if you want to transfer callers to the operator

2. If you selected **this number**, do one of the following:
 - Use the Value Selector to specify a value.
 - Select the telephone icon, and type the number where you want to transfer the caller.
 - Right-click the **This number** box, select **New**, and type the phone number that you want.



For information about selecting a dynamic value using a Value Selector, refer to [Selecting values within elements using the value selector on page 226](#). For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#).

Assisted Transfer element



The Assisted Transfer element allows a transfer call recipient to accept or deny a call based on the caller's recorded name. You can enter the telephone number to which you want to transfer calls. Specify the conditions under which you want to prompt callers to record their name. When you enter a list of telephone numbers, calls are transferred to the numbers in the order that they appear. You can use the arrow buttons to change the order of numbers in the list.

A call moves to the element connected to the Call is accepted connector pin when the transfer recipient answers and accepts the call. A call that is answered, and then declined by the recipient, moves to the element connected to the Call is denied connector pin.

If the call is not answered, is answered but not accepted within 10 seconds, or is transferred to an invalid number, the call moves to the element connected to the Otherwise connector pin.



The default is to wait 20 seconds for an answer before recalling an assisted transfer attempt. To change this value, refer to [Setting assisted transfer options on page 26](#).

To specify a call number for the Assisted Transfer element

1. In the Call number(s) section, select the telephone icon to open the Enter Phone Number box.



You can also enter a phone number by right-clicking the Call number(s) box and selecting **New**.

2. Type the telephone number to which you want to transfer callers.
3. From the **Prompt** list, select one of the following prompt types:

Never	Never prompts the caller to record a name.
Always	Always prompts the caller to record a name.
Unknown caller	Prompts the caller to record a name if there is no match in the contact list.



Use the Value Selector to select a dynamic value for the element. For more information, refer to [Selecting values within elements using the value selector on page 226](#). For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#).

Notify Pager element

The Notify Pager element allows you to send a message from the UC server to a specified pager number.

Notify Pager element

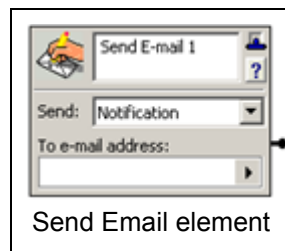
The message sent can be a numeric message with an attached caller ID.

To specify a pager number for the Notify Pager element

1. In the **Pager number to dial** section, do one of the following:
 - Use the Value Selector to specify a value. For more information, refer to [Selecting values within elements using the value selector on page 226](#). For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#).
 - Select the telephone icon and type the pager number that you want
 - Right-click the **Pager number to dial** box, select **New**, and type the pager number that you want
2. Enable the **Wait** check box if you want to specify a pause before paging.
3. In the **Wait...secs then** text box, type the number of seconds that you want the pause to last.

4. In the **Send Message** text box, type a prefix that you want as part of the message, before caller ID is displayed on the pager. Numeric prefixes and suffixes are useful to identify calls that are delivered. The prefix helps you differentiate between callers whose messages you have instructed the UC server to deliver, and other callers to the paging system. For example, you can use a prefix such as “123” or “000” to signal deliveries.
5. Enable the **Add Caller-ID** check box if you want to have caller ID sent with your messages.
6. In the **Append...to message** text box, type information that is part of the messages sent following any caller ID information displayed on your pager.

Send Email element



Using the Send Email element, you can specify the UC server to notify you, or someone else, by email when a caller moves successfully through the Send Email element. For example, you could have an email sent to your personal email address when a caller leaves a voice message for you after hours, or when you receive a fax. You can specify the type of message that you want to send and the address where you want to send it, including multiple email addresses. You can also send attachments with the email notification. For example, you could attach a fax of a property listing, or audio describing a property, or some other type of fax or audio specific to your particular business.



To use the Send Email element, make sure that you have a Microsoft Exchange Server or SMTP Server configured.

Selecting a Custom notification allows you to type or select information that you want to include in the notification. You can combine typed text and selected values in the Subject and Body sections of the element. In the call flow, the text and values are linked together. From the Attach field you can also select attachments to send with the notification, such as announcements, greetings, temporary recordings, faxes, or database values (for example, audio or fax).

Temporary recordings are created using the Record Announcement element. Using the Send Email element with a temporary recording can be used as a form filling application, such as requesting quotes, or answering general customer service questions, and so on. For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

To specify the NetVanta UC Server to send an email when you receive a voicemail message

1. Select one of the following from the Send list:
 - **Entire Message**—sends an email of all the information associated with the voice message that you received, including the voice message (as a .wav attachment)
 - **Notification**—sends only the information associated with the voice message
 - **Short Notification**—sends only some of the information associated with the voice message. This mode is useful for small menu devices, such as cell phones, that are capable of receiving email.
 - **Custom**—allows you to add information that you want to include with the email notification. This includes sending attachments such as announcements, greetings, temporary recordings, faxes, or database values (for example, audio or fax).
 - **Custom With Message**—allows you to send a custom email with the latest mail message



Custom notification sends an email notification of calls to a user's mailbox, whether the caller left a message or not. All other types of notifications send email notifications only when a caller leaves a message.

2. In the **To email address** box, type the email address where you want the notification sent. If you want to enter more than one email address, type them with a semicolon separating each one.



*Refer to [Selecting values within elements using the value selector on page 226](#) if you want to select a dynamic value for the **To email address** box. You can enter only one dynamic value.*



Make sure that you enter a valid email address. The UC server does not authenticate email addresses used with the Send Email element.

To specify a custom notification

1. From the Send list, select **Custom**.
2. In the **To email address** box, type the email address where you want the notification sent. Use a semicolon to separate multiple email addresses.
3. In the **Subject** box, type a subject for the email, or use the Value Selector to select a value.
4. In the **Body** box, type the body of the email, or use the Value Selector to select a value.



*Refer to [Selecting values within elements using the value selector on page 226](#) to select a dynamic value for the **To email address**, **Subject**, **Body**, or **Attach** text box. You can enter only one dynamic value.*

5. In the **Attach** field, select a database value (audio or fax), fax, announcement, greeting, or temporary recording.

Send Fax element

The Send Fax element allows you to send a list of faxes to a specified fax number.



You can enter the fax number to which you want to send a fax message. A cover page can accompany your fax message and you can select the fax documents to include in the list. You can choose to send a list of faxes, the latest fax in your mailbox, all faxes in your mailbox, or all unread faxes in your mailbox.

To specify a fax number for the Send Fax element

1. On the Send Fax element, select the telephone icon.
2. In the **Enter phone number** text box, type the fax number where you want the fax messages sent to.
3. If you want to specify an account code, enable the **Use account code** check box and type an account code in the box.
4. Select **OK**.



Refer to [Selecting values within elements using the value selector on page 226](#) to select a dynamic value for the element. For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#).

To specify a list of faxes for a Send Fax element

1. Select **List of Faxes** from the Send list.
2. Follow all the steps in [To specify a fax number for the Send Fax element on page 243](#).
3. In the **To** section, select the Value Selector and select the database value or fax that you want to send. The fax appears in the **To** list.
4. Repeat step 3 until all the faxes that you want are added to the list.

To include a cover page for the Send Fax element

1. On the Send Fax element, enable the **Include cover page** check box, which opens the Fax Cover Page Details box.
2. Select the browse button to open the **Select Cover Page** box.
3. Select the fax cover page that you want to include.
4. Select **Select**. The name of the fax cover page appears in the Select cover page text box.

5. In the Fax Cover Page Details box, type information about the fax recipient, the subject of the fax, and comments that you want to include on the cover page. You can use a combination of typed text and dynamic values.



Refer to [Selecting values within elements using the value selector on page 226](#) to select dynamic values for the cover page information.

6. Select **OK**.

To send faxes in your mailbox using a Send Fax element

1. From the Send list, select one of the following:
 - **Latest Fax in Mailbox**
 - **All Faxes in Mailbox**
 - **All Unread Faxes in Mailbox**
2. Follow all the steps in [To specify a fax number for the Send Fax element on page 243](#).
3. If you want to include a cover page, follow all the steps in [To include a cover page for the Send Fax element on page 243](#).
4. From the **From** list, select the mailbox from which you want to send the fax. Called Extension refers to the identity originally called. Selected Extension refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as the Called Extension.

Receive Fax element



Your Personal Business Attendant (PBA) can automatically receive faxes without using a Receive Fax element, however you may want to use the Receive Fax element in order to customize the receive fax behavior, or to create a service specifically for receiving faxes.

The default behavior for incoming faxes for any PBA users is that the UC server detects the tone that signals a fax, the UC server receives the fax, and then stores it in the user's mailbox. No additional configuration is required, however using the default behavior, no notifications are sent to the user that they have received a fax.

The default behavior for admin users using an attendant service is that the UC server detects the standard CNG tone that signals a fax, the UC server receives the fax, and then stores it in the default mailbox. An office administrator can then send received faxes by email to a user's mailbox.

The Receive Fax element receives incoming faxes and places them into a specified mailbox or database. Use the Receive Fax element along with other elements in a service to customize receive fax behavior.

Depending on the configuration, you may have a dedicated fax number or identity to receive faxes. In this case you may want to use the Receive Fax element in a service to initiate some behavior, such as storing the fax in a database. Alternatively, you may have a shared voice and fax number or identity, where you receive voice and fax messages on the same identity or extension. If you are using a shared voice and fax

number, then you may want to create a **Receive Fax** service, for example, in order to receive notifications that you have received a fax.

Using the Receive Fax Element in a Service

Save Fax to Mailbox

Called Extension	Selecting this mode enables you to save incoming faxes to your mailbox, or the extension number that the caller called.
Selected Extension	This mode refers to the Select Extension element, which is available to admin users only. If an admin user creates a service using the Select Extension element, and makes this service available to other users by saving it to the Shared folder, then PBA users will have this mode available. However, PBA users should not select this mode when creating their own service. If an admin user selects this mode then they must place a Select Extension element on page 249 somewhere before the Receive Fax element in the service.
User Extension	Note: Only the administrator can select a specific user's mailbox.
Save Fax to Database	Selecting this mode enables you to save incoming faxes to a database, if your system is licensed for database integration. If you have the Database tab available in the UC client Service Editor, then you can use the Data Source Selector to save incoming faxes to a database. For more information, refer to Using the Data Source Selector on page 297 . Note: If the <i>Save Fax to Database</i> mode is selected and the database is unavailable, the incoming fax is delivered to the default mailbox.
Otherwise	If a problem occurs receiving a fax, then the call exits through the Otherwise connector pin, and the fax is not received in the selected mailbox.

When the Receive Fax element is used in a voice service, it's typically placed after a [Menu element on page 232](#). Callers are instructed using a [Play Announcement element on page 229](#) that they can press a key on the telephone dialpad to send a fax. For example:

"If you would like to send a fax, please press 5 and then press the Send button on your fax machine."

When the Receive Fax element is used in a dedicated fax service it is typically the first element in the service, and it can be combined with other elements to create a custom fax service, such as saving the fax to a database, or receiving notifications of the received fax.

Using the Receive Fax service

The **Receive Fax** service is a particular type of service that has additional requirements beyond regular service requirements. If you are using the same identity to receive both voice and fax messages, then you may want to use the **Receive Fax** element to create a specific **Receive Fax** service, for example, to send a notification to a pager, a phone number, or an email address to notify you that you've received a fax.

Your system administrator may have created a **Receive Fax** service for all users, available in the Services\Shared folder. If a shared **Receive Fax** service is available then you do not need to create your own service, unless you want to create a custom service. If you do not have a shared **Receive Fax** service available, or if you want a custom service, then you can create your own custom **Receive Fax** service.

Requirements for using a Receive Fax service on a shared voice and fax identity

- The Receive Fax service must be called **Receive Fax**.
- The Receive Fax service must contain a Receive Fax element.
- The Receive Fax element must appear as soon as possible within the service, typically the first element of the service.
- You do not need to activate the Receive Fax service. When a fax call comes in, your active service will be initiated. When the UC server hears the fax tone, it automatically stops the regular active service and activates the **Receive Fax** service.
- The Receive Fax service must be saved in one of the following two locations:
 - The shared folder so that it is available to all users in the system (that is, \Services\Shared\Receive Fax)
 - The user's root folder so that it is only available for that user (that is, \Services\Receive Fax)

Creating a Receive Fax service to receive fax notifications

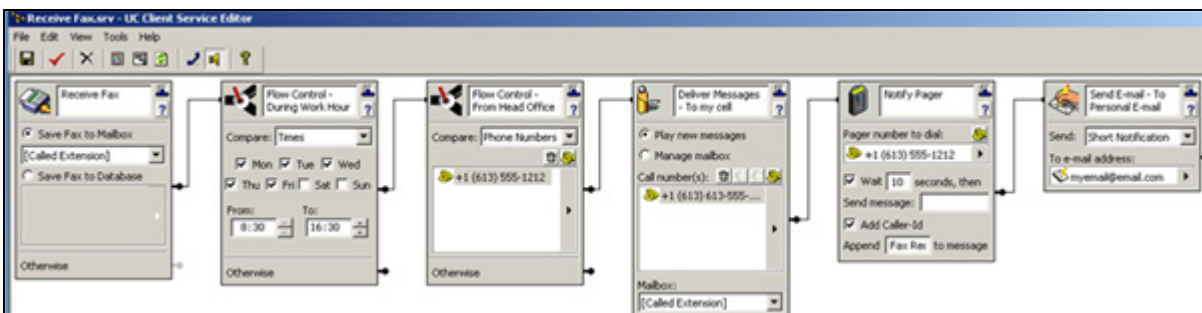
In order to receive a notification of a received fax you can follow the Receive Fax element in a **Receive Fax** service with any or all of the following elements:

- Send Email element
- Deliver Messages element (to send a phone message)
- Notify Pager element

If you want to specify the days of the week or times of day when you want notifications sent, or specify from which phone numbers you want to be notified of a received fax, then use the [Flow control element on page 233](#).

Receive Fax Service Example

The following example shows how you might use the Receive Fax element in a Receive Fax service to notify you during particular days or times, that an incoming fax has been received from a specific fax number. The example below illustrates that when a fax is received during business hours from a specific number, then notifications are sent to the user's cell phone, pager, and personal email account.



To specify a general mailbox

1. In the Administrator application, select **Tools > Options**.
2. Select the **Operator Definition** tab.
3. Select a mailbox from the Default Mailbox list.

4. Select **OK**.



To troubleshoot any problems with the Receive Fax element, refer to [Troubleshooting Services on page 223](#).

Fax-on-demand element

The Fax-on-demand element allows the UC server to send a fax to callers who move through this element. You can select the fax document to be made available and a cover page to accompany the fax.

Some callers might cancel their request. You can add an element to the service to handle such cases and route the call to the next element in the call flow. For information about configuring fax specifications for the Fax-on-demand element, refer to [Setting fax message delivery options on page 26](#).

To select a fax document for a Fax-on-demand element

On the Fax-on-demand element, select the Value Selector to select a fax document or dynamic database value. The name and icon of the document or value appear in the Fax document to send list.



To preview a fax document that you selected, select the [...] button in the Fax document to send section. The preview fax button works only with a real fax, not a database reference.

To select a fax cover page to accompany the fax document for a Fax-on-demand element

1. On the Fax-on-demand element, enable the **Include cover page** check box.
2. In the Fax Cover Page Details dialog box, select the Browse button.



Browse button

3. Select a fax cover page.
4. Select **Select**. The cover page title appears in the Select cover page file text box of the Select Cover Page dialog box.
5. In the Subject and Comments text boxes, type information that you want to include on the fax cover page.
6. If you want to include dynamic information in the Subject and Comments text boxes, use the Value Selectors.



You can include both text and value information in the Subject and Comments boxes of the Fax-on-demand element.

7. Select **OK**.

Using Advanced Elements

Select the Advanced tab on the tool palette to view the advanced elements. The tab is shown on the left. A brief description is provided for each element that is available.

Table 15–2: Advanced elements

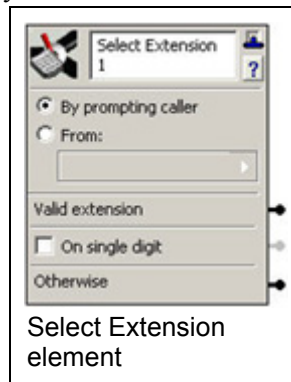
Select Extension element on page 249	Waits for callers to enter an extension number using the telephone keypad; can be used for selecting mailboxes
Record Announcement element on page 252	Allows callers with user accounts to record over existing announcements or greetings, create temporary recordings, or record into a database, using a remote telephone
Take Message element on page 255	Triggers the recording of an incoming message for storage to a mailbox
Advanced Menu element on page 256	Allows callers to make a selection using the telephone keypad
Manage Mailbox element on page 257	Allows callers to retrieve messages remotely
Change Mailbox Password element on page 258	Allows users to change their mailbox password
Verify Password element on page 258	Allows the service to request a password from callers and ensure that a given password matches a specified one or the password for a particular mailbox
Loop Counter element on page 259	Allows you to limit the number of times the call flow will execute a path in the call flow
Text To Speech element on page 260	Reads text to the caller
Gather Digits element on page 260	Allows users to keep track of information that is entered by callers
Compare Data element on page 261	Allows users to compare patterns of data and directs call flow accordingly
Create Log Entry element on page 262	Allows users to capture and export detailed call log entry information
Page Live element on page 263	Connects the caller to the selected paging group, and allows the caller to do a live page.
Page Recorded element on page 264	Queues a prerecorded page for a specified paging group. The request is associated with an audio file to be played when the paging request is initiated.
Generate Tones element on page 266	The Generate Tones element enables a service to insert a dual tone multi-frequency (DTMF) tone to the current call.
Queue Outdial element on page 267	This element enables a service to launch an outdial request after a specified period of time.
Execute Program element on page 269	Enables a service to launch an external program.
Wait for Silence element on page 271	This element is typically used to enable an outdialing service to wait for a period of silence.
Detect Remote Party element on page 271	Detects whether a call is answered by a person, an answering machine, or a fax machine, typically during an outdial service.



The Select Extension, Page Live, Page Recorded, and Execute Program elements are available to Administrators only and not to Personal Business Assistant (PBA) users. However, Admin users can create shared services that contain these elements, and these services can then be assigned by a PBA user to one of their identities.

Select Extension element

The Select Extension element is typically used in services to be shared by several users.



Callers can be prompted to enter an extension number using their telephone keypad. When a caller enters a valid extension number, the call is routed to the next element in the call flow.

Alternatively, an extension number can be selected based on some information automatically provided, such as the called number, or the caller-ID number. The call information provided is compared to information in a database, and the call is routed accordingly. The Select Extension element validates the information provided. For example, if you use the voicemail element, and choose Selected Extension within the voicemail element, then you must have a Select Extension element in the service to provide the selected extension number.

You can use the Select Extension element to change the Select Extension system value to a different Identity number. The element also changes the Selected User Name system value to the User name associated with the new valid Identity number. The Select Extension element works with several elements including the Manage Mailbox, Verify Password, and Take Message elements, and generally works with any element that can reference system values.

You can select an Identity number from a Data Source value. If you select this mode, then include a database element (for example, Fetch Data, Dial for Data, Move Current Row, or Prompt for Current Row) that indexes the Data Source value that you wish to use. For example, use a database element to verify the password before the Verify Password element in a call flow, so that you can reference it in the Verify Password element's Data Source selector.

Requirements

- The Select Extension element is available to administrators only. However, an administrator can create a service that uses the Select Extension element, and then share the service so that it is available to PBA users.
- Ensure that the Data Source value used is a Number data type, since only data source values of this type

can be selected.



Data Source values that contain non-numeric characters will not provide a valid Identity and will cause the element to follow the Otherwise connector pin.



In order to use the the Select Extension element, you must define the list of extensions which can be selected. For more information, refer to [Configuring non-user extensions and identities on page 31](#).

When a caller enters a valid extension number, or the data source referenced returns a valid Identity number, the call follows the Valid extension connector pin to the next element in the call flow.

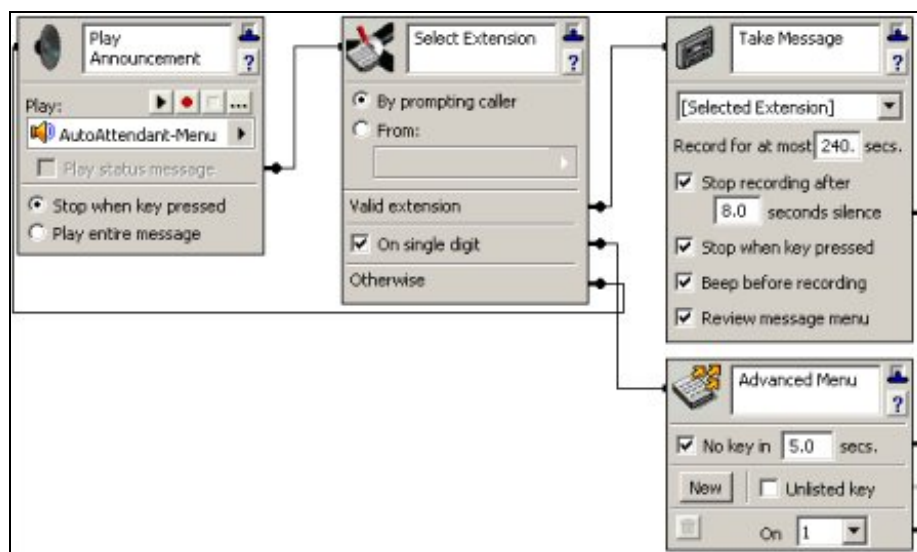
If a caller enters a single digit when On single digit is enabled, the call moves to the element connected to the On single digit connector pin. This mode is useful if you want to connect an Advanced Menu element. Callers can then enter a single digit from 0 to 9, from the options available (for example, “Press 1 to speak to an operator”).

When using caller prompt, a call follows the Otherwise connector pin when three sequential selections fail to match either an extension or a single digit (if enabled). When using a Data Source, a call follows the Otherwise connector pin if the Data Source does not return a valid Identity number.

Applications

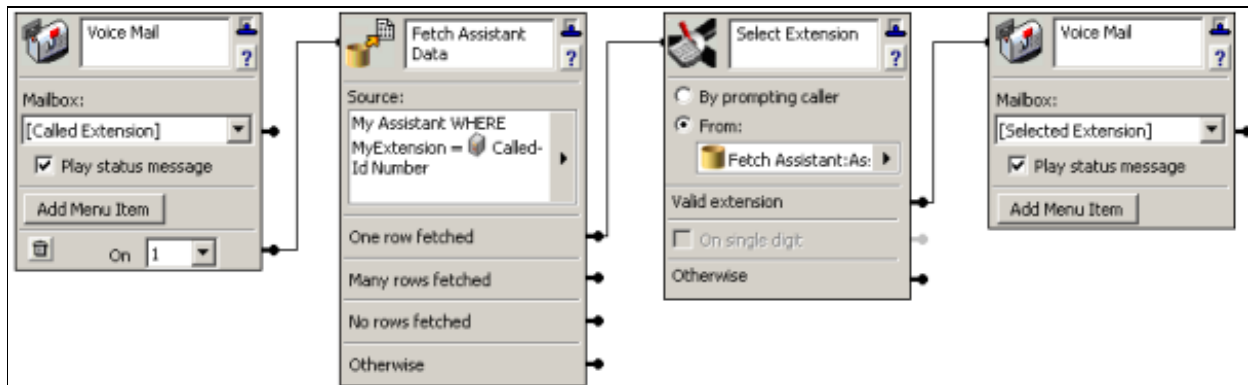
The Select Extension element is particularly useful when you want to share a service so that it can be used by many users, and you want to select a particular extension in which to apply some call answering behavior. The examples below illustrate some ways in which you can use the Select Extension element in a service.

To record a message in a user’s mailbox or select from a list of menu options



In the example above, the caller first hears the main auto attendant menu. The caller is then prompted to enter an extension number or select from a list of menu options. If the caller enters a valid extension number then the caller is transferred to the voicemail of the selected extension, where their message is recorded. If the caller enters a single digit, then the caller is transferred to the menu mode that corresponds with that digit (for example, “Press 1 to speak with an operator”).

To transfer caller's to an assistant's mailbox



In the example above, callers call into a direct line to a user's voicemail, and are routed to that user's assistant's mailbox. For example, “You have reached the voicemail box of John Smith. To leave a message for my assistant, Tony Moore, press 1.” It is not practical to create individual services to associate each user with their particular assistant, so instead one service is created and shared among many users. The service determines whose extension was called, and based on the called number, fetches the data from the database to determine who the assistant is for the called number, and directs the caller to that user's assistant's mailbox. To determine the “Selected Extension” in the second voicemail element, the Select Extension element must be included in the service. To determine the appropriate assistant's extension to direct the caller to, information must be gathered from a database, in this case using the Fetch Data element, and using a data source table that includes fields for user's extensions, and their assistant's extensions.

To prompt callers to enter an extension number or enter a single digit to access a menu item

1. In the Select Extension element choose **By prompting caller**. The service prompts callers to enter an extension number using their telephone keypad. If a caller enters a valid extension number, then the call exits through the **Valid extension** connector pin.
2. If you want callers to be able to enter a single digit to access a menu item, then enable **On single digit**. If a caller presses a single digit on their telephone keypad, the call follows the **On single digit** connector pin.

To select an extension automatically based on some value selected or on information provided

1. In the Select Extension element choose **From**.
2. Select **Database Values**, **Gather-Digits Values**, or **System Values**.

3. Select a specific value from the list available. If you select a database value, then you must include a database element in the service. If you select a gather-digits value, then you must include a Gather Digits element in the service. The **On single digit** mode is not available if a data source value is selected.

Record Announcement element

The Record Announcement element allows callers to record over an existing announcement or mailbox greeting, record a temporary recording, or record into a database, using a remote telephone.



This element is useful for audio information services that must be updated on a daily basis or at regular intervals. A user can update the recorded information remotely when you include this element in a service.

You can specify where to record an announcement or mailbox greeting. You can choose a specific file location and an associated greeting type. Make sure to include a Select Extension element before the Record Announcement element in the call flow if you want callers to record to a different extension than the one they called.

Instructing callers how to make a recording

The Record Announcement element sends a voicemail prompt tone to the caller, and then starts to record the caller. You may want to use a Play Announcement element before the Record Announcement element to instruct the caller on how to stop the recording. The caller must press a telephone key to stop recording.

Action	Key	Description
Stop recording\Plays menu options	4-9, 0, #	Stops recording and prompts the caller with the menu options for 1, 2, 3, and the * key on the telephone keypad. If the caller does not subsequently press a key, then after repeating the menu options three times, the element keeps the current recording and the call flow proceeds. If a caller presses keys 4 to 9 or 0, the prompt menu is repeated the first time, then subsequently has the same effect as if the 1 key was pressed.

Action	Key	Description
Listen to the recording	3	Stops recording, plays the recorded audio to the caller, and then prompts the caller with the menu options.
Record again	2	Stops recording, discards the current recorded audio, sends a voicemail prompt tone to the caller, and allows the caller to start recording again. The caller must press a telephone key to stop the recording.
Keep the recording/cancel	1	Stops recording, keeps the recorded audio, and the call flow proceeds.

Recording an announcement

To enable callers to change a specified announcement using a remote telephone

1. Enable the **Specific** button.
2. Select the Value Selector arrow, select **Announcements**, and select the recording that you want. The name of the announcement appears in the Specific text box.

Recording a greeting

To enable callers to change a specified extension or user greeting using a remote telephone

1. Enable the **Specific** button. Select the Value Selector arrow, select **Greetings**, and select one of the users or [Called Extension]. Called Extension refers to the identity originally called. Selected Extension refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as the Called Extension.



Only Admin users can add a Select Extension element to a service. However, if an Admin user shares such a service with a PBA user, then the Select Extension element within that service becomes available to the PBA user.

The name of the greeting appears in the Specific text box.

2. From the **Greeting** type list, select one of the following greeting types:
 - **Internal Greeting**
 - **External Greeting**
 - **Out-of-Office Greeting**
 - **Mailbox Identifier**



Put a Verify Password element before this element if you want to prevent unauthorized access.

Recording a temporary recording

You can use the Record Announcement element to record audio in a specific temporary recording, and

elements like Play Announcement can be used to play the audio. Temporary recordings exist only for the duration of the execution of a service. That is, when a service starts, none of the temporary recordings exist. If a caller records a temporary recording, then the recording is stored in a temporary file that can be played by another element, and the recording is discarded when the service ends. Temporary recordings can be referenced by the following elements:

- Play Announcement element
- Menu element
- Send Email element (custom)
- Page Recorded element
- Add Data element
- Update Data element



To use the Add Data and Update Data elements, the system must be licensed for database integration.

For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

To enable callers to create a temporary recording using a remote telephone

1. Enable the **Specific** button.
2. Select the Value Selector arrow, select **Temporary Recordings**, and select the temporary recording that you want. The name of the temporary recording appears in the Specific text box.

Recording into a database

Record the announcement to a database by selecting a data source and specifying the filtering information.



To access database information, the system must be licensed for database integration.

To enable callers to change an announcement in a database

1. Enable the **Database** button.
2. Select the **Data Source Selector** arrow. For more information, refer to [Deliver Messages element on page 237](#).
3. Specify the Data source, Filter field, Equals, and Update information that you want.

Advanced Record Announcement Options

The Advanced Record Announcements allow you to define additional attributes for the recording. To change the advanced options select on the show or hide advanced options button.

You can limit the length of the message that can be recorded and avoid recording silence. Enable the corresponding check boxes to allow callers to signal the end of their message, enable a beep before recording, and enable review message menu.

To limit the length of a recorded message

Type a number of seconds in the **Record for at most** text box. Callers cannot leave a message that lasts longer than the time limit that you specify.

To avoid recording silence

1. Enable the **Stop recording after** check box.
2. Type a number of seconds in the text box.



If you set the number of seconds of silence too low, callers might be cut off while pausing between phrases or speaking quietly.

To set Take Message specifications

- Enable any of the following check boxes:
 - **Stop when key pressed**—callers proceed to the next element after they press a key
 - **Beep before recording**
 - **Review message menu**

Take Message element

The Take Message element records an incoming message. Called Extension refers to the identity originally called. Selected Extension refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as the Called Extension.

This element can receive voicemail messages and store them in a specific mailbox.



First, select Called Extension from the list. Make sure to include a Select Extension element before the Take Message element in the call flow so that callers can enter the extension number that they want. If you do not include a Select Extension element in the service, the message goes to the Administrator's mailbox.

You can limit the length of the message that can be recorded and avoid recording silence. Enable the corresponding check boxes to allow callers to signal the end of their message, enable a beep before recording, and enable review message menu.

To limit the length of a recorded message

Type a number of seconds in the **Record for at most** text box. Callers cannot leave a message that lasts longer than the time limit that you specify.

To avoid recording silence

1. Enable the **Stop recording after** check box.
2. Type a number of seconds in the text box.



If you set the number of seconds of silence too low, callers might be cut off while pausing between phrases or speaking quietly.

To set Take Message specifications

- Enable any of the following check boxes:
 - **Stop when key pressed**—callers proceed to the next element after they press a key
 - **Beep before recording**
 - **Review message menu**

Advanced Menu element

The Advanced Menu element allows callers to make a selection using the telephone keypad. It is advisable to put an audio element (such as Play Announcement) that specifies menu options before the Advanced Menu element.

The Advanced Menu element can also be attached to the “on single digit” connection of certain elements.

Some callers might not make a selection from the menu.

For this purpose, the No key in check box is provided. Select the check box and then type in a number of seconds to wait in the associated text box. You can specify the amount of time the UC server waits for a response before it moves to the next element.

To allow callers to move to the next element if they do not select a key

1. Enable the **No key in** check box.
2. In the text box, type the number of seconds to wait.

To set selection keys on a menu

1. In the first **On** list box, select the number of the key that you want. The number one appears by default. When callers press a key, they are sent to the element that is adjacent to the connector pin, depending on their choice.
2. Select **New** to add an On list box for setting an additional key. The UC server enters the keys in numerical sequence. You can also select a key from the list. You can set a maximum of 12 keys. Keys range from one to nine, then include zero, * and #.

When you have finished setting selection keys, you can connect the subsequent element for each selection key that you set, as applicable.

To remove a selection key from a menu

Select the trashcan icon to the left of the key that you want to delete.

Manage Mailbox element



The Manage Mailbox element allows callers to retrieve messages remotely over the telephone. Called Extension refers to the identity originally called. Selected Extension refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as the Called Extension.

As Administrator, you need to specify the mailbox that a caller can manage or insert a Select Extension element to allow UC servers to select a mailbox. In the Mailbox list, select Called Extension. Make sure to include a Select Extension element before the Manage Mailbox element in the call flow so that callers can enter an extension number.

You can set up the next element in the call flow depending on whether callers enter a valid extension number or cancel the management attempt.



This element prompts callers for a mailbox password before allowing access.

Change Mailbox Password element



The Change Mailbox Password element allows users to change their mailbox password.

The element prompts callers for a new password, waits until they enter a new password, and then confirms it by playing it back to callers. Callers must accept the change or reject it.

In the Mailbox list, select Called Extension. Make sure to include a Select Extension element before the Manage Mailbox element in the call flow so that callers can enter the desired extension number.

Called Extension refers to the identity originally called. Selected Extension refers to the identity selected in the last successful Select Extension element in the service. If there was none, or the last one failed, it defaults to be the same as the Called Extension.

For added security, you might want to put a Verify Password element before this element.

Verify Password element



The Verify Password element enables the service to request a numeric password from a caller, and verifies that a given password matches a specified password, the password for a particular mailbox, or the password from a Data Source. The Verify Password element would typically be used to access or to change information based on authentication.

Administrators may use any of the following sources to verify the password provided.

1. The password associated with any of the following mailboxes:

Called Extension	This refers to the mailbox associated with the identity originally called.
Selected Extension	This refers to the mailbox associated with the identity selected in the last successful Select Extension element executed in the service. If you choose this mode, ensure that you include a Select Extension element somewhere before the Verify Password element in the call flow, to select the mailbox associated with the identity for which you want to verify the password. If there was no selection, or the last selection failed, it defaults to be the same as the Called Extension.

A selected user

2. A password contained in a Data Source value.

If you select this mode, then be sure to include a database element (such as the Fetch Data, Dial for Data, Move Current Row, or Prompt for Current Row elements) that indexes the Data Source value that you wish to use to verify the password. The database element must occur before the Verify Password element in the call flow, so that you can reference it in the Verify Password element's Data Source selector.

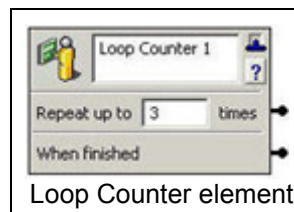
Data Sources cannot be referenced directly, so you must reference a filtered Data Source created by one of the database elements referenced above. Ensure that the Data Source value that you use is a Number data type, since only Data Source values that have this type can be selected.

The Otherwise connector pin is followed when:

- The password does not match the selection criteria after three sequential attempts, or
- The caller has not entered any digits after three sequential attempts.

You can set up the next element in the call flow depending on whether a caller enters a valid extension number or cancels the attempt.

Loop Counter element



The Loop Counter element allows you to limit the number of times the call flow makes a connection. This element affects only call flow control. Callers do not hear anything or perform an action.

In the Repeat up to text box, enter the number of times you want a connection to be followed.

Once the loop counter is encountered the number of times you have specified, the call follows the When finished connection.

All loop counters are reset at the beginning of each call.

Text To Speech element

The Text to Speech element allows users to convert text to speech. The supplied text is “spoken” to callers by the Text-to-Speech engine. You can use this element when you prefer not to make audio recordings. Type the text manually, or use the Value Selector to specify a dynamic value for the text.



For more information, refer to [Selecting values within elements using the value selector on page 238](#). You can also use a combination of typed text and selected values.

Gather Digits element



The Gather Digits element allows the caller to enter digits on the telephone keypad that are compared to a Compare Data element in the same service. The Gather Digits element is typically used with an element that plays audio, such as the Play Announcement element, to give the caller instructions. When the UC server has gathered the specified number of digits, the caller moves to the connected element.

If no digits are gathered, the caller moves to the element connected to the No digits collected connector pin.

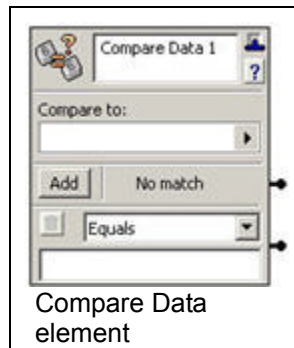
If too few digits are gathered or the time limit for entering digits is exceeded, then the caller moves to the element connected to the Otherwise connector pin.

The Gather Digits element works with the Compare Data element. The Gather Digits element must appear in the call flow before the Compare Data element but does not need to be connected directly to it. You can include elements between them in the call flow.

To specify the number of digits to gather

1. Enable one of the following:
 - **Stop after**—the number of digits is fixed. It lets you specify the number of digits.
 - **Gather up to 20 digits**—the number of digits is variable, up to 20
2. If you selected the Stop after button, then type the number of digits in the text box.

Compare Data element



The Compare Data element compares data that is collected by a Gather Digits element, system values (for example, Today's Date), or database values, and directs the call flow accordingly. For example, you might want a caller to enter a four-digit number to continue in the call flow. Once the caller enters the number and it is stored by the Gather Digits element, the Compare Data element can let you direct a specific caller through the call flow based on that four-digit number.

The Compare Data element executes as follows:

- The first comparison value is evaluated for a match against the comparison selected. If the comparison is true, then the call exits through the associated connector. If the comparison is not true, then the next comparison value is evaluated for a match.
- If no match is found when comparing data, then the call proceeds to the element connected to the No match connector pin.

Note that if comparisons other than Equals are selected, then the order of comparison values is important, and can be used to match between two values. For example, if the Compare to value is a numeric value and the following comparisons are used:

- Less Than: 5
- Greater Than: 10
- Less Than/Equal: 10

Then the following results occur:

- If the Compare to value is less than 5, then the first connector is taken.
- If the Compare to value is greater than 10, then the second connector is taken.
- If the Compare to value is between 5 and 10 (including 5 or 10), then the third connector is taken.
- If the Compare to value had no value, then the No match connector pin is taken.

To direct a caller using the Compare Data element

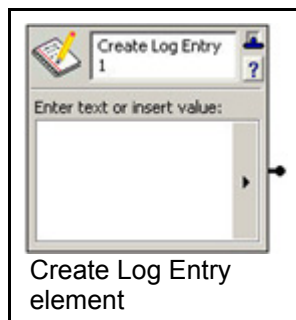
1. From the Compare to Value Selector list, select the Gather Digits element, System Value, or database reference that you want to compare to.



Refer to [Selecting values within elements using the value selector on page 238](#) if you want to select a dynamic value for the Compare to box.

2. From the drop-down list select a comparison mode that you want to associate with a specific caller.
3. Select **Add** to select additional comparison options. You can add up to ten compare data connectors.

Create Log Entry element



The Create Log Entry element allows detailed call information to be exported as a tab separated file or XML-based file. You can specify the type of information that you want to include in the call log entry. Use the Create Log Entry element with the Gather Digits element if you want to log the digits that a caller enters.

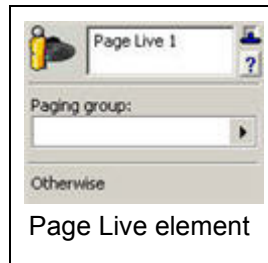
To specify types of information using the Create Log Entry element

1. In the **Enter text or insert value** section, select the Value Selector arrow. For more information, refer to [Selecting values within elements using the value selector on page 238](#).
2. Select dynamic values for the log entry, and type any text that you want to appear in the log.
3. Repeat steps 1 and 2 until all the information that you want to include appears in the list.



Some of the information types appear blank in the call log, depending on the PBX.

Page Live element



If you are using SIP paging, then you can use the Page Live element to initiate a live page from a call answering service. The element connects the caller to the selected paging group, and allows the caller to do a live page into the phone once they are connected to the paging group. Audio is spoken by the caller and played immediately to the paging group. Using the Page Live element in a service enables you to send broadcast announcement pages to phones, identities (that is, extensions), and Multicast paging groups.



The Page Live element is not available to Personal Business Assistant (PBA) users. However, Admin users can create shared services that contain the Page Live element, and these services can be assigned by a PBA user to one of their identities.



For information on configuring paging, refer to [Configuring Paging on page 61](#).



Using the Page Live or Page Recorded elements in a service is supported if your system is configured for SIP paging. Using the paging elements in a service is not supported for paging using a sound card or a gateway.

To select a paging group to send a live page to

Under the Paging group submenu select one of the following:

Database Values	Your UC server version must be licensed for database integration in order to use this mode
Gather-Digits Values	Use the Gather Digits element in the service so that the caller can select a paging group from a list provided.
Parameters	View > Parameters must be enabled.
Paging Groups	Select from one of the paging groups available. The paging group must be pre-defined using the Paging Group Wizard. For more information, refer to Creating a Paging Group on page 65 .

If the value selected matches a paging group, then the caller is transferred to the paging system, as if they had dialed the access code for the specific paging group. If the value selected does not match a paging group, then the element exits through the Otherwise connector pin.

Page Recorded element

If you are using SIP paging, then you can use the Page Recorded element in a service to send prerecorded pages to phones, identities (such as extensions), and Multicast paging groups.



The Page Recorded element queues a prerecorded page for a selected paging group. The request is associated with an audio file to be played when the paging request is initiated.

The Page Recorded element is typically used with temporary recordings. This allows a caller to verify broadcast paging announcements before they are played to the selected paging group. Alternatively, prerecorded announcements can be selected through a service to be played to the selected paging group. For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

Paging requests are processed by the system in the order they are received. All queued paging requests are processed before any Active Message Delivery (AMD) or Outdial requests.



The Page Recorded element is not available to Personal Business Assistant (PBA) users. However, Admin users can create shared services that contain the Page Recorded element, and these services can be assigned by a PBA user to one of their identities.



For information on configuring paging, refer to [Configuring Paging on page 61](#).



Using the Page Live or Page Recorded elements in a service is supported if your system is configured for SIP paging. Using the paging elements in a service is not supported for paging using a sound card or a gateway.

To select a paging group to send a prerecorded page to

1. Under the Paging group submenu select one of the following:

Database Values	Your UC server version must be licensed for database integration in order to use this mode
Gather-Digits Values	Use the Gather Digits element in the service so that the caller can select a paging group from a list provided.
Parameters	View > Parameters must be enabled.
Paging Groups	Select from one of the paging groups available. The paging group must be pre-defined using the Paging Group Wizard. For more information, refer to Creating a Paging Group on page 65 .

If the value selected matches a paging group, then the caller is transferred to the paging system, as if they had dialed the access code for the specific paging group.

If the value selected does not match a paging group, then the element exits through the Otherwise connector pin.

2. Under the Recorded audio submenu select one of the following to specify the audio to be played:

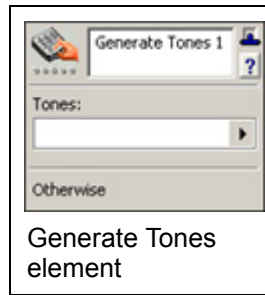
Database Values	Your UC server version must be licensed for database integration in order to use this mode
Announcements	
Greetings	
Temporary Recordings	Temporary recordings can be used to record audio temporarily, after a service has started. Use the Record Announcement element to record the audio for a specific temporary recording. The recordings are then stored in a temporary file, which can be played by another element such as Page Recorded. All temporary recordings are discarded when the service ends. Up to 10 temporary recordings can be made.

If the paging group and recorded audio values selected are valid, then the page request is queued in the system, the element exits through the Paging group connector pin, and service execution continues.

If the paging group value selected does not match a paging group, then the element exits through the Otherwise connector pin.

If the audio selected does not exist or is zero length, then the element exits through the Otherwise connector pin.

Generate Tones element



The Generate Tones element enables a service to insert a dual tone multi-frequency (DTMF) tone to the current call. DTMF tones are also known as touch-tone dialing.

The Generate Tones element would typically be used in an outbound dialing service. For example, an outbound dialing service might be required to enter DTMF digits to dial an extension number for a specific caller in an automated attendant, to access an external paging system, or to enter an account code.

The characters 0-9, *, and # produce the associated tones. You can introduce pauses between digits by entering commas between digits. For example, “1,2,3,4”.

If the tones are generated successfully, then the call exits through the Tones connector pin. If the tones cannot be generated for any reason, for example, the call does not exist, then the call exits through the Otherwise connector pin.

To select the tones to generate

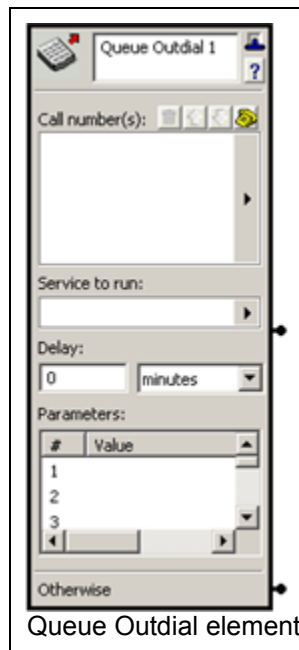
Under the Tones field select one of the following value reference controls to indicate the DTMF tones to produce.

Database Values	Requires licensing for database integration. If you have a Database elements tab in addition to the Standard and Advanced elements tabs, then you are licensed for database integration.
Gather-Digits Values	Use the Gather Digits element on page 260 prior to the Generate Tones element to enable callers to input digits into their telephone keypad.



NOTE: For information on creating services for outbound dialing, refer to [Creating Outbound Dialing Services on page 206](#).

Queue Outdial element



You can create a service that uses the Queue Outdial element to launch an outdial request after a specified period of time. When an outdial element is executed, a new outdial request is inserted into the server, as if the outdial command had been executed.



For information on the outbound dialer, refer to [Using the Outbound Dialer on page 197](#).

You can enter one or more phone numbers manually in the Call numbers box, or you can use the value selector to select phone numbers from a database or contacts list.



For information on using the value selector, refer to [Selecting values within elements using the value selector on page 226](#). For information about contact address book synchronization, refer to [Synchronizing contacts on page 226](#)

You can use the value selector to select a specific service to run when the outdial request is made, and you can specify a delay to indicate when the outdial request should be processed.

You can specify up to 10 parameter values to be used in the outdial request. Using the value selector you can select from a database value, a value obtained from the Gather Digits element, or a System Value, for example, today's date.

The Queue Outdial element executes as follows:

- The call exits through the first connector pin when the outdial request is successfully created. The element executes without waiting for the request to be performed.

- If the outdial request was unsuccessful, then the call exits through the Otherwise connector pin.



The Queue Outdial element is available for administrators only. If an administrator creates a service using the Queue Outdial element, and makes this service available to other users by saving it to the Shared folder, then PBA users will have this option available.

To select numbers for the outdial request to call

1. From the Queue Outdial element in the Call numbers list, select the Value Selector arrow.
2. Select a database value or contacts from the Contacts list.



*You can also enter phone numbers manually by selecting the telephone icon to open the Enter Phone Number dialog box. Type the phone number that you want in the Telephone number box, and select **OK**. Alternatively, you can enter a phone number by right-selecting in the Call numbers box and selecting **New**.*

To select a service to run

Use the value selector arrow to select a database value or an existing service.

To set a delay before the outdial request is processed

1. Select one of the following:
 - minutes
 - hours
 - days
2. Enter a number in the Delay field.

To specify parameter values to be used in the outdial request

1. Select in the Parameters Value field and select the Value selector arrow.
2. Select one of the following:
 - Database Value
 - Value obtained from a Gather Digits element
 - System Value
3. Continue the steps above to enter up to 10 parameter values.

Execute Program element



The Execute Program element provides the ability to launch an external program from within a service.



The Execute Program element is only available to Admin profile users when creating and editing services, and is not available to Personal Business Assistant (PBA) users.



To avoid security concerns, the Execute Program element is limited to server side configuration files.

Applications for the Execute Program element include the following:

- Integrate with an electronic control system, for example, door controls, alarm systems, or climate control systems, to retrieve information or to invoke changes to the system.
- Integrate with an Instant messaging (IM) client to provide another way of notifying callers of messages.
- Invoke an application that validates information received from a user, for example, a credit card transaction.
- Integrate with a third party display device.

Enabling programs to be executed within a service

You can select which programs can be executed from within a service by including them in a configuration file stored on the server. A sample configuration file, *programs.sample.xml*, has been provided for your use, and is available in the default location listed below.

The XML configuration file defines any programs that are permitted to be executed by a service. Copy and rename the sample file provided, and using a text editor such as Notepad, manually edit the file to define the following:

- The programs which can be executed from within a service.
- The complete path of the programs to be executed.

- A unique, logical name for the programs. The logical name is what is displayed in the Execute Program element, in the drop down list of programs to execute.
- A list of parameters to be passed to the application.
- The computer credentials to be used by the application. Refer to the *programs.sample.xml* file for more information on how to edit the configuration file.



The UC server monitors the configuration file for changes, so any configuration changes are automatically recognized.

Requirements

The configuration file must be called:

programs.xml

In order for the file to be used by the Application Services, the configuration file must be stored in the following folder:

Data\System

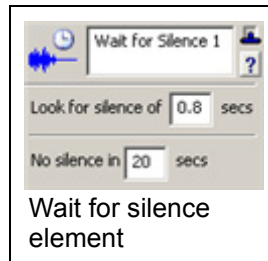
If the UC server was installed in the default location, then the path is:

C:\Program Files\ADTRAN\UC server\Data\System

To use the Execute Program element

1. From the Execute Program element, select a program to execute from the drop down list provided.
If your application does not appear in the list provided, then you must configure the configuration file on the server.
2. Enter the command line parameter names and values as required by the program selected.
Quotes will be inserted if necessary at run time.
 - The call flow exits through the Parameters connector pin if the return code from executing the selected program is zero.
 - The call flow exits through the Execution failure connector pin if the program executes, but returns a non-zero return code.
 - The call flow exits through the Otherwise connector pin if the program cannot be executed for any reason.

Wait for Silence element



The Wait for Silence element is typically used in an outdialing service to enable the call to wait for a period of silence.

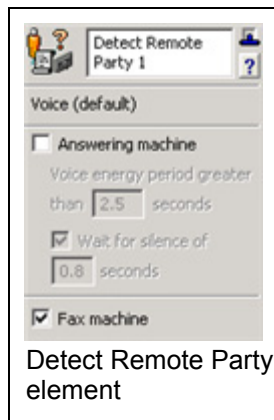
Look for silence in X secs

Determines whether a specified period of silence has been detected. If a continuous period of silence is detected, then the call exits through the silence connector pin. The default is 0.8 seconds.

No silence in X secs

If the specified time period elapses without continuous silence being detected, then the element exits through the No silence connector pin. If any other significant event occurs, such as the call being dropped, then the element exits through the No silence connector pin. The default value to wait for silence is 20 seconds.

Detect Remote Party element



The Detect Remote Party element is used to detect whether a call is answered by a person, an answering machine, or a fax machine. The element analyzes the incoming sound, and determines by the percentage of sound versus silence, whether it's a live voice or an answering machine, or alternatively, a fax machine. The Detect Remote Party element is typically used with outdial requests.



For more information on creating outbound dialing services, refer to [Creating Outbound Dialing Services](#) on page 206.

The purpose of the element is so that the call flow can be modified when a call is answered by an answering machine. Usually, this element is the first, or close to the first, element in a service. As an example, you may want to create a service that prompts callers to complete an audio survey. If a person

answers the phone, you can instruct them on how to complete the survey, and then questions can be responded to with dual tone multi-frequency (DTMF, also known as touchtone dialing) tones. If it is determined that the call has been answered by an answering machine, then you may want to leave alternate instructions instead.

When the element is first executed, the Voice connector pin is taken, unless the UC server has already determined that the remote party is a fax machine. As service execution continues, the UC server monitors the call to determine if the remote party is a fax or an answering machine. It will continue monitoring the call for the entire duration as specified by the voice energy period. If during this period it is determined that the called party is an answering machine, the default voice path is terminated, and the call flow exits the appropriate connector. If an answering machine is detected, it exits through the Answering machine connector pin when there is a minimum period of silence that exceeds the value defined in Wait for silence of.



You can setup the service so that if a call is answered by a fax machine, then a log entry is made to note that the number is a fax number, so that it can be removed from the list of numbers to dial in future, or alternatively you can configure a fax to be queued to be sent to the same telephone number. Also, you will want the service to disconnect the call if it is answered by a fax machine.

Voice (default)	By default the call is considered to be answered by a live voice, and exits through the Voice connector pin, unless it is determined to be answered by a fax or an answering machine.
Answering machine	Enable the Answering machine checkbox if you want the voice to be analyzed to determine whether the call is being answered by an answering machine.
Voice energy period greater than X seconds	The period of time the element analyzes the audio before making a determination of the caller. The default value is 2.5 seconds.
Wait for silence of	Enable this check box if you want the service to wait for the period of silence noted, before continuing with the service.
X seconds	If enabled, this is the period of time that silence must exist before proceeding with the Answering machine behavior. This is an assumed amount of time when the answering machine or voice messaging system stops speaking and plays a beep. The default period is 0.8 seconds, however this number can be modified as required.
Fax machine	If a fax machine is detected as answering the call, then the element immediately exits via the Fax machine connector pin.

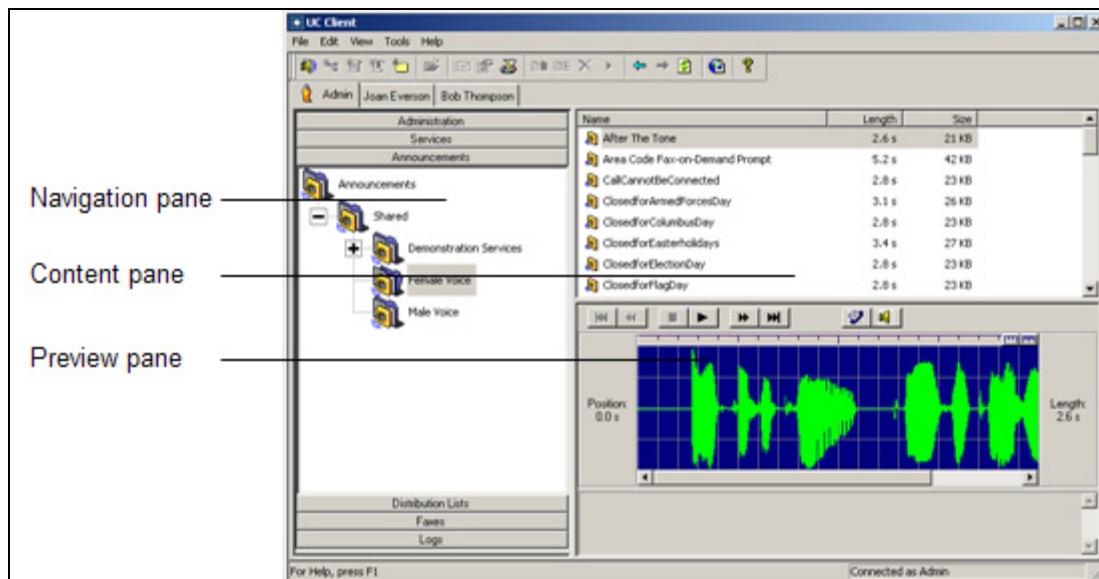
16 Using Announcements in Call Answering Services

This chapter provides information about how to select prerecorded announcements for the auto attendant. It also explains how to record personalized announcements using the Audio Editor. Make sure that there is a sound card installed on the computer that you are using. This chapter covers the following topics:

- [Overview on page 273](#)
- [Using Prerecorded Announcements on page 274](#)
- [Recording Announcements on page 275](#)
- [Managing Announcements on page 282](#)

Overview

Announcements are the outgoing messages that callers hear when they telephone the auto attendant. These announcements typically initiate the call flow in a service and are used to guide the caller through the call process. Announcements can inform callers of your business hours, explain menu options, or provide instructions about how to leave messages or request information by fax.



As system administrator, you are responsible for managing announcements and incorporating them into the services that you assign to identities.

The navigation pane presents folders used to store announcements. Select a folder to view its contents in the content pane. Below the content pane is the preview pane. Select an announcement to view its waveform representation in the preview pane. A variety of convenient, prerecorded announcements are available. You can also create custom announcements using the Audio Editor.

Using Prerecorded Announcements

Male and female versions of prerecorded announcements are supplied with the application. They are stored in the Male Voice and Female Voice folders on the navigation pane of the Announcements menu. For information about sharing prerecorded announcements, refer to *Importing and Exporting on page 311*.



You can get additional custom audio recordings that are in the same voices as the ones that ADTRAN supplies for system announcements and sample audio. Refer to the following website: www.voiceinternational.com.

Some of the elements that you use to build services include a Browse button that allows you to select an announcement to be played at that point in the call flow. Before selecting the announcements you want to play while the service runs, you can preview them in the Announcements menu.

To preview a prerecorded announcement



1. Select the **Announcements** navigation bar to get the Announcements menu.
2. Select a folder to list its contents in the content pane.
3. Select the announcement that you want to preview.
4. Select **Download** in the preview pane. A waveform representation of the announcement appears in the preview pane with a set of audio commands.
5. Select the computer speaker icon.
6. Select play to listen to the announcement.



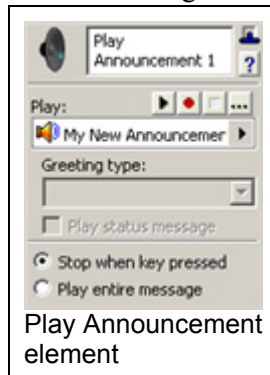
You can also preview announcements using the Service Editor while you are building a service. You must first select a Play Announcement element and then use the Browse button to choose an announcement to preview.

After you preview the announcements and note those that you want to use, open the service to which you want to add prerecorded announcements. Locate an element that you want to change, or add one to the service.

To select a prerecorded announcement for an element

1. Select the **Services** navigation bar to obtain the Services menu.
2. Double-click a service to open it and start the Service Editor.

3. Select a **Play Announcement** element and then drag it to the canvas.



4. Select **Browse** to open the Play Announcement dialog box, which displays a list of your announcement folders.
5. Double-click the prerecorded announcement folder that you want to open. The folder's announcements are listed.
6. Double-click the announcement that you want to select for the element in your call flow. The title of the announcement appears in the text area of the element.

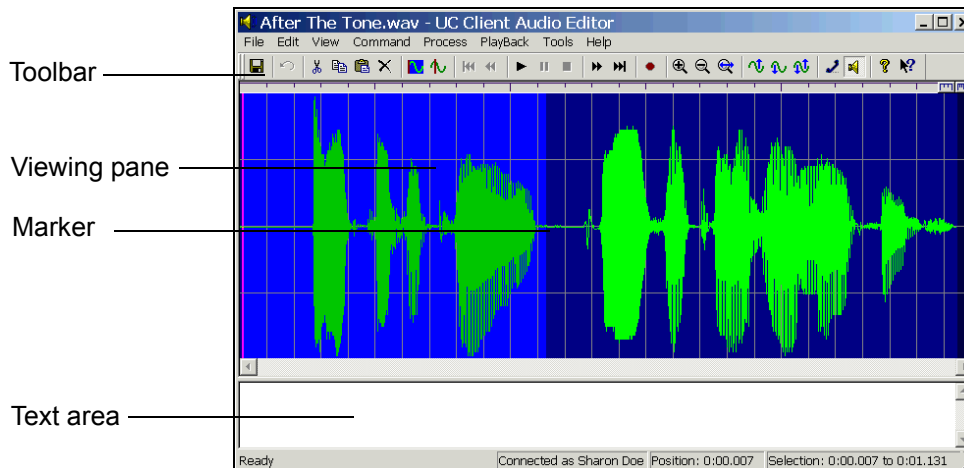


Use the Go back arrow on the Play Announcement dialog box to return to the previous folder.

Recording Announcements

You can create custom announcements to be played on the auto attendant. These announcements can also be copied, renamed, or deleted according to your needs.

Custom announcements are created using the Audio Editor, shown below.



The Audio Editor has a Viewing pane and a Text area.

The **Viewing pane** contains a waveform representation of your announcement. As you play an announcement using the audio commands, the vertical announcement **Marker** moves across the viewing pane from left to right so that you can identify the exact location of different parts of your recording. You can edit any part of your announcement by manipulating the waveform and using the audio commands. You can also remove parts of the message that are no longer required.

The **Text area** allows you to type your announcement before recording it. You can read the message from the menu while recording to help you eliminate pauses in your announcement.

The **Toolbar** contains visual shortcuts to the most frequently used menu items. Hold the pointer over a button to show the tool tips. The toolbar contains **View**, **Select**, and **Process** commands, as well as standard **Audio** commands found on most applications.

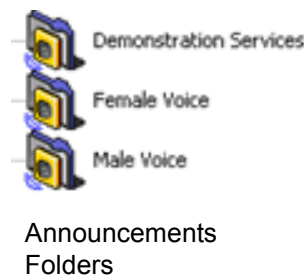


Make sure that there is a sound card installed on the computer that you are using.

Creating and recording announcements

To create an announcement

1. Select the folder where you want to store your new announcement. The folder's contents are listed in the content pane.
2. Right-click anywhere in the content pane.



3. Select **New > Announcement**. A new announcement is added to the list in the content pane. The announcement icon denotes the new announcement.
4. Type a name for the announcement.

To record an announcement

1. Create an announcement and give it a name by following all the steps in [To create an announcement on page 276](#).
2. Double-click the announcement that you want to record. The Audio Editor starts.



In the text area of the Audio Editor, you can type the announcement that you want to record. This is useful for eliminating pauses when you are recording.

3. On the toolbar, select record to open the Record Audio dialog box.
4. Select the computer microphone. The icon is enclosed by a dotted line.
5. In the Insert new audio box, select **at the beginning**.
6. Select record and then record your message. As you record, the clock runs and the red record button turns grey.



The maximum length of a recorded announcement is 15 minutes.

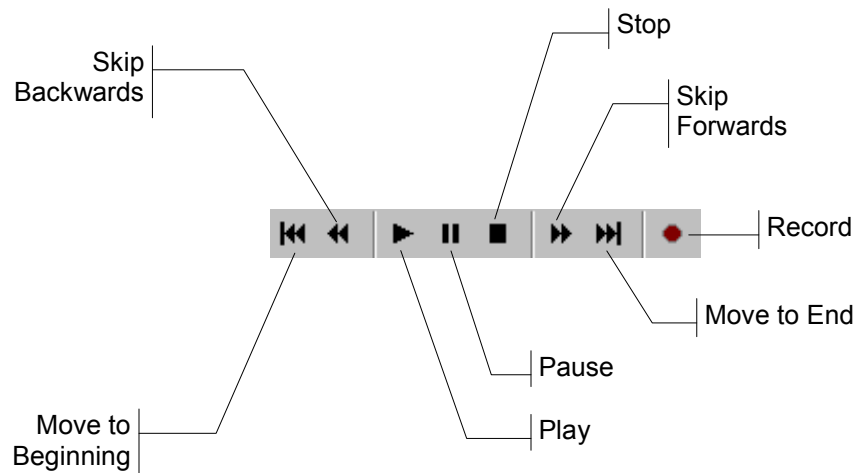
7. Select stop when you are finished.
8. Select **OK** to keep the announcement.
9. Select **File > Update**.

Changing announcements

You can change an announcement by deleting a part that you no longer need, adding recorded audio to it, or adjusting the volume. Delete portions of the waveform by selecting a portion in much the same way as you select text using a word processor. A number of commands allow you to change announcements. The following illustrations provide details about the command icons located on the Audio Editor toolbar.

Audio commands

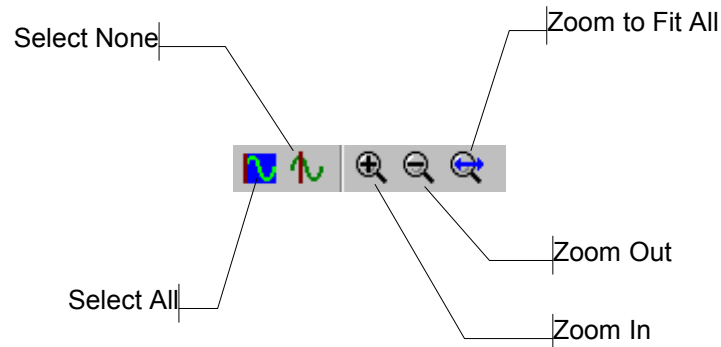
Use the audio commands to play announcements, position the marker in the waveform, and make recordings.



Before you delete part of an announcement or add to it, you can listen to it and identify the location on the waveform where you want to make a deletion or insertion. If you have selected a part of the waveform, audio commands apply only to the selected portion. The **move** and **skip** commands are used to position the marker within the selection.

View and select commands

Use the view and select commands to select a portion of the audio.



The **view** commands allow you to zoom in and out when you view the waveform so that your selection is as accurate as possible.

The **select** commands allow you to select and deselect the entire waveform. If you do not make a selection, the changes that you make apply to the whole waveform.

To select part of the waveform

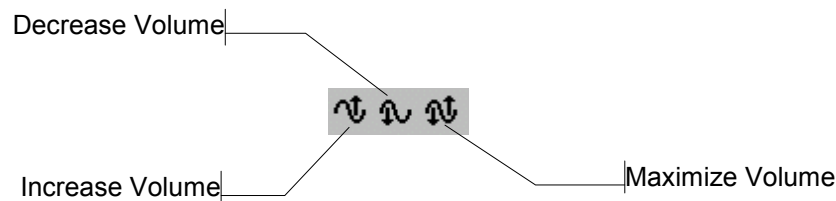
1. Select anywhere in the waveform.
2. Drag across the portion of the audio that you want to select.



Deselect the waveform by selecting anywhere in the waveform.

Process commands

Use the process commands to adjust the volume.



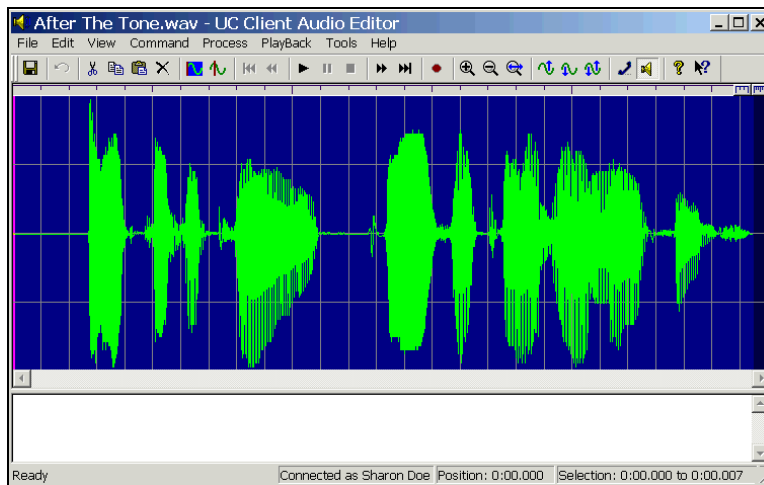
To delete part of an announcement

1. Select the announcements folder that contains the announcement part you want to delete. The announcements contained in the folder are listed in the content pane.



Announcements
Folder

2. Double-click the announcement that contains the part you want to delete. The Audio Editor displays the announcement as a waveform.

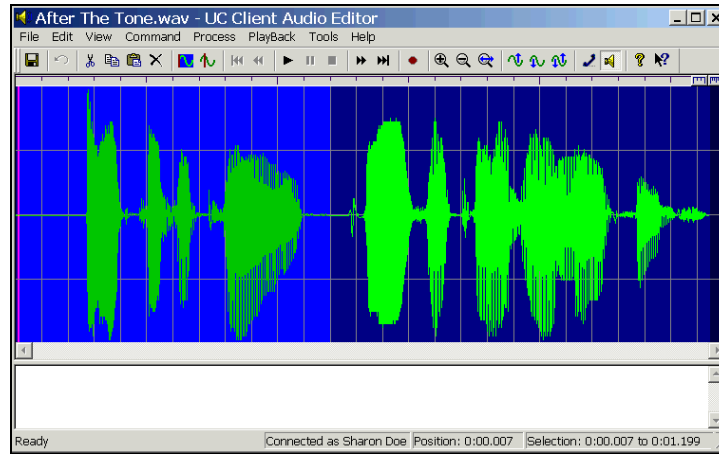


3. Select **View > Zoom to Fit All**.
4. Select the computer microphone icon.
5. On the toolbar, select **play**. Listen to the announcement once or twice to pinpoint the portion of the waveform that you want to delete.
6. Select **stop** on the toolbar to stop playback before the part that you want to delete. Note the marker's position in the Position indicator at the bottom of the menu.

Position: 0:00.000

Position Indicator

7. Select the waveform where you want to begin deleting and drag to select the portion of the announcement that you want to delete.



If you do not select part of the waveform, the changes that you make affect the entire waveform.

8. Select **play** to review the selected portion of the announcement.

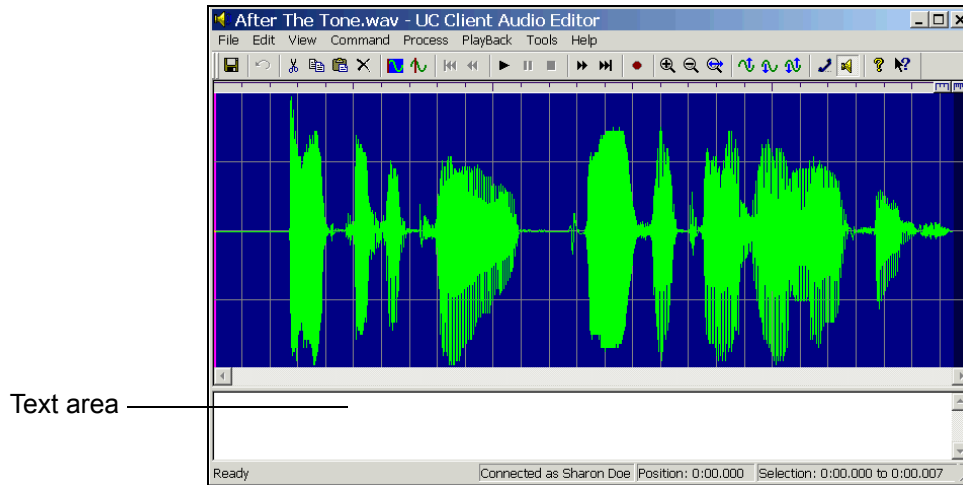


After you select part of an announcement, the audio commands apply only to the selected part.

9. Select **Erase** on the toolbar of the Audio Editor to delete the selection.
10. Select **File > Update**.

To add to an announcement

1. Select the announcement folder that contains the announcement that you want to add to. The announcements contained in the folder are listed in the content pane.
2. Double-click the announcement that you want to add to.
The Audio Editor displays the announcement as a waveform.



In the text area of the Audio Editor, you can type the additional text that you want to add to your recording. This is useful for eliminating pauses when you are recording.

3. Select **View > Zoom to Fit All**.
4. Select the computer microphone icon.
5. On the toolbar, select **play**. The announcement plays. Listen to it once or twice to pinpoint the portion of the waveform to which you want to add. Position the pointer over the stop command on the toolbar so that you can stop the playback just before the part to which you want to add. Use the view commands for accuracy. Note the marker's position in the Position indicator at the bottom of the menu.
6. Select **stop** on the toolbar to stop playback on the waveform at the point where you want to add to the announcement.
7. On the toolbar, select **record** to open the Record Audio dialog box.
8. Select the computer microphone icon.
9. In the Insert new audio text box, select **at the current position**.
10. Select **record** and then record your message. As you record, the clock runs and the red record button turns grey.

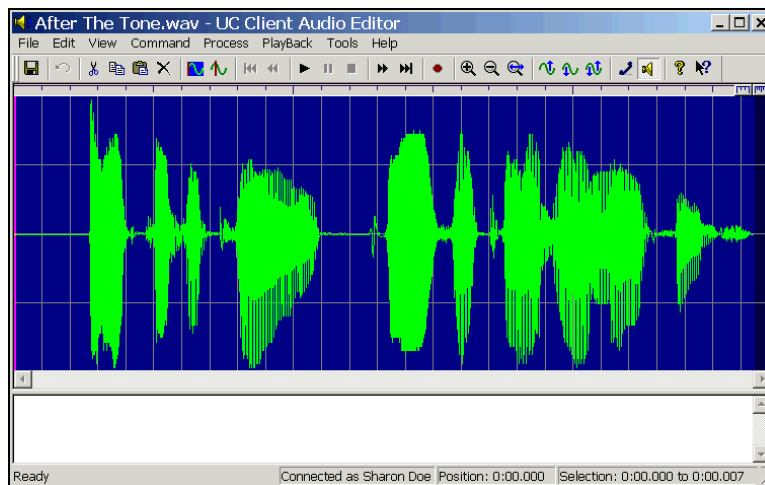


The maximum length of a recorded announcement is 15 minutes.

11. Select **stop** when you are finished.
12. Select **play** to review the announcement.
13. Select **OK** to keep the announcement.
14. Select **File > Update**.

To change the volume of a recording

1. Select the announcements folder that contains the announcement of which you want to change the volume. The announcements contained in the folder are listed in the content pane.
2. Double-click the announcement that you want to change. The Audio Editor displays the announcement as a waveform.



3. Select **Edit > Select All**.
4. On the **Process** menu, select one of the following:
 - **Increase Volume**
 - **Decrease Volume**
 - **Maximize Volume**

It is advisable to maximize the volume to make sure that your recording is audible to callers.

5. If you select **Increase Volume** or **Decrease Volume**, type the percent amount by which you want to increase or decrease the volume of the announcement, and then select **OK**.
6. Select **play** to review the changes.
7. Select **File > Update**.

Managing Announcements

You can organize announcements more efficiently by creating folders for them.

To create a folder for announcements

1. Right-click anywhere in the content pane.
2. Select **New > Folder**. A new folder is added to the list in the content pane.



Announcements
Folder

3. Type a name for the folder.

To put an announcement in a folder

Select the announcement that you want to store and drag it to the folder where you want to store it.



Double-click a folder to view its contents. On the toolbar, use the Go back arrow to return to the list previously displayed in the content pane.

To copy an announcement

1. Select the announcements folder that contains the recording that you want to copy. The announcements stored in the folder are listed in the content pane.
2. Right-click the announcement that you want to copy.
3. Select **Duplicate**. A copy of the announcement appears in the content pane.

To delete an announcement

1. Select the announcements folder that contains the recording that you want to delete. The announcements stored in the folder are listed in the content pane.
2. Right-click the announcement that you want to delete.
3. Select **Delete**.
4. Select **Yes** to confirm the deletion.

To rename an announcement

1. Select the announcements folder that contains the recording that you want to rename. The announcements stored in the folder are listed in the content pane.
2. Right-click the announcement that you want to rename.
3. Select **Rename**. Type a new name for the announcement and then select its icon to make the change.

17 Using Databases in Call Answering Services

This chapter introduces how to use access database information using database elements in call answering services, and includes the following topics:

- [Overview on page 285](#)
- [Managing Data Sources on page 285](#)
- [Accessing Data using Database Service Elements on page 297](#)
- [Using Database Elements on page 300](#)



To access database information the system must be licensed for database integration. If you are not licensed for database integration then the Database tab of elements is not visible.

Overview

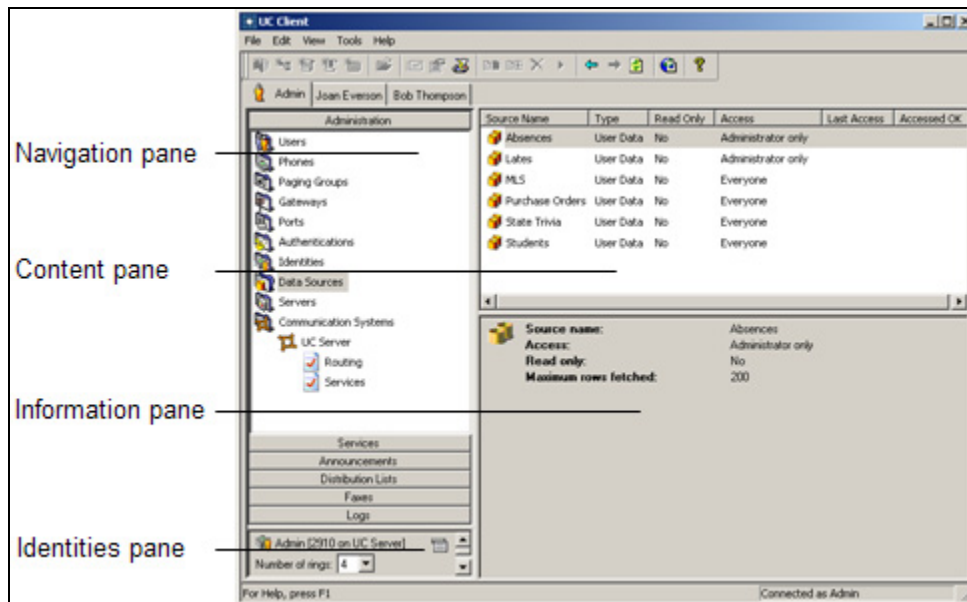
You can use the UC server to access dynamic database information using services. A user with access to the Administrator tab can create user data tables. Each user has a copy of the table where they can save their personal data. All the user data tables that are used in your active services are displayed in the Assistant pane.

For information about creating user data tables, refer to [Managing Data Sources on page 285](#)

Managing Data Sources

Using the Administrator tab, you can create, and users can manage, data sources. Data sources allow users to access and change information using the computer or over the telephone. For example, a real estate agent can create a data table that includes information, such as address or price, for each property that the agent has listed. While away from the office, the agent can call and access the data table to add, change, or

delete information about a particular property.



Data sources are set up like tables. Each table row corresponds to a data row or record. Each table column corresponds to a field (for example, a name, address, or fax).

There are two types of data sources:

- **User Data Sources**—the data is managed by the UC server, and users have their own copy of the data.
- **External Data Sources**—the data is managed by another server or process (for example, a relational database management system), and is shared by UC server users and others. External data is accessed using Microsoft's ODBC (Open Database Connectivity), so these sources are typically called ODBC database sources.



Installing a database directly on the UC server is strongly discouraged. Doing so might cause performance problems.

Using the Administrator menu, you can manage the following aspects of data sources:

- Defining ODBC data sources
- Defining user data sources
- Managing data in user data tables

When you define ODBC data sources, the definition includes table and field information. It also includes security information, which allows you to define a user name and password to access the database.

User data tables allow a user to manage personal data without setting up an entire database. After the Administrator creates data tables, users can manage them by adding, changing, and deleting the rows within the tables.

A set of elements allows users to access information in data sources while defining a service. These elements allow users to retrieve data, manage data sources, and select from sets of values. Most elements allow users to use information retrieved from data sources.



The data sources feature is available only if database integration is licensed.



To make a user data source available to a user, it must be accessed by an element in the user's active service.

Defining ODBC data sources

To use an ODBC data source, you must define an ODBC Data Source Name (DSN) that uses the related driver. The DSN name, user ID, and password must be configured to connect to the source.



You can define an ODBC data source by using the Administrator utility provided with Microsoft Windows.

After defining a DSN, you can use the Data Source Editor to define ODBC (Open Database Connectivity) data sources. ODBC data sources include relational databases and other kinds of data, for example, Excel spreadsheets, SQL server, or Visual FoxPro database.

You can edit the field attributes of an ODBC data source. For example, you can change the name of the field, the data type, or whether you want the field to be visible to the user. Other options include filtering and sorting ODBC data.

To name a new ODBC data source

1. Right-click anywhere in the Data Sources content pane.
2. Select **New > ODBC Source**.
3. In the **Name** box, type a name for the data source that you want to define.
4. If you do not want to further define the data source, you can select **OK** to save it.



To use the data source, you must define ODBC connection parameters.

To change the access status of a data source

1. In the Data Sources content pane, right-click the data source that you want to change the access status of.
2. Select **Access**.
3. Select one of the following access types:
 - **Disabled**—does not allow access to the data source
 - **Admin Only**—allows only the Administrator to access the data source
 - **Everyone**—allows everyone to access the data source
4. If you selected Everyone, select **Yes** to confirm.



Allowing everyone to have access to an ODBC data source is a potential security risk. Make sure that you want to allow all callers to access the data source.

To define ODBC connection parameters

1. In the Data Sources content pane, double-click the ODBC source that you want to define.
2. Select the data source that you want from the ODBC DSN list.
3. In the **User ID** box, type the user ID required to access the ODBC data source.
4. In the **Password** box, type the password to access the ODBC data, if required. Some ODBC DSNs contain the user ID and password so you do not have to enter that information.
5. Select the **Connect** button to connect to the ODBC data source.



An icon beside the Connect button indicates a successful or unsuccessful connection. You must have the correct user ID and password to successfully connect.

To define advanced ODBC connection options

1. In the Data Sources content pane, double-click the ODBC source that you want to define.
2. Select the **Advanced Connection Options...** button.
3. From the **Timeout** list, select the number of seconds that you want ODBC to wait for a connection and successful login. The default timeout is 15 seconds.
4. In the **Maximum rows to fetch** box, type the number of rows that you want the application to fetch during a query. The default number of rows is 200.
5. Type the **Extra Drive Specific Attributes To Be Added To The DSN** in the box, if required.
6. Select **OK**.

To define ODBC tables and fields

1. In the Data Sources content pane, double-click the ODBC source that you want to define.
2. From the **Table** list, select the table that you want to associate with the data source.

3. From the **Available Fields** list, select the fields that you want to include.
4. Select the **Select>>** button to include the selected fields.
5. Repeat steps 4 and 5 until all the fields that you want are included.



*If you want to include all the available fields, select the **All>>** button.*



*If you want to exclude fields that you have included, select the fields and select the **<<Selected** button, or select the **<<All** button to exclude all the selected fields.*

6. Select OK.

To change ODBC field attributes

1. In the Data Sources content pane, double-click the ODBC source that you want.
2. From the **Table** list, select the table that contains the fields that you want to change.
3. In the **Included Fields** list, select the value for the field attribute that you want to change.
4. In the new window, type the change.
5. Select outside the window to close it.
6. Select **OK** to save the changes.

Filtering and sorting ODBC data

A Fetch Data or Dial for Data element can retrieve more than one row depending on the number of rows in the source, and your filter conditions. When one or more rows are fetched, a reference to a field by another element is always a reference to the field's value in the "current" row, in the set of rows fetched. The Move Current Row and Prompt for Current Row elements can be used to change which row is current. Immediately after a Fetch Data or Dial for Data element is executed, the current row is the first row.

- When only one row is fetched, the first row is the single row that was fetched.
- When multiple rows are fetched, which row is first depends on whether you defined a sort in the data source definition.

Defining a sort order

If you define a sort order in a data source, this defines the default order of rows fetched by a Fetch Data or Dial for Data element. However, a Fetch Data element can override this order.

When an ODBC data source has a defined sort order, the sorting is done by the ODBC server itself, and not by the UC server. In most cases this distinction is unimportant, however some data types supported by the UC server are not directly supported by ODBC. Thus in some cases, the sort order produced by the ODBC server is not the same as what would have been produced by the UC server.

- If no sort order was defined, then the order of the rows is determined by the data source, and you should not make any assumptions about the order.
- With a defined sort order, the first row is the first row in the sort order specified. However, if the sort field contains duplicate values, the order of rows with the same sort value is determined by the ODBC server and not by the UC server, and you should not make any assumptions about the order.

For example, assume that you have a data source with a Date field, and that each row in the data source has a unique date value. If the data source defines a sort order based on this Date field and descending order, then any Fetch Data element that fetches from the data source will retrieve rows in descending date order. In this case, the sort order produced by ODBC is the same as the sort order defined by the UC server data type. If a Fetch Data element fetches multiple rows, then the first row will be the row in the fetched set of rows with the newest value in the Date field. Thus another element referencing this Fetch Data element would always reference the newest row. Without the sort, you cannot predict which row in the retrieved set would be first.

In another example, a data source containing a field defined to be a character string is defined as Day of Week in the ODBC data source. A sort order would return the values by character comparison, resulting in the following order: Friday, Monday, Saturday, Sunday, Thursday, Tuesday, and Wednesday. Day of Week in the UC server would return the results in the following order: Sunday, Monday, Tuesday, Wednesday, Thursday, Friday and Saturday.

Thus, the actual sort order, as produced by ODBC, is not the same as the order that you would get from sorting using the UC server data type.

Defining a sort order using the NetVanta UC Server

You can specify that the UC server sorts the rows fetched by the Fetch Data or Dial for Data elements, by specifying a sort order in a second Fetch Data element. Sorting using an explicit sort defined in a Fetch Data element is always done by the UC server.

To define ODBC filtering and sorting

1. In the Data Sources content pane, double-click the ODBC source that you want to define.
2. From the **Filter Field** list, select the field that you want to test.
3. In the **Filter Value** box, type the value that you want to compare to.
4. From the **Sort Field** list, select the field that you want to sort.
5. From the **Sort Order** list, select one of the following:
 - **ascending**—sorts information in ascending order
 - **descending**—sorts information in descending order
6. Select **OK**.

Defining user data sources

The User Data Source Editor allows you to create user data tables. Display options allow you to define how the user data table is displayed for the user. Decide which fields you want to include, and how they are sorted and retrieved. A user data source is defined once, and each user, including the Administrator, has an

individual copy of the table.

User Data Source - House Listings

Data source properties

Name:

Description:

Instructions:

"Add button" label:

Database fields

Seq	Name	Data Type	Decimals	Default Value	Display Width
1	ListingID	Number	0	0	12
2	Address	Text		Specify the address	30
3	# Bedrooms	Number	0	2	15
4	Price	Number	0	0	10
5	FAX Description	Fax			20
6	Audio Description	Audio			20

Sorting and retrieval

Sort field:

Sort order:

Maximum rows:

OK Cancel Help

Users cannot share user data tables; each user adds data and maintains individual data tables. Only the Administrator and Personal Business Assistant users can edit the data in their copy of a user data table.



To make a data source available to a user, an element in the user's active service must access the data source.

Other options include sorting and retrieving user data.

To create a new user data source

1. Right-click anywhere in the Data Sources content pane.
2. Select **New > User Data**.
3. Type a name for the new user data source in the Name box.
4. If you do not want to further define the user data source, you can select **OK** to save it.

To edit user data source display options

1. In the Data Sources content pane, double-click the user data source that you want to edit.
2. In the **Description** box, type the title of the user data source.
3. In the **Instructions** box, type the instructions that you want the user to follow when using the data source.
4. In the **Add Button Label** box, type the title that you want to display on the user add button.
5. Select **OK** to save the changes.

To add a field in a user data source



If you add a new field to an existing table, then the value of that field for all existing records in that table will be set to a null value. This appears in the user interface as [NO VALUE].

1. In the Data Sources content pane, double-click the user data source that you want to add a field to.
2. Select the **New** button.
3. Select the **Name** section of the new field.
4. Type a name for the field.
5. Select the **Data Type** section of the new field to obtain a list of data types.
6. Select a data type from the **Data Type** list.



*The **Default Value** section changes depending on the data type that you select.*

7. Repeat steps 3 to 7 until you have added all the fields that you want.
8. Select **OK**.

To change the order of a field in a user data source

1. In the Data Sources content pane, double-click the user data source that you want to change.
2. Select the field that you want to change the order of.
3. Select one of the following buttons:
 - **Up**—moves the selected field up one row in the list of fields
 - **Down**—moves the selected field down one row in the list of fields
4. Select **OK**.



*The field order defines the order in which the fields are displayed in the **User Data** edit window, which is used to edit and change the entries in a user data table.*

To delete a field in a user data source

1. In the Data Sources content pane, double-click the user data source from which you want to delete a field.
2. Select the field that you want to delete.
3. Select the **Delete** button.
4. Select **OK**.

Sorting and retrieving user data

A Fetch Data or Dial for Data element can retrieve more than one row depending on the number of rows in the source, and your filter conditions. When one or more rows are fetched, a reference to a field by another element is always a reference to the field's value in the current row, in the set of rows fetched. The Move Current Row and Prompt for Current Row elements can be used to change which row is current.

Immediately after a Fetch Data or Dial for Data element is executed, the current row is the first row.

- When only one row is fetched, the first row is the single row that was fetched.
- When multiple rows are fetched, which row is first depends on whether you defined a sort in the data source definition.

Defining a sort order

If you define a sort order in a User Data data source, this defines the default order of the rows fetched by a Fetch Data or Dial for Data element. However, a Fetch Data element can override this order. Sorting is always done by the UC server for this type of data source.

- If no sort order was defined, then the order of the rows is determined by the data source, and you should not make any assumptions about the order.
- If a sort order is defined, the first row is the first row in the sort order specified. However, if the sort field contains duplicate values, the order of rows with the same sort value is determined by the data source, and you should not make any assumptions about the order.

For example, assume that you have a data source with a Date field, and that each row in the data source has a unique date value. If the data source defines a sort order based on this Date field and descending order, then any Fetch Data element that fetches from the data source will retrieve rows in descending date order. If a Fetch Data element fetches multiple rows, then the first row will be the row in the fetched set of records with the newest value in the Date field. Thus another element referencing this Fetch Data element would always reference the newest row. Without the sort, you cannot predict which row in the retrieved set would be first.

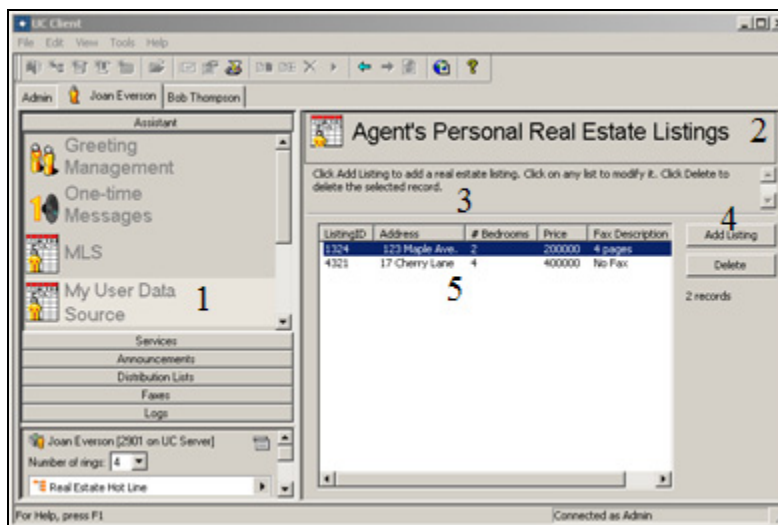
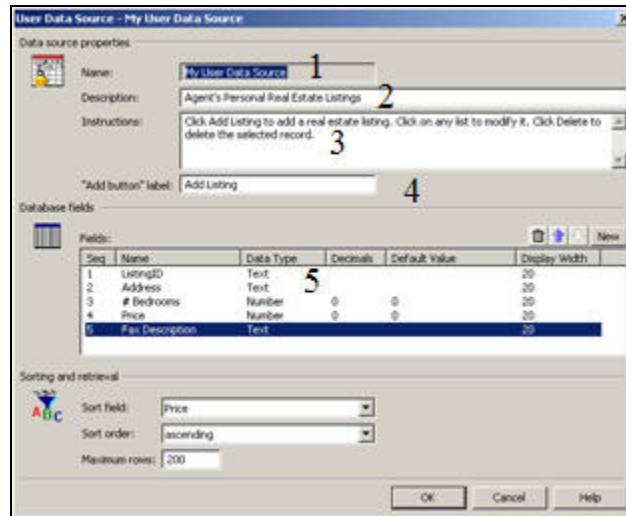
To define user data source sorting and retrieval options

1. In the Data Sources content pane, double-click the user data source that you want to define.
2. From the **Sort Field** list, select the field that you want to sort.
3. From the **Sort Order** list, select ascending or descending.
4. In the **Maximum Rows** box, type the number of rows that you want to retrieve for each query.
The default number of rows is 200.
5. Select **OK**.

Managing data in user data tables

After you create a user table, you can add, change, and delete the information included in your copy of the

table.



The information—represented above by the numbers 1 to 5—that you enter when you create a user data table in the administrator application (on the top), appears in the user’s copy of the table (on the bottom).

NOTE *The user data tables are only visible if a service that is associated with an identity, has an element in the service that uses the data source. This allows the administrator to create multiple user data sources, without users getting confused by information that is not relevant to them.*

To add information to a user data table

1. In the Data Sources content pane, right-click the user data source that you want to add information to.
2. Select **Edit** to open the user data source.

3. Select the **Add** button to add a row to the table. Each field has the default value defined in the data source definition.

**NOTE**

*The **Add** button is editable and might have a different title.*

4. Select **Close** to close the table.

To change information in a user data table

1. In the Data Sources content pane, right-click the user data source that you want to change.
2. Select **Edit** to open the user data source.
3. Select the field value that you want to change to open a new window.
4. In the new window, type the change.
5. Select outside the window to close it.
6. Select **Close** to close the table.

To delete a row in a user data table

1. In the Data Sources content pane, right-click the user data source in which you want to delete a row.
2. Select **Edit** to open the user data source.
3. Select the row that you want to delete.
4. Select the **Delete** button.
5. Select **Close** to close the table.

Data Type Definitions

Data Type	Value	Examples
Text	A string of characters	123 Maple Avenue
Number	A number with optional sign, decimal places, and commas	160,000.00
Logical	A string whose value can be only true , false , 1 or 0 . The numbers, 1 and 0 , correspond to true and false, respectively.	True False 1 0
Phone Number	Any string that can be parsed as a phone number into canonical form. A canonical telephone number represents a dialable number that can be understood and interpreted around the world, for example: +Country Code (Area Code) Subscriber Number In North America, the following is an example of a canonical telephone number: +1 (613) 555-1212	+1 (613) 555-1212 6135551212
Audio	The value of the field in valid .wav format	PropertyDescription.wav
Fax	The value of the field in valid .tif format	Data Sheet.tif
Day of Year	A date without a year, for example, January 1 is New Year's Day	September 16 09-16 Sep 16
Date	A date that includes the day, month, and year. Valid formats include YY-MM-DD, YYYY-MM-DD, and text versions as illustrated in the examples.	September 16, 2007 Sep 16, 2007 07-09-16 2007-09-16
Day of Week	The full day name in English, as a string. The strings can also be in abbreviated formats.	Sunday Sun
Time of Day	A time that includes the hour, minutes, and seconds (optional). Valid formats include HH:MM AM/PM. The AM/PM is optional, however AM is assumed unless the hour is greater than or equal to 12. For example, 12:02 means 12:02 PM.	2:00 PM
Time Range	Two times of day separated by a hyphen, which indicates a time range. Valid formats are HH:MM AM/PM - HH:MM AM/PM, with AM/PM being optional. As above, AM is assumed unless the hour is greater than or equal to 12. For example, 1:00 means 1:00 AM.	2:00 PM - 4:00 PM

Accessing Data using Database Service Elements

Using database integration, you can set up services that allow callers to access current information. Also, the service creator can call in to a service to change information remotely at any time, so that callers have access to the most current information. Data accessed by data sources is dynamic and can be updated at any time.

Some elements, depending on their function, allow you to specify values or data sources. Database elements use database integration to provide information to callers and receive information from callers during a service. For example, callers to a real estate office can hear updated information about a particular property when they enter the property number over the phone during a service. In this case, the Fetch Data element processes the query and searches the database for a match.

Likewise, the real estate agent can make changes to the property information in the database by accessing the service remotely and entering the new information to the database. If the agent wants to change the list price for a property, he can access the Update Data element in the service, access the price value for the correct property, change the value, and update the database accordingly.

For more information about accessing data sources using service elements, refer to [Using Elements to Create Call Answering Services on page 225](#).



To troubleshoot services using data sources, refer to [Troubleshooting Services on page 223](#).

Using the Data Source Selector

You can use the Data Source Selector feature to specify a data source, and to filter the data that is returned. The Data Source Selector is available for elements that require access to database information as part of their functionality, such as the Fetch Data, Update Data, and Dial for Data elements.

You can filter data by selecting a data source field, a comparison operator, and a value. You can filter the data by selecting the following comparison options:

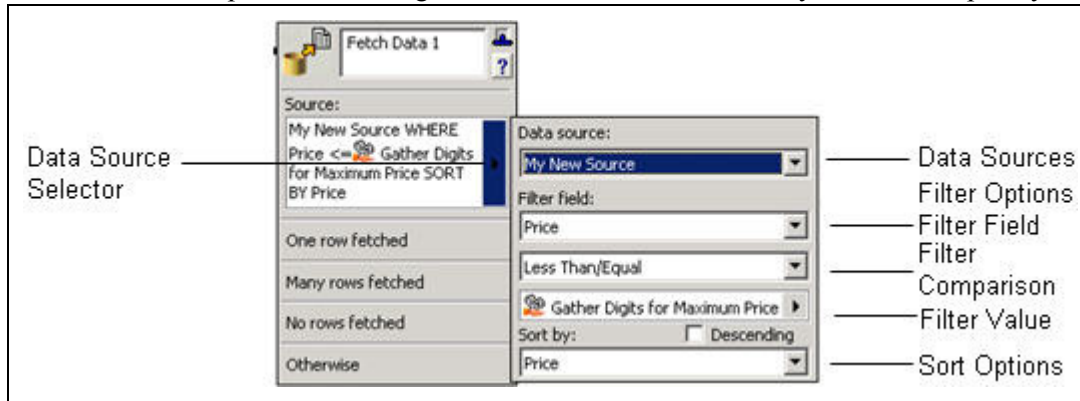
- Equals
- Less Than
- Less Than/Equal to
- Greater Than
- Greater Than/Equal
- Between
- Not Equal

Using the value selector you can specify the value that you want to compare to the filter field.

An easily recognizable icon appears in the Source dialog box to indicate the type of data that you are using for the comparison value. For example, the Gather Digits icon appears with the name of the Gather Digits

element that is included in the service.

The values that are available depend on the data source that you choose; the other elements included in the call flow, for example, a Gather Digits element; and the values that you have set up for your system.



To specify a data source for an element

1. Select the arrow in the Source section of the element.
2. Select the Data Source that you want from the list.
3. Select the Filter Field that you want from the list.
4. Select the comparison mode that you want. In the example above, “Less Than/Equal” was selected.

NOTE *If you select “Between” A and C for example, then the values A and C are included.*

5. Select the value type that you want. For example, Database Values, Gather-Digits Values, or System Values.

NOTE *You can also type a filter value manually in the filter value box.*

6. Select a specific value from the value types list.

NOTE *The values that are available depend on the data source; the other elements that precede the element in the call flow, for example, a Gather Digits element; and the values that you have set up for your system.*

7. Select a sort mode, and enable Descending if you wish the results to be sorted in a descending order. When you choose a System Value in an element of a service, the information is taken from the current call

and from within your system and used, for example, to fetch a value for a Fetch Data element.

Table 17–1: System values

Name	Description	Example Format
Today	A UC server Date value representing today's date.	December 25, 2006
Current Time	A UC server Time of Day value representing the current time.	Format depends on location. For US and Canada, 12:02 PM.
User Name	The user profile name of the profile that owns the called extension.	Paul Smith
User Extension	The extension number where the call was initially received.	224
Selected User Name	If a Select Extension element is executed, this is the user profile name of the profile that owns the selected extension. Otherwise, it is User Name.	Mark Jones
Selected Extension	If a Selected Extension element is executed, this is the selected extension. Otherwise, it is User Extension.	225
Caller Name	The same as Matched Contact Name, if the matched contact name is not blank. If it is, the value of Caller-Id.	Paul Smith
Caller-Id	A UC server Text Value representing the caller. Typically, it is the form "name [number]". However, if the name is blank, it is in the form "number". It might also have special values, such as <Private Number>, <Long Distance> or <Unknown Caller>. Phone number format depends on the underlying transport.	Paul Smith [613-599-9698]
Caller-Id Name	A UC server Text Value containing the caller name as provided by the underlying transport.	Paul Smith
Caller-Id Number	A UC server Text Value containing the raw caller telephone number as provided by the underlying transport.	613-599-9698
Caller-Id Canonical Number	A UC server Phone Number Value containing the complete, formatted number.	+1 (613) 599-9698
Caller-Id Area Code	A UC server Text Value containing the area code of the number.	613
Caller-Id Area Code/Exchange	A UC server Text Value containing the area code and the exchange.	613599
Caller-Id Exchange	A UC server Text Value containing the exchange.	599
Caller-Id Local Number	A UC server Text Value containing the local number without the area code and country code.	5999698

Table 17–1: System values (Continued)

Called-Id	A UC server Text Value representing the called party. Typically, it is the form “name [number]”. However, if the name is blank, it is in the form “number”. It might also have special values, such as <Private Number>, <Long Distance> or <Unknown Caller>. Phone number format depends on the underlying transport.	Mark Jones [613-599-9698]
Called-Id Name	A UC server Text Value containing the called party name as provided by the underlying transport.	Mark Jones
Called-Id Number	A UC server Text Value containing the raw called party’s telephone number, as provided by the underlying transport.	613-599-9698
Port Name	The name of the port the call is using.	UC server
Trunk Name	The name of the trunk the call is using.	Admin Trunk 1
Current Date/Time	A UC server Text Value containing the current date and time	Format depends on the location of the computer, but is typically: December 25, 2006 12:02 PM
Matched Contact Name	The caller is matched to the called party’s contacts. This is a UC server Text Value containing the contact display name (or blank if no match).	Sharon Allen
Next Unique Id	A UC server Text Value containing a globally unique string (36 characters long). This value can be referenced only using the Add Data element and is used to assign a unique value to a database field. The Add Data element is typically followed by a Fetch Data element accessing the same table and using the Last Unique Id value to fetch the record just added. Do not depend on the specific format of these unique values	Although the actual format is not important, it is typically the following: 0E49D111-55AC-4794-9361-1B7320D694BD
Last Unique Id	A UC server Text Value containing the most recent unique string generated by Next Unique Id.	Refer to Next Unique Id.

Using Database Elements



To access database information the system must be licensed for database integration. If you are not licensed for database integration then the Database tab of elements is not visible.

Database elements rely on database integration to provide data to callers during a service. Database integration allows services to be dynamic and data driven.



The Database elements tab is shown above. A brief description is provided for each element available to the Administrator.

Table 17–2: Database elements

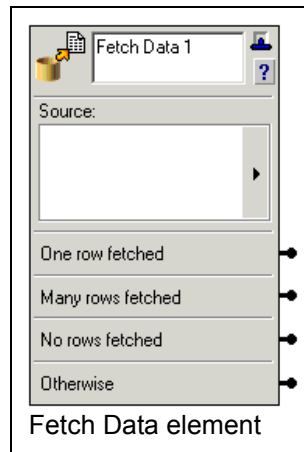
Fetch Data element on page 301	This accesses the specified data source and fetches the required data
Dial for Data element on page 303	This accesses the specified data source and fetches the data specified by the caller
Add Data element on page 305	This allows a caller to add data to a specified data source
Delete Data element on page 306	This allows a caller to delete data from a specified data source
Update Data element on page 306	This updates the data contained in a specified data source
Move Current Row element on page 307	This allows callers to modify which row is the current row in a multi row set, retrieved by either the Fetch Data or Dial for Data elements.
Prompt for Current Row element on page 308	Using a series of prompts, this allows callers to use the keypad to select which row becomes the current row.

Fetch Data element

The Fetch Data element allows you to specify a data source using the Data Source Selector, and query the data source so that the UC server fetches the data that you want.

Use a second Fetch Data element to further filter or sort the results of the first, a third to further filter or sort the results of the second, and so on. For example, a real estate service is set up to access a database of real estate listings. A Fetch Data element is specified to fetch all rows that match a certain price range. If several rows are fetched that match that criterion, the call moves through the Many rows fetched connector pin to another Fetch Data element that further searches those listings and fetches rows based on the number of bedrooms, until it finds the single listing that matches all criteria.

Fetch Data results



- If the Fetch Data element fetches one row, the call exits from the **One row fetched** connector pin.
- If the Fetch Data element fetches more than one row, then the call exits from the **Many rows fetched** connector pin.
- If no rows of data are fetched, then the call exits the **No rows fetched** connector pin.
- If any other condition exists, such as an inability to access a Data Source (such as an external database or internal User Data Table), then the call exits the **Otherwise** connector pin.

Sorting data rows

A Fetch Data element can retrieve more than one row depending on the number of rows in the source, and your filter conditions. When one or more rows are fetched, a reference to a field by another element is always a reference to the field's value in the current row, in the set of rows fetched. The Move Current Row and Prompt for Current Row elements can be used to change which row is current. Immediately after a Fetch Data or Dial for Data element is executed, the current row is the first row.

- When only one row is fetched, the first row is the single row that was fetched.
- When multiple rows are fetched, which row is first depends on whether a sort order was defined by the Administrator.

If no sort order was defined, then the order of the rows is determined by the data source, and you should not make any assumptions about the order. If a sort order is defined, the *first* row is the first row in the sort order specified. However, if the sort field contains duplicate values, the order of rows with the same sort value is determined by the data source, and you should not make any assumptions about the order.

For example, assume that you have a data source with a Date field, and that each row in the data source has a unique date value. If the data source defines a sort order based on this Date field and descending order, then any Fetch Data element that fetches from the data source will retrieve rows in descending date order. If a Fetch Data element fetches multiple rows, then the *first* row will be the row in the fetched set of records with the newest value in the Date field. Thus another element referencing this Fetch Data element would always reference the newest row. Without the sort, you cannot predict which row in the retrieved set would be *first*.

Selecting a row of data

When a Fetch Data element fetches one or many rows, the first row becomes the *current* row. The order of the rows is determined by the sorting modes selected, as described above.

Other elements can refer to fields fetched by the Fetch Data element. These elements reference the value in the current row. A different current row can be selected, using the [Move Current Row element on page 307](#) or [Prompt for Current Row element on page 308](#) elements.

For information about specifying data sources, filter fields, and values, refer to [Using the Data Source Selector on page 297](#).

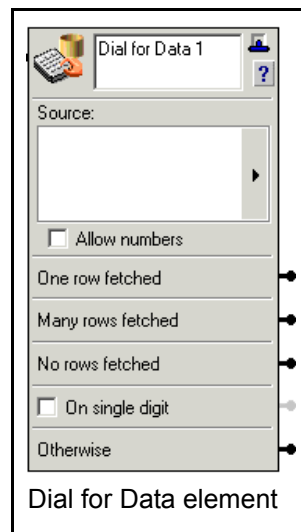
Dial for Data element

The Dial for Data element allows you to specify a data source using the Data Source Selector. It is similar to the Fetch Data element, except the caller uses the telephone keypad to define the data that is fetched. It is also similar to the Dial By Name element, which is available only to the administrator, except the search is done on a defined database.

The Dial for Data Source Selector allows you to specify the following:

- data source
- optional filter condition, for example a value
- Select On field

When a caller encounters the Dial for Data element, the element selects the records from the source that match the filter condition (it fetches all the records if no filter condition is specified). Then the element prompts the caller to enter the value for the Select On field. A search using the Dial for Data element ignores blanks and special characters. The default search criteria include only alphabetic characters. If you specify Allow numbers, numeric characters are also included in the search.





The letters *Q* and *Z* appear on different keys on different phones. The Dial for Data element works whether the letter *Q* corresponds to the 0 or the 7 key, and whether the letter *Z* corresponds to the 0 or the 9 key.

Dialing for Data results

- If the Dial for Data element fetches one row, the call exits from the **One row fetched** connector pin.
- If the Dial for Data element fetches more than one row, then the call exits from the **Many rows fetched** connector pin
- If no data rows are fetched, then the call exits the **No rows fetched** connector pin.
- If you enable the **On Single Digit** check box, the call exits that connector pin when the caller enters only one digit, or the call times out after one digit is entered.
- If any other condition exists, such as the inability to access a Data Source, such as an external database or internal User Data Table, then the call exits the **Otherwise** connector pin.

Sorting data rows

A Dial for Data element can fetch more than one row depending on the number of rows in the source, the filter conditions, and the field the user will select on. When one or more rows are fetched, a reference to a field by another element is always a reference to the field's value in the first row, in the set of rows fetched.

- When only one row is fetched, the first row is the single row that was fetched.
- When multiple rows are fetched, which row is first depends on whether a sort order was defined by the Administrator.

If no sort order was defined, then the order of the rows is determined by the data source, and you should not make any assumptions about the order. If a sort order is defined, the *first* row is the first row in the sort order specified. However, if the sort field contains duplicate values, the order of rows with the same sort value is determined by the data source, and you should not make any assumptions about the order.

For example, assume that you have a data source with a Date field, and that each row in the data source has a unique date value. If the data source defines a sort order based on this Date field and descending order, then any Dial for Data element that fetches from the data source will retrieve rows in descending date order. If a Dial for Data element fetches multiple rows, then the first row will be the row in the fetched set of records with the newest value in the Date field. Thus another element referencing this Dial for Data element would always reference the newest row. Without the sort, you cannot predict which row in the retrieved set would be first.

Selecting a row of data

When a Dial for Data element fetches one or many rows, the first row becomes the *current* row. The order of the rows is determined by the sorting modes selected, as described above.

Other elements can refer to fields fetched by the Dial for Data element. These elements reference the value in the current row. A different current row can be selected, using the [Move Current Row element on page 307](#) or [Prompt for Current Row element on page 308](#). For information about specifying data sources,

filter fields, and values, refer to [Using the Data Source Selector on page 297](#).



You can put a Play Announcement element before a Dial for Data element to give the caller instructions about how to use the feature.

Add Data element



The Add Data element allows callers to add to the data in a specified data source. The Record Announcement element can be used to record temporary recordings, which can be used with the Add Data element as a form filling application, in order to request quotes, have callers answer general customer service questions, and so on. For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

- If the data is added successfully, the caller exits the regular connector pin.
- If the data is not added successfully, for any reason, the caller exits the Otherwise connector pin.

To specify a source and fields for the Add Data element

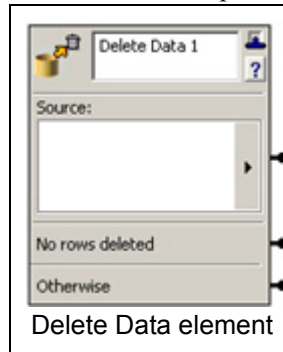
1. Select a data source from the **Source** list.
2. Select the **Add** button to add a Field Assignment to the list.
3. Select the **Field** column to obtain a drop-down list of fields in the data source.
4. Select a field from the list.
5. Select the **New Value** column to specify a value for the field.
6. Type a value, or select the value selector arrow in the blank new value box.
7. Select anywhere outside the New Value box.
8. Repeat steps 2 to 7 to add more fields to the Field Assignments list.

To delete a field assignment for the Add Data element

1. Right-click the **Field** column of the field assignment that you want to delete.
2. Press **Delete**.

Delete Data element

The Delete Data element allows callers to delete data in a specified data source.

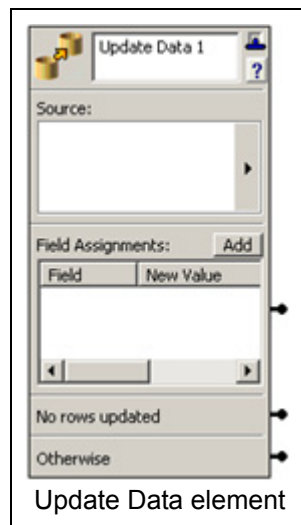


If some rows are deleted successfully, the caller exits the Source connector pin. If no rows are deleted, the caller exits the No rows deleted connector pin. If any other condition exists, the caller exits the Otherwise connector pin. For information about specifying data sources, filter fields, and values, refer to [Using the Data Source Selector on page 297](#).



*For security reasons, when including a Delete Data element in a service, you might want to put a Verify Password element before it. Make sure that you specify exactly the data that you want to make available for deletion. For example, if you do not include a filter field and value, **all rows** are deleted.*

Update Data element



The Update Data element allows callers to update the data in a specified data source. The Record Announcement element can be used to record temporary recordings, which can be used with the Update Data element as a form filling application, in order to request quotes, have callers answer general customer service questions, and so on. For more information on temporary recordings, refer to [Allowing callers to make temporary recordings within a service on page 227](#).

- If some rows are updated successfully, the caller exits the regular connector pin.

- If some rows are not updated, the caller exits the No rows updated connector pin.
- If any other condition exists, the caller exits the Otherwise connector pin.

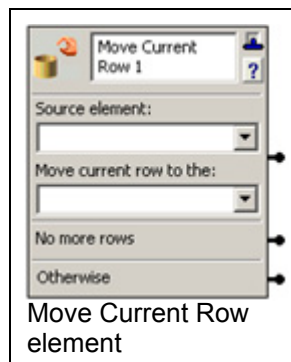
To specify field assignments for the Update Data element

1. Use the data source selector arrow to specify the data source that you want.
2. Select the **Add** button on the Update Data element to add a Field Assignment to the list.
3. Select the **Field** column to obtain a drop-down list.
4. Select a field from the list.
5. Select the **New Value** column to specify a value for the field.
6. Type a new value, or select the value selector arrow in the blank new value box.
7. Select anywhere outside the New Value box.
8. Repeat steps 2 to 7 to add more fields to the Field Assignments list.



*Make sure that you specify the filter condition correctly. For example, no filter condition means that **all rows** are updated using the specified assignments.*

Move Current Row element



When using the [Fetch Data element on page 301](#) or [Dial for Data element on page 303](#), a set of rows are fetched from the database. The **Move Current Row** element moves the current row in a set of rows fetched. The current row can be moved to be the first row in a set of rows, the last row, the next row, or the previous row.

The Source element for a Move Current Row element can be the name of any other Fetch Data or Dial for Data element in the service. The current row can be moved to the first row, last row, next row, or previous row in the set of rows associated with the Fetch Data or Dial for Data element.

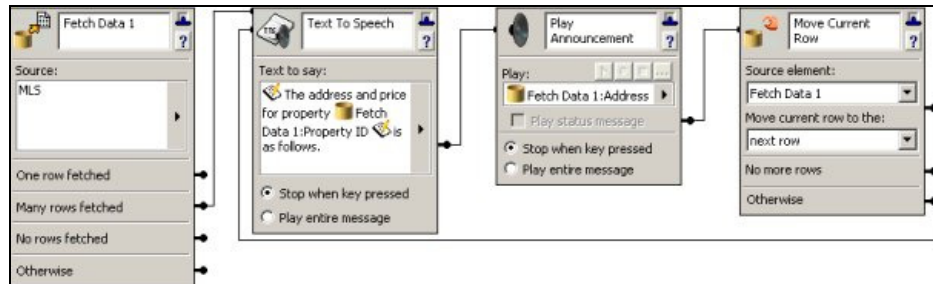
- If the current row moves to another row, then the element exits through the **Source element** connector pin.
- If moving the current row moves it beyond the last row, before the first row, if there are no rows, or if the Source element has not been executed yet, then the element exits through the **No more rows** connector pin.

- If an error occurs, then the element exits through the **Otherwise** connector pin.

Applying the move current row element

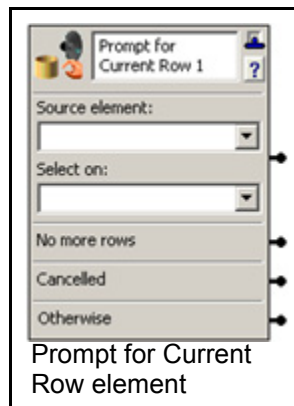
We'll use a simplified version of a real estate listing service to illustrate how the Move current row element works. For example:

- The first element is a Fetch Data element that fetches some real estate listings.
- The second element is the Text to Speech (TTS) element which speaks the property identifier from the current row, and introduces the audio that follows.
- The third element is a Play Announcement element that plays the audio for the current row, and contains the address and price for the current listing.
- The fourth element is the Move current row element, which moves the current row to the next row.
 - If there is another row, the service loops back to the TTS element, which now uses the value from this row.
 - If no more rows are available, the Move current row element exits through the No more rows connector pin.



Prompt for Current Row element

The Prompt for Current Row element prompts callers to select a specific row from multiple rows fetched by a [Fetch Data element on page 301](#) or a [Dial for Data element on page 303](#). Callers press keys on the keypad in response to prompts such as "For..., press 1, for..., press 2..." Callers select a specific item which becomes the current row.



The Source element for a Prompt for Current Row element allows you to select a Fetch Data or Dial for Data element in the service. The Select on list contains the names of fields in the Source element, with the exception of Audio or Fax fields.

- If the Source element has exactly one row, the element exits immediately through the **Source element** connector pin.
- If the record set indicated by the Source does not exist (the element has not been executed yet), or if it has no rows, then the element exits immediately through the **No more rows** connector pin.
- If the caller presses an invalid key, or fails to press a key within a few seconds, then the last prompt is repeated. If this occurs three times, or if an internal error occurs, then the element exits through the **Otherwise** connector pin.

Prompting callers when more than one row is returned

If the Source element has more than one row, using Text to Speech (TTS), the element prompts the callers with "Press 1 for <value 1>, Press 2 for <value 2>," and so on, where the values of the Select on field are from the set of rows associated with the source element. The caller is prompted for up to six choices.

- The element follows with the prompt "Press 8 to repeat these choices."
- If there are more than six rows, then it prompts with "Press 9 for more choices."
- Finally, the element prompts with "Press 0 to start over."

Caller input and results

If the caller enters the number associated with a specific row value, the associated row becomes the current row, and the element exits through the **Source element** connector pin.

- If the caller presses 8, the prompts are repeated.
- If the caller presses 0, the same prompt for the first group of rows is repeated.
- If there are more than six rows and the caller presses 9, the element prompts with "Press 1 for <value>..." as before, but the values are now taken from the next group of rows. This prompt is followed by "Press 7 for previous choices," and the other prompts as noted above. Pressing 7 causes the prompt for the previous group of rows to be played.

The process continues until the caller selects a specific row. The caller uses the 7 and 9 keys to move backwards and forwards through the set of rows.

Caller prompts

Key Pressed	Results
0	The prompt for the first row set is played.
1-6	This selects the associated row value, and becomes the current row. The element exits through the Source element connector pin.
7	If there are previous row sets, the prompt is played for the previous six rows.
8	The previous prompt is replayed.
9	If there are more rows, the prompt is played for the next six rows or less.

18 Importing and Exporting

This chapter explains how to use the Import/Export Wizard and covers the following topics:

- [Overview on page 311](#)
- [Starting the Import/Export Wizard on page 311](#)
- [Importing and Exporting Items on page 311](#)

Overview

The Import/Export Wizard allows you to share announcements, services, faxes, and fax cover pages by importing items from another location, or exporting them from your computer to another location so that other people can import them and use them.

Starting the Import/Export Wizard

To start the Import/Export wizard

1. Select **File > Import-Export Wizard** to open the wizard welcome menu.
2. Select **Next**.

Importing and Exporting Items

The Import/Export Wizard guides you through the process of importing or exporting an item.

To import an item

1. In the Task Selection box, select **Import data from the File System**, and then select **Next**.
2. In the Import Directory box, make sure that the destination folders are correct.



The default destination is: C:\Documents and Settings\username\My Documents\Export

You can use the Browse button to change the default destination folder.

3. Select **Next**.
4. Select all the items that you want to import, and then select **Next**.
Make sure that the box next to the item has a check mark in it before you proceed.
5. If you want to delete an item in the change destination box, select the item and then select the **Delete** button.

6. If you want to change the destination of an item, select the item and then select the **Change destination** button.
7. Select **Next** to transfer the items.
8. Select **Next** when the wizard is done.
9. Select **Finish**.

To export an item

1. In the Select task to perform box, select **Export to the File System**, and then select **Next**.
2. Select the directory where the items that you want to export are located, and then select **Next**.
3. Select all the items that you want to export, and then select **Next**. Make sure that the box next to the item has a check mark in it before you proceed.
4. Select **Finish** when the wizard is finished exporting the items.

19 Maintaining the UC Server

Basic maintenance procedures such as backups are recommended in the event that disaster recovery is required. For more information, refer to the following section:



For information on restoring files and settings, refer to [Restoring NetVanta UC Server Files and Settings](#) on page 328.

Backing up the UC Server

Backing up the UC Server and its data are essential for disaster recovery. The UC Server backup should conform to the site policy for backups.



It may be useful to backup the UC Server in a working state using third-party tools. Combined with current data backups, a good image can result in a quick system restore in the event of a total system failure.



*For more information on backing up and restoring the UC Server, refer to the technical note **Backup and Restore Procedures**, available from the ADTRAN website at <http://kb.adtran.com>. This and other documents are also located on the UC Server installation media.*

20 Troubleshooting

Overview

Occasionally problems may occur where the UC server does not function as expected. Some potential problems and their solutions are noted below. Note that this is not an exhaustive list.

You can use UC server logs to review calls, events, system errors or warnings.

You can configure the UC server to automatically generate diagnostic information in the event of errors or problems. You can also configure the diagnostic system options such that ADTRAN, your reseller, and/or yourself is notified if the UC server is not functioning properly.

If you cannot find the information you are looking for in this manual or in the online help, then you can go to the ADTRAN website for further information such as training materials, a quick reference card, SIP phone user guides, and technical notes. If you are still unable to resolve the issue then you can contact your reseller, or ADTRAN Technical Support if a support contract is in place. For more information, refer to the following sections:

- [Troubleshooting Playback and Recording on page 315](#)
- [Troubleshooting Voicemail on page 316](#)
- [Troubleshooting Message Waiting Indicators on page 317](#)
- [Troubleshooting the NetVanta UC Server on page 318](#)
- [Troubleshooting Performance Issues with Exchange Server on page 319](#)
- [Using the NetVanta UC Server Logs on page 320](#)
- [Enabling Diagnostic Information on page 327](#)
- [Accessing Documentation and Training on page 327](#)
- [Accessing Technical Support on page 327](#)
- [Restoring NetVanta UC Server Files and Settings on page 328](#)



For information on configuring diagnostic system options, refer to [Configuring diagnostics system options on page 37](#).

Troubleshooting Playback and Recording

Symptoms

The beginning of message playback, announcement playback and recording, and greeting playback and recording are being cut off when being played or recorded using the handset.

Cause

You are connecting the UC server through a gateway to an existing PBX, and the attached PBX does not have true answer supervision.

Solution

To configure the PBX to avoid cutting off the beginning of playback and recording using the handset

1. In the Communication System content pane, right-click the Generic PBX (via Gateway) that you want to change, and select Open.
2. In the Options dialog, enable the Prompt for handset playback and recording option.
3. Select Change.

Troubleshooting Voicemail

Symptoms

- Users complain that when they login to the telephone mailbox, there is a delay before their mailbox and message information is available.
- You are using an IMAP server to access voicemail messages.

Causes

IMAP servers vary in the methods they support for retrieving a user's voice and fax messages, and the speed at which they retrieve such messages.

- *No query optimization*, the slowest method, was selected, and a user has a large number of messages in their inbox.
- Another search method was selected that was not supported, then the UC server automatically defaulted to the *No query optimization* method when the selected method failed, and a user has a large number of messages in their inbox.
- The email server is slow in responding, possibly due to a heavy load, a large number of messages in a mailbox, or slow or unreliable network connections.

Solutions

To optimize IMAP server performance and to minimize delay between when a user logs in to their telephone user interface (TUI) to check their mailbox, and when the UC server can play mailbox and message information, such as the number of new messages, ensure that you select the fastest search method available that your IMAP server supports.

- If you are using Gmail, GroupWise, or Lotus Notes, then ensure that you have selected the recommended method for message retrieval. Refer to [Configuring IMAP Server on page 141](#) for more information.

- If you are not using Gmail, GroupWise, or Lotus Notes with IMAP, then ensure that you selected the fastest search method available that was successful, as outlined in [Determining a Method of Filtering Messages for IMAP Servers on page 143](#).



Selecting a method that is unsupported for your configuration will decrease server performance and increase message retrieval delay as the selected method is attempted and fails, before the UC server defaults to the No query optimization method.

- If you are using the *No query optimization* method, then encourage users to avoid storing large numbers of messages in their inbox as this will delay the message retrieval process.
- Reduce the number of messages in the user's inbox.
- Ensure that the email server is operating correctly and that it is not overloaded, or that it was not overloaded at the time of the complaint.
- Ensure that the network connection between the UC server and the email server is reliable and has sufficient bandwidth in both directions.

Troubleshooting Message Waiting Indicators

Sometimes message waiting indicators (MWI) on phones are incorrect following a power failure or changes to the PBX.



For information on configuring message waiting indicators system-wide, refer to [Configuring message waiting indicator options on page 21](#). For information on configuring a user's message waiting indicator, refer to [Configuring a user's message waiting indicator on page 99](#). For information on synchronizing message waiting indicators when messages are accessed through the web or through a mobile device, refer to [Synchronizing message waiting indicators for messages accessed through the web or mobile devices on page 139](#)

Symptoms

The MWI on phones are incorrect.

Possible Causes

- A power failure has occurred.
- Changes were made to the PBX.

Solution

You can resynchronize the MWI on phones.

To resynchronize message waiting indicators on phones

Select **Tools > Resynchronize All message waiting indicators**.

Troubleshooting the NetVanta UC Server

ADTRAN has observed that third-party firewall and antivirus products can cause problems with the proper operation of the UC server.



The UC server is tested with Microsoft security products, such as Windows Firewall and Windows Defender.

Symptoms

- UC server services keep shutting down with Trend Micro software installed.
- The UC client crashes, doesn't load forms, or produces unknown software errors.
- ucCompanion doesn't load extensions with Symantec software installed.

Possible Causes

Sometimes problems can arise from third-party firewall or antivirus software installed on computers hosting the server or client software.

Solutions

Proceed with the following potential solutions in the order in which they appear. Only when solution A and B have been attempted and failed, should solution C be tried.



If a problem occurs, ADTRAN Technical Support advises that the third-party product be removed from the system prior to contacting Technical Support for assistance. For more information, refer to [Accessing Technical Support on page 327](#).

Solution A: Verify that the NetVanta UC Server files are excluded from antivirus software scanning

Verify that the following folders are excluded from antivirus software scanning (as noted in the NetVanta UC Server *Deployment Guide*, and in the NetVanta UC Server *Installation Guide*).

- C:\Program Files\ADTRAN

Or any folder in which the UC server application is loaded.

- C:\ADTRANLogs

Or the folder to which logging application is configured.

- C:\Program Files Microsoft SQL Server

Or the default location of the database folder.

Solution B: Uninstall the third-party antivirus or firewall software

1. Uninstall the third-party antivirus or firewall software.
2. Shutdown the UC server and restart.



Antivirus and firewall behavior is unpredictable even after shutting down or uninstalling.

Solution C: Uninstall all software and reinstall

Wipe clean the computer hosting the server/client software, including the OS, and reinstall.

Troubleshooting Performance Issues with Exchange Server

Symptoms

- Reduced performance on Microsoft Exchange Server.

Causes

Mailbox monitoring is enabled on the Exchange Server, and the default setting for mailbox monitoring for users is set to on.

Alternatively, mailbox monitoring is enabled on the Exchange Server, and the default setting for mailbox monitoring for users is set to off, but a significant number of individual users have mailbox monitoring enabled.

Mailbox monitoring enables synchronization of message waiting indicators when users access messages through Microsoft Outlook Web Access or through a mobile device. However, mailbox monitoring can negatively impact system performance.



For more information on mailbox monitoring, refer to [Synchronizing message waiting indicators for messages accessed through the web or mobile devices on page 139](#).

Mailbox monitoring increases the traffic between the Exchange Server and the UC server. Depending on the number of users configured for mailbox monitoring, the bandwidth between the Exchange Server and the UC server, and the CPU power, performance may be affected. On systems with more than 100 users, system design and engineering must consider the increased load that keeping open large numbers of mailboxes will have on the Exchange Server. Enabling monitoring of more than 500 mailboxes requires careful consideration and probably should not be done without a slow roll-out procedure to validate the performance of both Exchange Server and the UC server in the specific customer environment. If your Exchange Server is at or near capacity, then you may want to disable mailbox monitoring.

Solutions

The following are two possible solutions to improving performance on Exchange Server by disabling or decreasing mailbox monitoring. Select the option you prefer, and then follow the relevant instructions below to implement it.

- Disable mailbox monitoring system-wide so that mailbox monitoring is disabled for all users, regardless of individual user settings.
- Enable mailbox monitoring, and set the default setting for mailbox monitoring for users to off, then enable mailbox monitoring only for those users who require it.

To disable mailbox monitoring system-wide

1. In the Administration navigation pane select Servers.
2. Double-click on Exchange.
3. Under Mailbox monitoring uncheck Enable mailbox monitoring.
Mailbox monitoring is now disabled for all users, regardless of each users' mailbox monitoring settings.

To enable mailbox monitoring and set the default for users to off

1. In the Administration navigation pane select Servers.
2. Double-click on Exchange.
3. Under Mailbox monitoring ensure that Enable mailbox monitoring is checked, and set the Default setting for users to Off. The default setting for users is to have mailbox monitoring turned off, however individual users can have mailbox monitoring enabled.

To enable or disable mailbox monitoring for an individual user

1. In the Administration navigation pane, select Users.
2. Right-click on the user whose mailbox monitoring settings you want to change, and select Open.
3. Select the Messaging tab.
4. Under mailbox monitoring select Enabled or Disabled to turn mailbox monitoring on or off for the user.



For more information, refer to [Configuring a user's message storage on page 96](#).

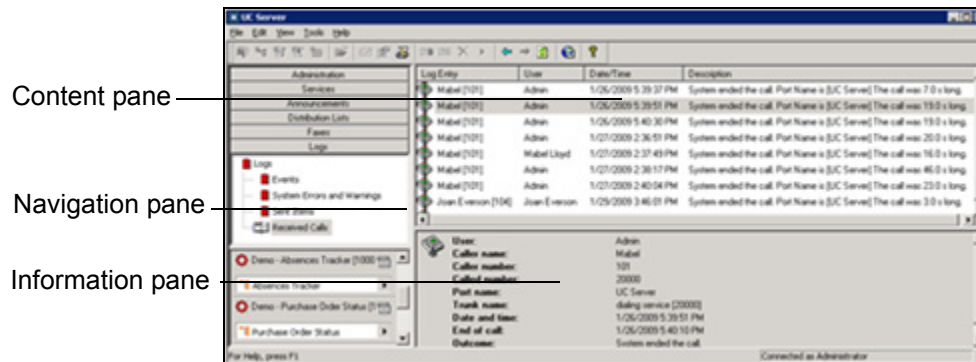
Using the NetVanta UC Server Logs

This section explains how to manage alert notifications of errors and warnings, logging statistics, and logs and the information that they record. This section includes the following topics:

- Errors and Warnings Alert Notifications

- Mailbox message logging statistics
- Managing logs

Logs record events that occur in the system. The administrator can view logs for events on trunks and user extensions. Four categories of events are recorded in logs: events, system errors and warnings, sent items, and received calls. You can view logs and store them for future reference, track activity that takes place, and troubleshoot occasional problems. Using the Import/Export wizard, you can export call logs as a tab separated or XML-based file. The call log files can be used by other reporting applications to provide historical reports on voicemail activity. For more information about the Import/Export wizard, refer to [Importing and Exporting on page 311](#).



The **Events** log tracks information related to user attempts to perform mailbox management.

The **System Errors and Warnings** log provides information that you can use to troubleshoot problems that occur with your call answering.



To troubleshoot specific system errors and warnings, refer to [Troubleshooting Warning Logs on page 339](#).

The **Sent Items** log tracks information on outgoing faxes and events related to the use of the Notify Pager and Deliver Messages elements. Details for each delivery or dispatch attempt are recorded, including the date and time of the attempt; the phone number dialed; the port that handled the attempt; and the final outcome of the delivery or notification attempt.

The **Received Calls** log tracks incoming calls, regardless of whether the UC server answered them or not. Information logged includes the caller name and phone number, if available; the date, time, and length of the call; whether the call was answered, and by which port; and whether a message was left or not.

Errors and Warnings Alert Notifications

If an error or warning occurs, for example, the UC server was unable to deliver a message, then the UC server generates an error or warning log. When an error or warning log is produced, and there were no previous unacknowledged errors or warnings, then UC server also produces an entry in the Windows event log to indicate what has happened. Administrators will see an orange bar displayed at the bottom of the UC client window as an alert that some action may be required. To learn more about the alert, refer to [Appendix C - Troubleshooting Logs on page 337](#). Conversely, when errors and warnings are acknowledged, and

there are no more unacknowledged errors or warnings, then a different entry is created in the Windows event log, and the orange bar in the UC client disappears.

To view system errors and warning logs

1. On the left side of the UC client, select the **Logs** navigation bar to open the Logs menu.
2. Select **System Errors and Warnings**. On the right side in the Logs content pane, the errors and warning logs appear.

To acknowledge the alert notification

On the right side in the Logs content pane, right-click on an error or warning log to accept or acknowledge the error or warning. The orange bar will no longer appear, and the error or warning message is no longer visible at the bottom of the UC client window.

Managing NetVanta UC Server Alerts from Windows SBS 2008

Microsoft Windows Small Business Server (SBS) 2008 records unacknowledged errors or warnings in the UC server. This simplifies management such that Administrators do not need to have the UC client open in order to be aware of any error or warning messages.

When the UC server is installed on Windows SBS 2008, a custom alert is automatically configured for the SBS 2008 Console. If the UC server is not installed on the SBS 2008 system, but is installed on another system on the network, then consult the Microsoft documentation for how to create a custom alert.

To create a custom alert definition when the NetVanta UC Server is not installed on the SBS 2008 system

1. Refer to www.msdn.microsoft.com/en-us/library/cc721719.aspx for details on how to create a custom alert.
2. Specify the following:

Path	Application
Provider	UC server
SetEventID	256
ClearEventID	512



The Windows SBS Console display is refreshed every 30 minutes. This means that the console display will not change immediately when an error or warning occurs, or when all errors or warnings are accepted or acknowledged.

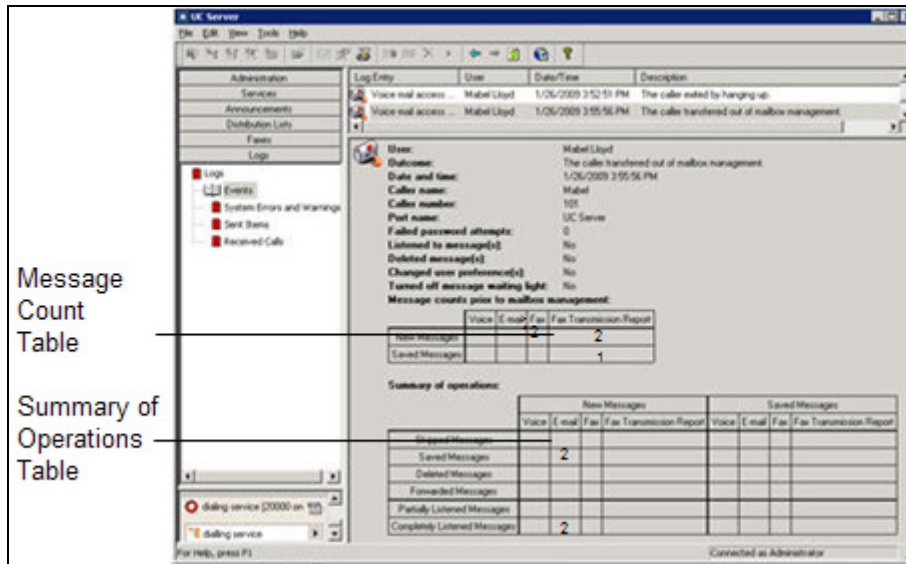
To access the NetVanta UC Server alerts from Windows SBS 2008

1. From the **Start** menu access the **Event Viewer**.
2. Open the **Windows SBS Console**, and select the **Network** tab. Under Computers, any UC server unacknowledged errors or warnings will appear.

Mailbox message logging statistics

In the information pane of the Logs menu, two tables provide a visual representation of the message summary and the summary of operations done by all users on the system that are managing their mailboxes:

- Message Count table
- Summary of Operations table











NOTE *If message summary is skipped during mailbox management, the Message Count table is not displayed in the information pane.*

The log entry types are represented by different icons. The following table explains some of the icons that

might appear in your list of log entries.

Table 20–1: Log entry icon definitions

Icon	Definition
	Error—represents an error, for example in voice message delivery
	Warning—represents a warning
	Received calls—represents calls that come in on a specified port
	Voicemail access—represents a voice message event, for example, a voice message sent from a UC server user to another user
	Pager access—represents events in which UC server users send or receive a page
	Message delivered
	Voicemail access—represents user voicemail access, for example, users accessing their own mailbox to receive voice messages
	Log entry

Managing logs

To get the Logs menu, select the Logs navigation bar. You can retrieve and view, remove, and archive logs. You can rename and delete log archives, and export a call log using the Import-Export wizard.



You can configure the UC server to automatically archive logs. For more information, refer to [Setting Administrative, Maintenance, and Diagnostic Options on page 36](#).

To retrieve and view a log

1. Double-click the log folder that you want to view. The contents of the log are displayed in the content pane. When a log folder is open, its icon appears as an open book.



Closed log folder



Open log folder

2. Use the vertical scroll arrows to view the list of log entries.
3. Select a log entry to view information about it in the details pane.

To remove old log entries

1. Right-click anywhere in the content pane.
2. Select **Clear log**.
3. Select one of the following:
 - **Everything**
 - **Entries created before today**
 - **Entries created before the beginning of the month**
 - **Entries created before a specific date**



If there are unacknowledged errors or warnings, you cannot archive or clear log items. You must accept the notifications first.

4. If you enabled **Entries created before a specific date**, type the date in the text box.
5. Select **OK**.

To archive a log

1. Right-click anywhere in the content pane.
2. Select **Archive log**.
3. Select one of the following:
 - **Everything**
 - **Entries created before today**
 - **Entries created before the beginning of the month**
 - **Entries created before a specific date**



If there are unacknowledged errors or warnings, you cannot archive or clear log items. You must accept the notifications first.

4. If you enabled **Entries created before a specific date**, type the date in the text box. The new archive is put in the navigation pane as a log with a default title based on the current date.



20020701_0

Log archive

5. Select **OK**.

To rename a log archive

1. Select the archive that you want to rename.
2. Select **Edit > Rename**.
3. Type a new name for the archive.

To delete a log archive

1. Select the archive that you want to delete.
2. Select **Edit > Delete**.
3. Select **Yes** to confirm the deletion.

To export a call log using the Import-Export wizard

1. Select **File > Import-Export Wizard**, and then select **Next**.
2. Select **Export Activity Report**, and then select **Next**.
3. Select the types of logs that you want to export, and then select **Next**.
4. Enable either of the following, and then select **Next**:
 - **All logs**—exports all call logs
 - **Date range**—exports call logs from within a range of dates that you specify
5. Select the file type for the call log that you want to export.
6. Do either of the following, and then select **Next**:
 - Type a location for the file in the **Save exported file as** box.
 - Select **Browse** to find a location for the file.
7. If you want to save the configuration, type a name in the box.
8. Select **Next** to export the log.
9. Select **Close**.



View the call log files using the appropriate application, for example, Microsoft Excel.

Enabling Diagnostic Information

During the UC server installation, one of the prerequisite components installed was the UC server *Logging Tools*. In the event of errors or problems, this tool enables the UC server to automatically generate diagnostic information. You can configure the UC server so that this diagnostic information is automatically sent to the ADTRAN Technical Support team.



For information on configuring diagnostic information, refer to [Configuring diagnostics system options on page 37](#). ADTRAN does not take action in response to the emails received, but gathers the data for statistical purposes only to aid in improving the UC server. ADTRAN will not take action unless a support incident is raised, following standard support procedures. For information on contacting ADTRAN Technical Support, refer to the ADTRAN website at www.adtran.com.

Accessing Documentation and Training

If you want more information than what is available in this manual or in the online help, then go to the ADTRAN website at <http://kb.adtran.com>. Here you will find training materials, a quick reference card, along with SIP phone user guides, and technical notes.

Accessing Technical Support

Unless there is a support contract in place, please see your reseller for support information. For more information on ADTRAN Technical Support, refer to the ADTRAN website at www.adtran.com.

If you are contacting ADTRAN Technical Support, then please ensure that you have the following information readily available.

- Product edition (Standard, SIP, or CEBP)
- Product version
- Service code

The Service code is an identification number to be used when contacting ADTRAN Technical Support. If you are contacting ADTRAN Technical Support then you will need to enter the service code number over the phone, so have the number ready when you call. The service contract type and expiry date are available for your information.

To access the NetVanta UC Server product and service information

From the UC server select **Help > About UC Client**. Alternatively, select the ? button on the toolbar.

Restoring NetVanta UC Server Files and Settings

To restore data from a Microsoft backup file requires that the data has been previously backed up using appropriate backup procedures.



*For disaster recovery purposes, ensure that you backup the UC server regularly. For information on backing up the UC server, refer to [Backing up the UC Server on page 313](#). For more information on backup and restore procedures, refer to the technical note **Backup and Restore Procedures**, available on the ADTRAN website at <http://kb.adtran.com>. This and other documents are also located on the UC server installation media.*



Restoring the UC server from a backup file only restores the system to the state it was when the backup file was created. Any changes made to the system since the backup file was created will need to be repeated.

Appendix A - Glossary

Account Codes

An account code is a specified set of digits that the caller must enter after completing the dialing pattern. Account codes represent an accounting category, for example, expenditures, revenues, or salaries. They are also used to ensure security because a caller who does not have an account code cannot access specified information. Use account codes to allow long distance dialing, in which the telephone company applies the long distance charges to your account code.

Active Message Delivery

Active Message Delivery is a feature of the UC server. The UC server calls you to deliver your messages. It informs you of a new message and asks you to enter your password to review the message.

Active Service

Your active service is the service that is currently enabled and answering your calls.

Announcements

Announcements are outgoing recorded messages that play to callers. Announcements apply to trunk lines and are managed by the network administrator.

ALG

An application layer gateway (ALG, also known as an application-level gateway) is a security element that enhances a firewall or NAT. It allows legitimate data to pass through the firewall, that would have otherwise been restricted.

Assisted Transfer

An option that allows the call recipient to accept or deny a call based on the caller's recorded name.

Caller

A caller is a person who initiates an incoming call.

Call Park/Pickup

A type of hold that allows a user to "park" a caller on hold so that another user on the system can pick up the call on another telephone.

Call Transfer

Have your calls transferred to an operator, to another number, or to a different extension using the Call Transfer element in your service. Note that your workplace must have Call Transfer from your telephone service provider or PBX to be able to use the Call Transfer element.

CO

A Central Office is a telecommunications office that is centralized at a location specifically to handle the telephone service for that location. Phone lines are connected to the CO on a local loop. ISDN and DSL signals go through the CO.

Contacts

Contacts are individuals from whom you receive calls, email messages or faxes. They are entered in the Contacts database in Microsoft Outlook.

Cover Page Editor

The Cover Page Editor is the tool you use to create and design fax cover pages.

DHCP

The Dynamic Host Configuration Protocol is a network IP protocol that assigns dynamic IP addresses to devices on a network. When the DHCP server is queried for a dynamically assigned IP address, it assigns one from its range of valid addresses.

DID

Direct Inward Dialing is a service that allows numerous individual phone numbers for each user on a PBX system to run off a small block of dedicated telephone numbers. DID allows multiple lines to connect to the PBX without requiring each to have a physical line connecting to the PBX. A DID system can be used for voice and fax transmissions.

DLL

A Dynamic Link Library consists of executable functions or data that is available for a Windows application to use. A DLL can provide one or more functions and a program accesses the functions using either a static or dynamic link to the DLL.

DNS

The Domain Name Service is an Internet service that translates domain names into IP addresses. For example, the domain name `www.adtran.com` might translate to `198.105.222.1`

DTMF

Dual tone multi-frequency tones are also known as touchtone dialing.

E.164

E.164 refers to the international public telecommunication numbering plan, and defines the format of telephone numbers. An E.164 number can be up to 15 digits, and is usually written with a + prefix. To dial an E.164 number from a normal phone line requires the appropriate international call prefix.

Fax-on-demand

Fax-on-demand allows callers to dial in and retrieve a fax document completely unattended. The caller

follows instructions to select the desired document and the UC server automatically transmits it. The administrator sets up fax-on-demand services on trunk lines.

FQDN

A fully qualified domain name is a unique identifier that consists of a host and a domain name, such as www.adtran.com. A FQDN is distinguished from a regular domain name by a period added to the end, to indicate that the name has ended.

FTP

The File Transfer Protocol allows users to exchange files over the Internet. It is also used to log into the Local Area Network, list directories, and copy files between systems.

FXO

The Foreign Exchange Office is the interface on a VoIP device for connecting to an analog loop start or ground start, terminated from the central office. It can also be used to connect to an analog PBX extension. An FXO interface receives ringing from the central office. It behaves like a regular analog POTS telephone.

FXS

The Foreign Exchange Station is the interface on a VoIP device for connecting directly to phones, faxes, and CO ports on a PBX or key telephone systems.

Gateway

A node that connects multiple networks or subnetworks. The gateway is associated with both a router, which uses message information to determine where packets are sent, and a switch, which provides the path for incoming and outgoing packets.

Greetings

Users can record three types of standard greetings: an identifier, an internal greeting, and an external greeting. The identifier states the user's name. The internal greeting is played to internal callers, while external callers making incoming calls hear the external greeting.

Identities

Any dialable string of numbers, such as an extension or email address, that reaches a specific user is an identity. A user profile can have several identities.

Integrated Messaging

Integrated messaging can be enabled when a user's messages are stored locally. If Integrated Messaging is enabled for a user, then a Mailboxes navigation bar is added to the user's UC client. The user can then manage their voice and fax messages visually through the Inbox and Sent Items folders in the UC client.

ISDN

The Integrated Services Digital Network is an international communications standard for sending voice,

video, and data over telephone lines.

ITSP

Internet Telephony Service Providers offer Voice over IP (VoIP) services using broadband connections. VoIP is a combination of hardware and software that enables users to make and receive telephone calls over the Internet. The voice data is transmitted in packets using Internet Protocols (IP) instead of using the traditional public switched telephone network (PSTN).

LAN

A Local Area Network connects computers and other devices within one location.

Logs

Logs are records of activity within the UC server. The UC server logs all events and provides information on Errors and Warnings, Received Calls, Sent Items, and Events.

MAC Address

A Media Access Control address is a unique identifier for your computer or SIP telephone. Every device on the network must have a MAC address. When connected to the Internet, a table relates the IP address to a MAC address on the LAN.

NAT

Network address translation (NAT) is an Internet standard for reusing IP addresses to avoid a shortage of, and having to reserve, IP addresses. NAT enables a local area network (LAN) to use one set of IP addresses for internal traffic, and a second set of addresses for external traffic. NAT rewrites the source and/or destination addresses of IP packets as they pass from the LAN through a router or firewall to the Internet, enabling multiple hosts on a private network to access the Internet using a single public IP address.

One-time Message

A One-time Message is a message that you record for an expected caller. The message is caller-specific such that no others will receive it. When the message has been played, it is automatically erased.

Pager notification

Pager Notification is a feature of the UC server that calls your supported paging system to inform you of new messages in your mailbox.

PBX

The Private Branch Exchange is a system of telephone lines that serves an organization and uses telephone services, rather than supplies them. Users of the PBX share a certain number of outside lines for making external calls.

Phantom mailbox

Phantom mailboxes are not associated with a physical identity, such as an extension number, or a physical phone. Callers can dial directly into a phantom mailbox and the auto attendant can route voice messages to

the phantom mailbox. For example, a business person who travels frequently can use a phantom mailbox to retrieve messages. A phantom mailbox does not have message waiting indication and is not required to have a telephone configured on the communications system.

POTS

Plain Old Telephone Service, which is the standard telephone service that most homes use.

PRI

A Primary-Rate Interface is a type of ISDN service designed for larger organizations.

PSTN

The public switched telephone network refers to the international circuit-switched telephone system. It includes analog and digital lines.

Remote mailbox management

Remote mailbox management allows users to access their mailbox from an external, or remote, telephone.

Router

A device that forwards data packets along networks. A router is connected to at least two networks, commonly two LANs or WANs, or a LAN and its ISP's network. Routers are located at gateways, the places where two or more networks connect.

RTP

The Real-time Transport Protocol (RTP) transmits audio and video data over the Internet.

SIP

Session Initiation Protocol is a signaling protocol that establishes sessions in an IP network for features such as multi-media conferencing or call forwarding deployed over IP networks. This allows service providers to integrate basic IP telephony services with Web and email. SIP server also supports traditional telephony features, such as personal mobility, time-of-day routing, and call forwarding.

TCP

Transmission Control Protocol is one of the main protocols in TCP/IP networks. Whereas the IP protocol deals only with packets, TCP enables two hosts to establish a connection and exchange streams of data. TCP guarantees the delivery of data, and that packets are delivered in the same order that they are sent.

Text-to-speech

The text-to-speech engine reads back email messages over the phone when users call in to review their messages.

TFTP

Trivial File Transfer Protocol is a simple form of the File Transfer Protocol (FTP). It provides no security

features, such as password protection when downloading software or transferring files.

UA

A user agent is the client application used with a particular network protocol, particularly with regards to accessing the World Wide Web.

NetVanta UC Server access number

The UC server Mail access number is the number you enter using your telephone keypad when you want to access your messages from your telephone extension.

UDP

User Datagram Protocol, like TCP, runs on top of IP networks. UDP is a direct way to send and receive datagrams over an IP network. UDP is primarily used for broadcasting messages over a network.

URI

A Uniform Resource Identifier (URI) refers to a string of characters used to identify or name a resource or object. The primary purpose of this identification is to enable interactions with resources or objects over a network, usually the World Wide Web.

User

A user is someone who has a UC server profile on the system.

Unified Messaging

Unified Messaging allows users to manage all message types from one central storage location. This means a single inbox for voice, email, and fax messages that is accessible by both telephone and computer. The UC server provides a unified messaging solution that allows you to receive all your messages in the Microsoft Outlook Client. Users receive all of their messages at once, in one convenient location.

Appendix B - Installing and Uninstalling Software

Overview

This appendix outlines the procedures for installing and uninstalling software for the UC Server.

Installing Active Directory Users and Computers Console

If you do not have automatic access to Active Directory Users and Computers console, you must install the Microsoft snap-in. The DLLs that you require are:

- CAMMCUserExt.dll
- CAMMCUserContextMenuExt.dll

The server installation puts these files in the BIN folder with the other executable files. The server installation also “registers” the DLL with Windows so that, on the server, the UC Server extensions are available immediately after installation. This means that, if you start the Active Directory Users and Computers MMC snap-in, you can see the UC Server property page and context menu extensions.



You must have the Administrator pack installed on the client operating system for the DLLs to function properly.

You can use these extensions on another computer.

To make the MMC extensions available on another computer

1. Copy the DLLs from the server computer to another folder on the computer on which you want them to be available. For example: **C:\Program Files\ADTRAN\UC Server\Bin**
2. Register the DLLs by executing the following programs:
regsvr32 "C:\Program Files\ADTRAN\UC Server\Bin\cammcuserext" and
regsvr32 "C:\Program Files\ADTRAN\UC Server\Bin\cammcusercontent"



In both cases, a dialog appears indicating that the registration is successful.

You do not have to do further configuration. However, both extensions support the use of a configuration file called “DomainsToSearch.cfg”. Either extension searches for registered UC Servers on the local domain and any domains connected to it with which it has a two-way trust relationship. If any of the domains are inaccessible (for example, one domain is a test domain and its domain controller is shut down), it can take up to 20 seconds for MMC to determine this, which slows initialization. To overcome this problem, you can specify the explicit list of domain names to search for UC Servers. These names are put, one per line, in the configuration file. If there is no configuration file, then all domains are searched.

The configuration file (if it exists) is in the same folder as the DLLs.

Uninstalling Server Software

Uninstalling the server software does not automatically delete the UC Server user account or remove the Exchange mailbox or Exchange permissions for the UC Server.

To uninstall the server software

1. Select Windows **Start > Settings > Control Panel**.
2. Select the **Add/Remove Programs** icon.
3. From the list of programs, select **UC Server**.
4. Read the instructions and then select **Remove**.
5. Restart the computer. All components of the software are removed.

Deleting the server user account

Deleting the server user account does not remove the Exchange mailbox for or Exchange permissions.

To delete the server user account

1. Select Windows **Start > Programs > Administrative Tools (Common)**.
2. Select **User Manager for Domains** to open the User Manager for Domains dialog box.
3. From the list of users, select **UC Server**.
4. Press **Delete** on the keyboard.
5. Confirm the deletion at the warning prompts.

Appendix C - Troubleshooting Logs

Overview

The UC Server tracks system errors and warnings in log files, which may be used for troubleshooting. This section includes potential system errors and warnings that may occur, and possible causes for and solutions to these issues.



Not all errors and warnings necessarily require action.



For more information on troubleshooting, refer to [Troubleshooting on page 315](#).

Troubleshooting System Error Logs

The following is a list of potential system errors that may be noted in the System Errors and Warning logs. To troubleshoot a system error, search the list below for the log entry, and note the possible causes of and solutions for the error.

Severity	Log Entry	Possible Causes and Solutions
Serious Error	Failed to create {identity} %s. Internal error %d.	Verify that all UC Server processes are started, and retry creating the identity.
Serious Error	Failed to deliver a message to user '%s'	A message that was queued for message delivery has given up attempting to deliver the message. The messages will be delivered to the system defined default mailbox. Verify the connection and permissions to the message store.
Serious Error	Failed to get server object for user '%s'.	Verify network connectivity. Verify the UC Server's service accounts file permissions on the .UC Server folder.
Serious Error	Failed to update Active Directory user's extensions (hr=0x%08x).	Verify network connectivity and the connection to Active Directory. Verify the UC Server's service accounts file permissions on the .UC Server folder. Re-run the Server Configuration Wizard.

Severity	Log Entry	Possible Causes and Solutions
Serious Error	Internal error finding the Exchange Server message store.	Verify the connection to Microsoft Exchange Server. Validate the UC Server's service account has the appropriate permissions.
Serious Error	System diagnostics has detected a potential problem with port %s.(port displayed as [name [extension, deviceId]]).	The port auditing system has detected a problem with one or more of the system ports. Possible problems: PBX port is bad Wiring is connected to the wrong port on the PBX. Call each port individually and monitor in the UC Server Ports Manager to ensure that calls are being received on the expected port. Wiring is bad. Verify the wiring.
Serious Error	The minimum disk storage threshold has been exceeded. New voice messages will not be saved until the problem is corrected.	Increase the disk storage, or remove any files that are not required.
Fatal Error	Application Services have been restarted to recover from an error condition. In most situations, the system will be operating normally after this occurs. If this log entry appears regularly or frequently, then more investigation is required.	For more information, refer to the following section Troubleshooting the Application Server Restarted Log Error on page 338 .

Troubleshooting the Application Server Restarted Log Error

The following log entry may appear in the System Errors and Warning Logs:

Error: Application Server Restarted

Description

Application services have been restarted to recover from an error condition. In most cases the system will operate normally after this occurs. However, if this log entry appears regularly or frequently, then further investigation is required.

Potential Causes

- Microsoft Exchange Server is not responding in a timely manner.
- Slow Microsoft Active Directory responses
- Slow system behavior (e.g., the CPU is overloaded)
- Other activities are occurring which is suspending or slowing down operation (e.g., virus scanning or search indexing)
- Software issues

To investigate potential causes of the restart log entry

1. Validate the correct operation of voicemail and auto attendants.
2. Validate the NetVanta UC client connectivity to the server.
3. Validate access to user mailboxes on all mail servers.
4. Review previous logs of a similar nature to see if there is a pattern (for example, they all occur at 3:00 am).
 - If there is an obvious pattern in the logs, consider the possibility of other activities interfering, such as virus scans, application updates and so on.
 - If logs occur rapidly or frequently, there is likely a problem in the configuration or operation. In this case contact ADTRAN Technical Support.
 - If logs occur infrequently, then continue with normal operation.



For information on contacting ADTRAN Technical Support, refer to [Accessing Technical Support on page 327](#).

Troubleshooting Warning Logs

The following is a list of potential warnings that may be noted in the System Errors and Warning logs. To troubleshoot a warning, search the list below for the log entry, and note the possible causes of and solutions for the warning.



Not all warnings necessarily require action.

Log Entry	Possible Causes and Solutions
'%s' display pattern not defined	The Definity PBX configuration file (.\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and a required display pattern is missing.
Call reason code %s is invalid in key : %s	An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and a specified call reason code for an In Band Signalling key doesn't exist in the valid call reasons defined in the same file.
Cannot write to file xxx.	Verify the UC Server's service accounts file permissions on the .\UC Server folder. Re-run the Server Configuration Wizard.

Log Entry

Possible Causes and Solutions

<p>Configuration file could not be copied: %s</p>	<p>Occurs when the PBX-specific configuration file doesn't exist and the template configuration file cannot be copied. Verify the UC Server's service accounts file permissions on the .\UC Server folder. Re-run the Server Configuration Wizard.</p>
<p>Configuration file does not exist: %s</p>	<p>The file .\UC Server\Data\System\CallInfoOverride.cfg has been edited manually and one or more of the entries in this file are invalid. Consult the inline documentation for details about valid syntax rules.</p>
<p>Display format definition error, no matching \" found : %s Display format error, illegal character '%c' is found : %s</p>	<p>The Definity PBX configuration file (.\\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and an error was encountered in parsing a 'display format' entry.</p>
<p>Duplicate override spec : %s in key %s Duplicate pattern type %c in match pattern : %s in key %s</p>	<p>An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and an In Band Signalling key's match pattern has been duplicated.</p>
<p>Error : '%s' compiling regular expression : %s in key %s</p>	<p>An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and an In Band Signalling key's match pattern cannot be converted to a regular expression.</p>
<p>Error : '%s' compiling regular expression '%s' from display rule '%s'</p>	<p>The Definity PBX configuration file (.\\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and an error was encountered in parsing a 'display format' entry.</p>
<p>Error in reason code translation, at most two characters are allowed in each field : %s</p>	<p>The Definity PBX configuration file (.\\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and a reason code translation has more than 2 characters in a field (fields are separated by a ",").</p>
<p>Error in repeat count after '_' in : %s Error in repeat count after '%c' in : %s</p>	<p>The Definity PBX configuration file (.\\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and an error was encountered in parsing a 'display format' entry.</p>
<p>Exception %s caught, Wrong format in override rule file on line : %s Exception caught, Wrong format in override rule file on line : %s</p>	<p>The file .\UC Server\Data\System\CallInfoOverride.cfg has been edited manually and one or more of the entries in this file are invalid. Consult the inline documentation for details about valid syntax rules.</p>
<p>Failed to parse pattern line '%s', ignoring</p>	<p>The Definity PBX configuration file (.\\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and a display pattern has an invalid value.</p>

Log Entry	Possible Causes and Solutions
Failed to retrieve server address	<p>The UC Server is unable to find the network address on the associated network adapter. This commonly occurs if the files were copied from one UC Server machine to another.</p> <p>To correct, navigate to the communications system network information and re-select the adapter that you want to bind with the UC Server.</p> <p>Once selected, select OK.</p>
Failed to update %s's database with the received fax. The fax will be saved instead to the default mailbox.	<p>There was a problem communicating with the database defined in the Receive Fax element.</p> <p>Check your database connection parameters, and ensure that the database is available.</p>
Illegal characters found in allowedNondigitCharactersInExternalNumber %s	<p>The Definity PBX configuration file (.\\UC Server\\Data\\System\\Definity*Definition.cfg) in use has been edited manually and the "allowedNondigitCharactersInExternalNumber tag has one or more invalid characters.</p>
Illegal characters found in literals in %s	<p>The Definity PBX configuration file (.\\UC Server\\Data\\System\\Definity*Definition.cfg) in use has been edited manually and an error was encountered in parsing a 'display format' entry.</p>
Incompatible match pattern : %s or overrides in key %s	<p>An analog PBX configuration file (located in .\\UC Server\\Data\\System) has been edited manually and an In Band Signalling key's match pattern is invalid.</p>
Input file xxx does not exist	<p>Verify the UC Server's service accounts file permissions on the .\\UC Server folder.</p> <p>Re-run the Server Configuration Wizard.</p>
Invalid call progress tone spec : %s	<p>A PBX configuration file (located in .\\UC Server\\Data\\System) has been edited manually, and "StandardTone[x]" (where [x] = 1...9) is defined and formatted incorrectly.</p>
Invalid disconnect spec : %s	<p>An analog PBX configuration file (located in .\\UC Server\\Data\\System) has been edited manually and there is an error with the value for the tag "disconnectType." It must match one of the following values:</p> <ul style="list-style-type: none"> positive disconnectTone disconnectDTMFSequence disconnectSilence
Invalid disconnect tone spec : %s	<p>An analog PBX configuration file (located in .\\UC Server\\Data\\System) has been edited manually and the tag disconnectType has a value of disconnectTone. However, the required corresponding tag disconnectTone is missing from the same configuration file.</p>

Log Entry	Possible Causes and Solutions
Invalid override spec : %s in key %s	An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and an In Band Signalling key's match pattern is invalid.
Key 'disconnectType' not defined : %s	An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and the tag 'disconnectType' is not present.
No key is defined in section : %s	An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and all keys are missing from the section "In Band Signalling".
No match pattern are defined in key : %s	An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and an In Band Signalling key is missing a match pattern (i.e., the value after the ":").
No reason code are defined in key : %s	An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and the reason code (i.e., the value before the ":") for an In Band Signalling key is missing.
Pattern length(%d) differs from display length(%d) in pattern : %s	The Definity PBX configuration file (.\UC Server\Data\System\Definity*Definition.cfg) in use has been edited manually and an error was encountered in parsing a 'display format' entry.
Pattern type %d not supported yet	A PBX configuration file (located in .\UC Server\Data\System) has been edited manually, and in-band signaling protocol definition was added that is not yet supported.
System did not receive Caller ID information from the modem and hence, could not transmit it to your pager. Ensure that your phone rings at least twice before answering otherwise you will not receive Caller ID information.	Verify that Caller ID is available when attempting to send a pager notification which includes sending the Caller ID.
The current number of ports has exceeded the number of licensed ports.	<p>Verify that you have enough ports or application server channels licensed.</p> <p>To verify the licensing, navigate to Help > Licensing Information to refer to the existing licensing capabilities.</p> <p>Contact your ADTRAN reseller to purchase additional ports or application server media channels.</p>
The data source, %s, referenced in the element, %s, is unknown.	<p>ServiceStateMachine when an element refers to a data source that no longer exists. Verify the following:</p> <ol style="list-style-type: none"> 1.The ODBC data source, if applicable, is correctly configured 2.The fields in the reference data source may have been changed or deleted. Verify the data source configuration.

Log Entry**Possible Causes and Solutions**

The specified number was disallowed by dialing restrictions.

Occurs when the application server attempts to dial a number whose pattern matches a dial restriction entry (configured in Tools > Options > Dialing).
Verify the application server toll restrictions.

Unable to communicate with remote agent '%s'

The application server was unable to communicate with the IP Office remote agent.
Check the remote agent properties, IP address, port, and network connection state.

Unable to deliver a message to user '%s'

There was a problem delivering a voice or fax message to a user.
Check your message store and network connection states to ensure that the connection to the message store is available. This error occurs on the first message delivery failure.

Unrecognized character %c in match pattern : %s in key %s

An analog PBX configuration file (located in .\UC Server\Data\System) has been edited manually and an In Band Signalling key's match pattern has an invalid character.

Wrong format in override rule file on line : %s...

The file .\UC Server\Data\System\CallInfoOverride.cfg has been edited manually and one or more of the entries in this file are invalid.
Consult the inline documentation for details about valid syntax rules.