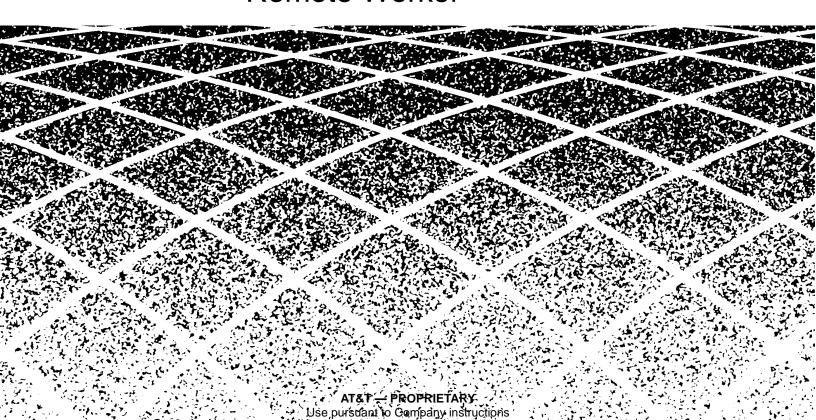


Voice DNA® User Guide for Voice DNA and Remote Worker



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AT&T Voice DNA User® Guide

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AT&T Voice DNA User® Guide

Welcome

Welcome to the AT&T Voice DNA® User Guide! This User Guide has all the information you will need to make best use of this exciting new telecommunications product.

This service is an Internet Protocol-based secure phone system that offers features you've never seen on your traditional phone—like Locate Me, which can ring a series of phone numbers, in sequence or all at once—and web-based conference calls—and voicemail that can be forwarded by e-mail—and call logs at your fingertips for both incoming and outgoing calls—and notifications by e-mail when you've missed an incoming call. It allows you to unite your office phone lines, cell phone numbers, home phones, e-mail and pagers into your own personal communications web—a system that can find you and follow you almost anywhere. It harnesses the power of the Internet and your computer to give you communication powers like those that you have never had, and the ability to organize your communications in extraordinary ways.

Using the AT&T Voice DNA® Service, you can make and receive calls from within your organization and to any telephone connected to the Public Switched Telephone Network.

You can access the service via the web site at the AT&T BusinessDirect Web site (https://www.businessdirect.att.com/). This User Guide covers the features available via the web site, using the Click to Call technology on your PC screen.

This User Guide is organized by task. Just look for the thing you are trying to do, and you will find detailed instructions there for making the best use your service.

Audience

This guide is intended to be used by Voice DNA users and the Voice DNA Remote Worker users. This guide provides a description of the features and tasks that users may perform in their normal daily operations.

Web Interfaces

There are two separate web interfaces:

- Administrator Lets phone managers add users and configure user features, numbers, and other company-wide features.
- Personal Lets users access and personalize their telephone features. This guide should be read by users who need to use the Personal Web Site and the Voice DNA phone to configure and use their voice services. This user guide assumes that the reader understands how to use an Internet browser.

Getting Started

To use AT&T Voice DNA Service, you will need the following:

- A Web Browser, preferably Microsoft Internet Explorer 6.0. Currently, Microsoft Internet Explorer 7.0 is not supported. The Safari browser running on an Apple Mac computer can be used with the Personal Web Site. Some features are not supported while others may work slightly different than when used on a PC. Refer to Section Mac/PC Differences for more information about Mac/PC differences.
- Your AT&T BusinessDirect Login. This login is supplied in your Welcome letter that is emailed to you. If the login or password has leading or trailing spaces, it will not work. If an answer to a security question was entered for you by your administrator, then the welcome letter will also contain your temporary password. If a security question was not entered by your administrator, then you will receive the Welcome letter and a separate e-mail containing your temporary password.
- An IP Phone desk phone or a softphone on your PC. The softphone needs the PC to have a microphone and speaker or a headset (earpiece and microphone).

Logon Procedures

Initial Logon

For the initial logon to Voice DNA, use the following procedure:

 Log on to AT&T BusinessDirect via the BusinessDirect logon page (Figure 1): https://www.businessdirect.att.com/

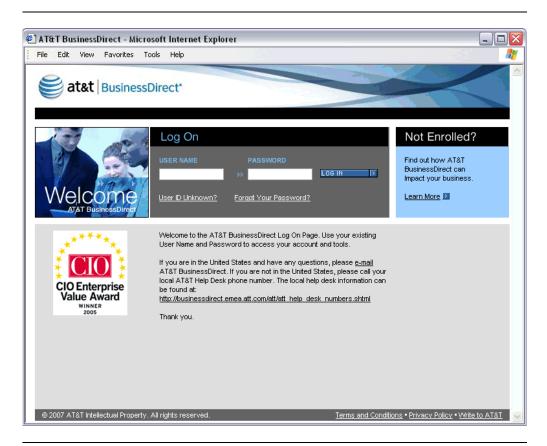


Figure 1. AT&T BusinessDirect Logon Page

2. Enter your user name and temporary password then click on the LOG IN button. The user name is supplied in your Welcome letter that was emailed to you. You should receive the Welcome letter and a separate e-mail containing your temporary password. If you cut & paste your user name and temporary password from the BusinessDirect Welcome letter, you may also copy leading spaces along with the text. If the login or password has leading or trailing spaces, they will not work.

- 3. When you login with your user name and temporary password, the Update Your Personal Profile screen is displayed. Located at the bottom of the screen are three security questions and answers which you are required to enter. The first Security Question is used by AT&T to authenticate your account. The second and third Security Questions are used to reset your password. Select a question from the Security Question drop down menu and enter the answer in the Security Answer field for all three items. Click on the SUBMIT button to save the data.
- 4. Since you are using a temporary password the first time you login, the system requires you to change this temporary password. After entering your user name, temporary password and **Security Questions**, the BusinessDirect Change Password screen similar to Figure 2 is displayed.

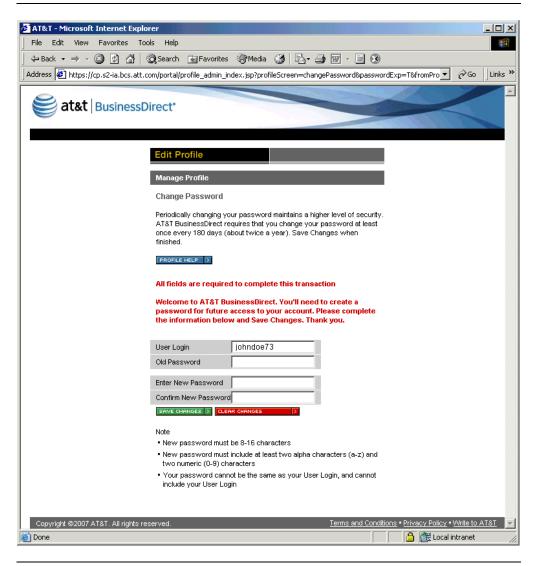


Figure 2. BusinessDirect Change Password Screen

- 5. Enter your old password in the **Old Password** field.
- Enter your new password twice. Once in the Enter New Password field and once in the Confirm New Password field. Both fields must contain the same entry.

You are allowed to choose your own password, however it must meet AT&T guidelines. Your password must follow these rules:

- a minimum of 8 but no more than 16 characters
- at least 2 alpha characters (a-z, A-Z)
- at least 2 numbers (0-9)
- special characters, such as @, #, and + are permitted.
- at least 3 characters must be different by position from the old (current existing) password
- when comparing the old and new passwords, case change of an alphabetic character by position will not qualify as a change (e.g., If Golf=2005 is the old password, gOlf=2005 would not be acceptable as a new password)
- must not be a reverse of the old (current existing) password
- password cannot be the same as your user ID and cannot include your user ID.

For greater security, you may wish to include a combination of Case Sensitive characters and/or Special Characters.

7. Click the **SAVE CHANGES** button to save the new password. The system responds with a confirmation screen similar to Figure 3.

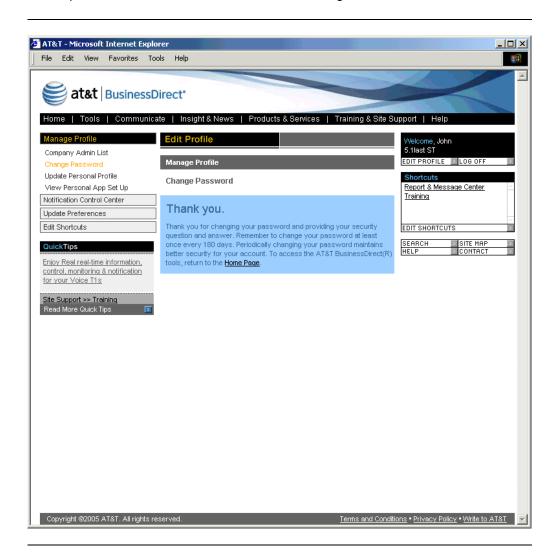


Figure 3. Successful Changing of Password Screen

8. Clicking on the <u>Home Page</u> link in the <u>Thank You</u> text (Figure 3) takes you to the BusinessDirect home page. From the BusinessDirect homepage, select the <u>Access Voice DNA Service</u> application link located under the <u>Inside Tools</u> menu. An example is shown in Figure 4.

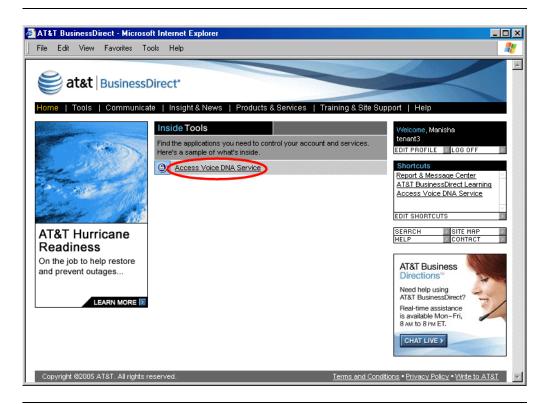


Figure 4. Inside Tools – Access Voice DNA Service

9. Clicking on the <u>Access Voice DNA Service</u> link takes you to the Personal Web Site similar to the following:

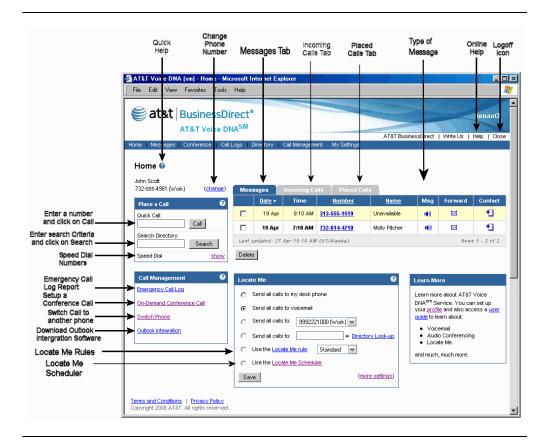


Figure 5. Personal Web Site

≡>NOTE 1:

Not all features shown in Figure 5 are available to all users. Depending on your Feature Package (Class of Service), some features may be unavailable to you. For example, the Call Management frame only contains the features to which you have access.

■> NOTE 2:

If the user is an extension only user, the extension (phone number) is not displayed on the Personal Web Site page.

Bypassing the AT&T BusinessDirect Home Page

Once on the Personal Web Site, bookmark the Personal Web Site (Figure 5). Going forward, clicking on this bookmark brings up the BusinessDirect logon page. Once you enter your user name and password, you will be taken directly to the AT&T Voice DNA application home page. This allows you to by-pass the BusinessDirect home page after the initial login to BusinessDirect.

Log Off Procedures

Logging Off

At the top right-hand portion of any page, click the **Close** link (see Figure 5). This hyperlink logs you off and closes the window. Clicking on the **Close** link causes the system to respond with a pop-up window asking **Do you want to close AT&T Voice DNA Service?** Click on the **OK** button to close the service or the **Cancel** button to continue. Use the **Close** link instead of closing the window by clicking on the **X** icon.

User ID Unknown

If you do not know you User ID, click on the <u>User ID Unknown?</u> link on the BusinessDirect login screen (Figure 1) and follow the instructions on the screen.

Forgot Your Password

If you have forgotten your password, click on the <u>Forgot Your Password?</u> link on the BusinessDirect login screen (Figure 1) and follow the instructions on the screen. To confirm your identity, you will need to provide your two Security Questions and Answers from your profile. You will have 3 attempts to enter this information correctly. After 3 incorrect attempts, your account will be locked and you will need to contact your Company Administrator for AT&T BusinessDirect. If you enter this information correctly, a **Change Password** screen is displayed. You can then enter your new password on this screen.

Password Aging

For greater security, you are required to change your password every 180 days. However, you have the option to change your password at any time.

If you attempt to log in after your password has expired, the **Password Change Page** is displayed where you will be required to change your password before proceeding.

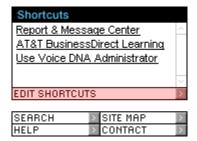
Default Window Size

The default window size and placement depends on your browser settings.

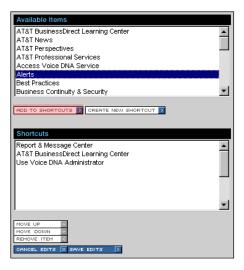
Alerts Settings

It is recommended for each user (Administrators and users) to add **Alerts** to their **Shortcuts** window. This link allows you to view **Alerts** on the BusinessDirect home page for maintenance activities. To add **Alerts** to the **Shortcuts** window, perform the follow steps:

1. Click on the **EDIT SHORTCUTS** link on the BusinessDirect home page.

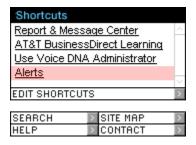


2. Select **Alerts** from the **Available Items** portion of the window and click on the **ADD TO SHORTCUTS** button.



The Alerts item will move to the Shortcut portion of the window.

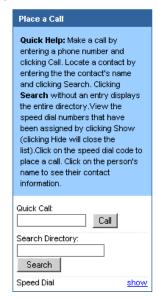
Click on the SAVE EDITS button. The Alerts item will now appear under the Shortcuts portion of your BusinessDirect home page.



How to get Help for AT&T Voice DNA

You can access help support for AT&T Voice DNA in the following ways:

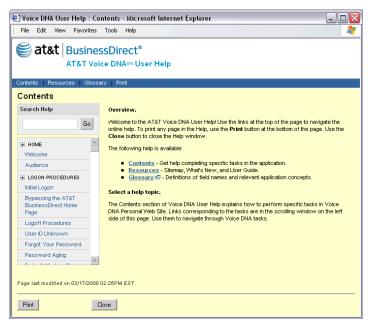
■ Quick Help – Quick Help provides short description of the item in a window that is integrated with the window. Any time that you see the symbol, clicking on the question mark shows the quick help for this item. The following shows an example of the Quick Help for Place a Call.



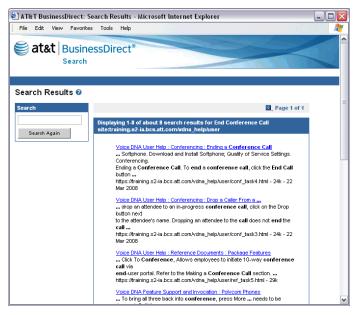
■ Online Help – Online help is accessed by clicking the header link Help or using the pull down menu as shown in the following example by clicking on the Help link.



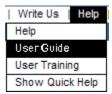
After clicking on the **Help**, the system responds with the online help as shown in the following example.



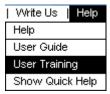
The online help contains all of the information that is included in the User Guide. You can use the **Search Help** box to search the help. For example, if you wanted to know how to end a conference call, enter **end conference call** into the **Search Help** box and click the **Go** button. The system responds with the help entries (as shown in the following example) that match the criteria that was entered in the **Search Help** box. You can then click on one of the matching links to view the help or perform an addition search if necessary.



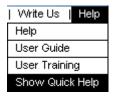
■ User Guide — The detailed User Guide is access through the header link Help by clicking on the User Guide link.



■ User Training – This link takes you to the online User Training.



■ Show Quick Help — To see the Quick Help on the window, click on Show Quick Help link under the Help Header.



This displays all of the Quick Help throughout the application. To hide the Quick Help, click on the **Hide Quick Help** link.

Placing a Call

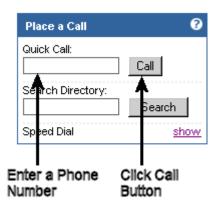


Figure 6. Place a Call Field

You can place and receive calls using your Web-enabled desk phone and via the Personal Web Site, using the Click to Call technology on your PC screen.

To make a call from the Personal Web Site:

- 1. Go to your Personal Web Site.
- 2. Find the Place a Call box.
- 3. Enter the number to call in the Quick Call field.
- 4. Click the **Call** button. The system responds with a Phone Call screen similar to the following.

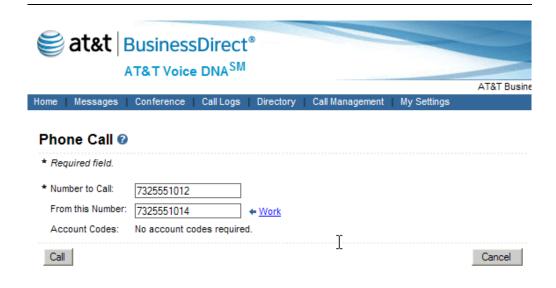


Figure 7. Phone Call Screen

■NOTE 1:

When using Click to Call to call off-net, the off-net prefix **must** be entered, especially in cases where the first digit of the area code is the same as the off-net prefix. For example, if my number is 732-555-3136 and I want to call 908-555-3889. On the Personal Web Site, I must enter 9-908-555-3889 or 9-1-908-555-3889 in order for the call to go through.

■NOTE 2:

For the following types of incomplete call attempts, the user will hear a fast busy:

- Invalid number
- Network congestion
- Number changed
- Number not in service
- Number not assigned

■>NOTE 3:

You should not use Click to Call when calling 911.

■>NOTE 4:

When using Click to Call your name and number will appear in the Caller ID window of the phone as you are making the call from as well as the called party.

5. Click on the Call button to initiate the call. Note that if there were no Account Codes entered by the administrator, none would appear on the screen. If the Account Code is not required, the message No account code required. appears in place of the drop down menu. For more information on Account Codes, see the section entitled Account Codes.

When placing a call using your desk phone, after dialing press the send/dial soft key (enter # after the dialed digits if using a Cisco ATA 186) to immediately send the dial string (see your specific phone's user manual for key terminology). Otherwise, you will hear momentary silence while the phone waits for additional digits.

■NOTE:

The Quick Call link can use Click to Call or Remote Click to Call:

- The Click to Call is a feature allows you can click on any underlined telephone number (from the directory, call logs, voicemail logs, etc.) on the Personal Web Site so that a pop-up window is displayed that allows you to launch a call that will first ring your work telephone number, then ring the number you selected.
- The Remote Click to Call is similar to the Click to Call feature except you're not constrained to using your work telephone number as the start number (e.g., you can specify your cell or any other number). The called party still sees your AT&T Voice DNA work telephone number as the caller id even though you really started the call from an alternate telephone number/device. Remote Click to Call requires a COS setting to be enabled for the user by the Administrator.

Placing a Speed Dial Call

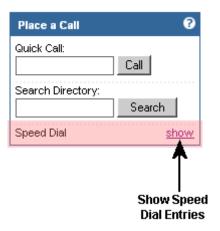


Figure 8. Speed Dial Link

You can place a Speed Dial call (previously programmed by you in the Settings menu on navigation bar) from your Personal Web Site like this:

- 1. Go to the Place a Call box on the Personal Web Site.
- Click the <u>show</u> link next to Speed Dial.
- Your Speed Dial numbers are displayed in tabular form as shown in the following example.



Figure 9. Example of Speed Dial Numbers

4. Click the Speed Dial number you want to call. The system responds with a Phone Call screen similar to the following. If you click on the name (e.g., Susan Blackwood) the Edit Contact screen populated with this person's contact information is displayed.



Figure 10. Phone Call Screen

5. Click the Call button in the Phone Call screen. Note that the Account Code for this call would be 10000. If the Account Code is not required, the message No account code required. appears in place of the drop down menu. The system responds with Call in Progress screen similar to Figure 11. When you phone rings, answer the phone to connect the call.

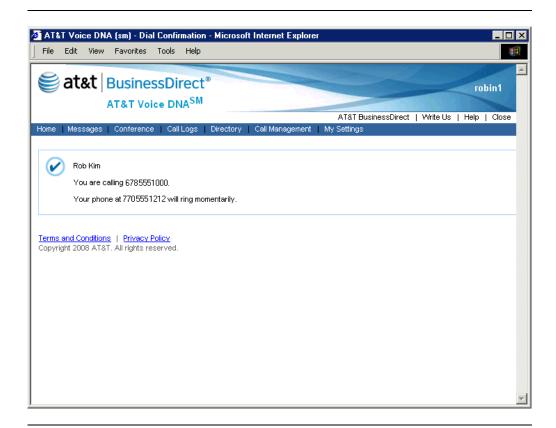


Figure 11. Call in Progress Screen

■NOTE 1:

For more information about Speed Dial, see the section entitled <u>Using Speed</u> <u>Dial</u>.

■>NOTE 2:

The Speed Dial and Optional Account Codes feature will not work together.

Messages

You can access your messages from the **Messages** tab on the **Home** page (Figure 5) or the **Messages** item (Figure 13) on the navigation bar menu. Messages can consist of voicemails, faxes, or e-mails from AT&T Voice DNA. However, you must be subscribed to the Voicemail feature (by your administrator) to have access to this feature.

Your mailbox capacity is set at 100MB. As a guideline, each one-minute message left in your mailbox equates to approximately 1 MB of storage capacity. If your mailbox capacity threshold is exceeded, an e-mail appears in the Messages list. Clicking on the e-mail icon displays the contents of the message.

The maximum number of messages you can store before affecting the date/number sorting functionality is 150 The following error message will be output as a warning when this threshold has been reached: "Your Mailbox messages have exceeded the maximum number of messages that can be sorted (150). Message sorting capabilities are disabled until you delete some messages."

Contact your Administrator to discuss storage capacity options.

■NOTE 1:

The light on your phone that indicates a new message will turn off once you have listened to all of your new messages (from either the phone or web site). New messages are displayed in bold font on your Personal Web Site.

■NOTE 2:

The **Name** column will only be populated if this information from the Voicemail product is available. Otherwise, the **Name** column will display **Unavailable**.

■NOTE 3:

If you are accessing your messages from an unsubscribed phone and Account Codes are mandatory, you must enter an Account Code to access your messages.

■NOTE 4:

When you delete a message from the phone or web site, it deletes the message for all views. For example, if you delete a message using the phone, it is also deleted from your Personal Web Site.

■>NOTE 5:

The sort order for voicemail messages is urgent messages first then it defaults to last-in-first-out (LIFO).

■NOTE 6:

Messages do not come in automatically or in **real time**. To get your latest messages, use the refresh icon on the browser or the **F5** key to get the latest messages.

■NOTE 7:

Messages that have not been read are displayed on the Personal Web Site in a bold font. Once the message has been read, it is displayed in a regular font. Messages retrieved via the Telephone User Interface (TUI) need to be saved in order for message to be flagged as having been read. Only listening to the message from the TUI does not flag the message as being read on the Personal Web Site.

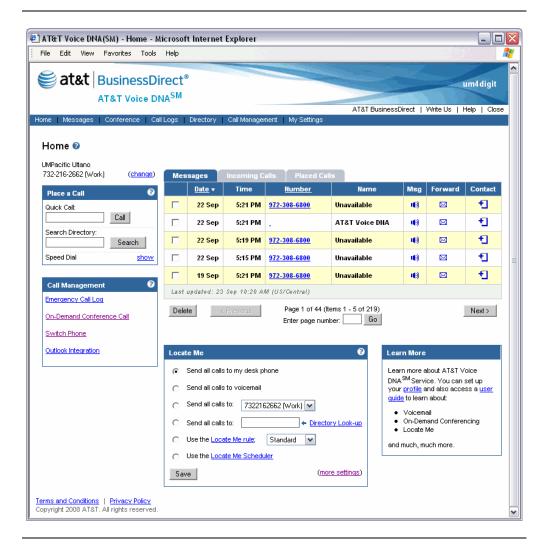


Figure 12. Messages Tab

≡>NOTE 8:

Only the last five messages are displayed on the Home Page. Use the **Previous** and **Next** buttons to step through the messages, or enter the page number and click on the Go button. You can click the **Messages** menu item to see a separate screen with a list messages. Figure 13 shows an example of **Messages** screen that contains a list of your messages.

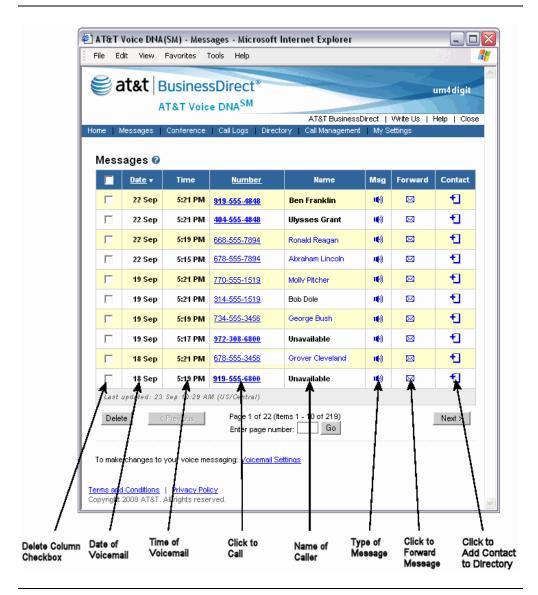


Figure 13. Messages List

From this page, you can listen to your Voicemail, click to dial callers back, edit and save Contact information, forward Voicemail via e-mail as **.wav** file attachments, and manage the **Messages** list.

≡>NOTE:

To listen to a voicemail, your PC must have Windows Media Player along with speakers or a headset. If you don't have Windows Media Player, you will need to save the file to your PC and then open it with the media player you have on your PC.

Fax Messaging

As an AT&T Voice DNA User subscribing to this Voicemail feature, you have the ability to receive inbound faxes in your AT&T Voice DNA Mailbox. This capability is in addition to other messaging functionality.

By dialing into the voicemail Telephone User Interface (TUI), you can hear header information for all voice and fax messages. Via the TUI, you can:

- Send a fax to a fax machine
- Forward a fax to another User's mailbox
- Disable or enable the fax messaging capability

When a fax is received, you are notified based on your message notification options, such as Message Waiting Indicator (MWI), email, and pager notifications. After you display the entire fax or listen to the fax header information on your phone, the MWI is deactivated.

Figure 14 shows fax messages and fax icons displayed on a User's Personal Website, Messages tab. The Messages tab displays both voicemail and fax messages, with fax messages indicated by a fax icon. From the Messages tab you can:

- Display the full fax message by clicking on the fax message
- Print the displayed fax message (on a printer, not a fax machine)
- Forward the fax message to an email address

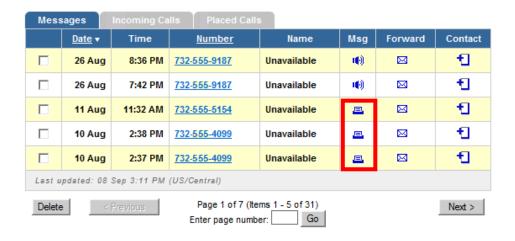


Figure 14. User's Personal Website - Messages Tab

Listening to a Message

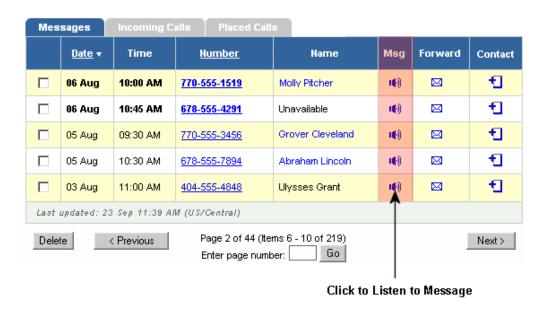


Figure 15. Messages Tab

Listen to Voicemail Message

1. Click the Voicemail Message icon 🗐 in the Msg column for the message.

Returning Calls to Mailbox Access Numbers

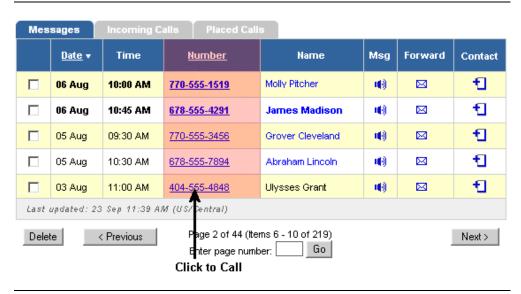


Figure 16. Messages Number Column

To call someone who has left you a message:

- 1. Click the telephone number in the **Messages** table.
- 2. This gives you the **Phone Call** screen.
- 3. Click the **Call** button to call that number.

Forwarding a Message by E-Mail

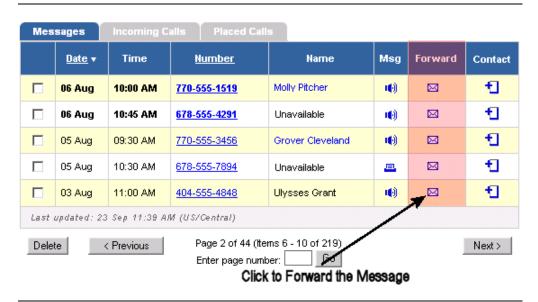


Figure 17. Messages Forward Column

To forward a Voicemail or Fax message by e-mail:

- 1. Click the envelope symbol

 or fax icon

 for that entry line.

 □
- 2. This gives you the following screen (Figure 19), from which you can forward the Voicemail or Fax (automatically attached) with a subject. Enter the e-mail address of the recipient in the Send To: field. You can use the Directory Look-up link if these participants are included in the Directory. To use the Directory Look-up, click on the Directory Look-up link. The system responds with Advanced Search window.



Figure 18. Advanced Search Window

Enter in the search criteria and click on the **Search** button. Note that, for each search string, you can specify **contains**, **exact**, **starts with**, or **ends with** (wildcard characters are not supported). The system responds with the results that match the search criteria. If you want to forward the message to a user, select the option button to the left of their name and click on the **Select** button. You can only enter 1 e-mail address to which to forward the message, however you can forward the e-mail to additional e-mail addresses but only one at a time.

When you click the **Forward** button, a Voicemail message is sent as a **.wav** file attachment. A Fax message is sent as a **.tif** file.

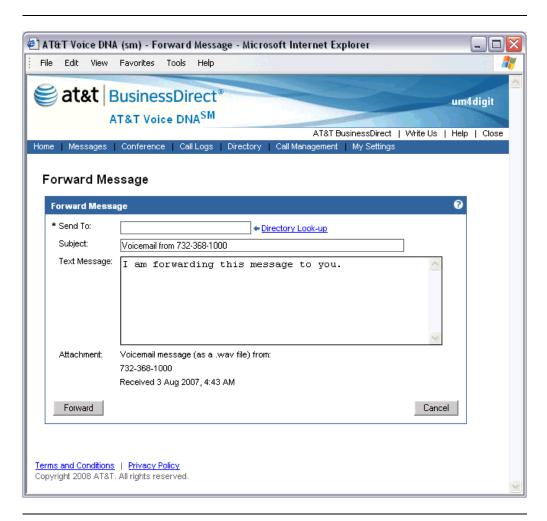
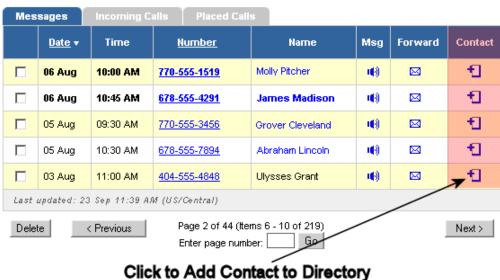


Figure 19. Forward Voicemail Message Window

Adding a Contact from the Messages List



Click to Add Contact to Director

Figure 20. Messages Add Contact Column

To add a contact from the Messages list to your Directory:

1. Click on the Contact symbol ¹ for that message.

Deleting Messages



Figure 21. Message Delete Checkbox and Button

To delete a message:

- 1. Click the checkbox located to the left of the message.
- 2. Click the **Delete** button just below the **Messages** table.

■NOTE:

Messages deleted from the Personal Web Site cannot be restored. Messages deleted using the Telephone User Interface (TUI) can be restored up to approximately 48 hours later.

Editing Directory Contact Information for a Message Name

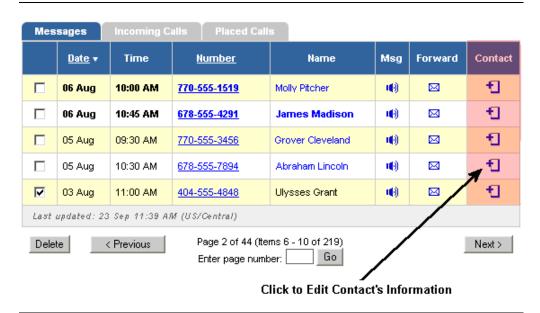


Figure 22. Messages Name Column

■NOTE:

The **Name** column will only be populated if this information is available from the Voicemail product. Otherwise, it will display **Unavailable**.

To edit the contact information for someone listed in the **Messages** table:

- 1. Click on the Contact symbol ¹ for that entry line.
- 2. This gives you the **Edit Contact** screen for that person, where you can edit the information listed. If the contact is in the Company Directory, most of the fields are display only.

■NOTE:

When editing a contact that is in the company directory, you can only make 2 changes:

- add the contact to a call group (for locate me treatments)
- assign a personal speed dial to the contact.

You cannot edit any of the corporate directory information for the contact. Only the administrator can add/edit corporate directory information.

Adding or Changing Mailbox Settings

Determine Your Mailbox Access Number

Your mailbox access number will be listed on the Mailbox Settings page. An example of this page is shown in Figure 23.

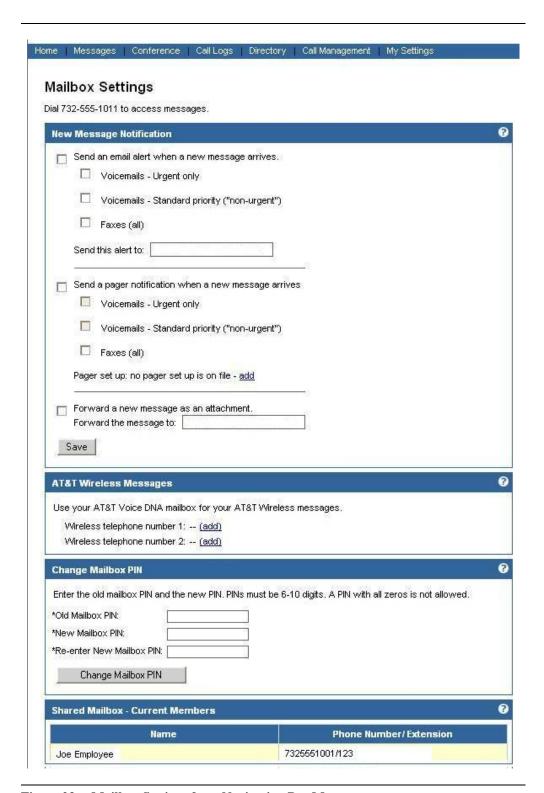


Figure 23. Mailbox Settings from Navigation Bar Menu

Changing Your Mailbox Settings

You can setup or change your Mailbox Settings using your phone or by using your AT&T Voice DNA Personal Web Site.

Some settings can only be set up or changed using your phone. Table 1 shows some examples of items you can do from your phone and items you can do from your Voice DNA Personal Web Site.

Table 1. Phone vs. Personal Web Site

Using Your Phone	Using Your Personal Web Site
Record your greetings.	Set up notification by email or pager.
Set preferences for listening to	• • • • • • • • • • • • • • • • • • • •
messages by phone.	Reset your PIN.
Toggle message notification on or off	
Create a distribution list.	
Reset your PIN.	
Add a wireless phone to your mailbox.	

To make changes to your Mailbox Settings using your AT&T Voice DNA Personal Web Site:

- 1. Click on the **My Settings** link on the navigation bar menu.
- Drag down to Mailbox Settings.
- 3. This gives you the Mailbox Settings screen. The page is broken up into 3 parts:
 - Set up notification by email or pager when you get a new message.
 - Add a wireless phone from AT&T to your AT&T Voice DNA mailbox.
 - Change your mailbox PIN.

■NOTE:

If you are a Shared Mailbox Owner, Shared Mailbox Members (whose messages are forwarded to your phone) are also listed on this screen.

New Message Notifications

Using the New Message Notification, you can receive an e-mail message or page each time a new message is received. New messages can be forwarded as email attachments to an e-mail address.

Send an e-mail Alert When a New Message Arrives

If you would like to get an e-mail when a message is received, select the **Send an e-mail alert when a new message arrives** checkbox and enter the desired e-mail address in the text entry box. By default, your e-mail address appears as the destination. You can change the default to any valid e-mail address. This e-mail address will receive the e-mail alert. You can also specify the type of message that triggers an e-mail alert. Choices include urgent voicemail messages, standard

priority voicemail messages, and faxes. Click on the **Save** button to save any changes.

■NOTE:

The TUI is the same as the Mailbox (or message) access number. You can see your message access number on the **Mailbox Settings** screen. The message access number is located under **Mailbox Settings** at the top of the page (see Figure 23).

Forward a New Message as an Attachment

If you would like the new messages to be forwarded as e-mail attachments, select the **Forward a new message as an attachment** checkbox and enter the desired e-mail address in the text entry box. By default, your e-mail address appears as the destination. You can change the default to any valid e-mail address. This e-mail address will receive the forwarded e-mail message. Click on the **Save** button to save any changes.

■NOTE:

The email address returns to the default registered address when the forward is un-flagged via the Personal Web Site, or disabled via the Telephone User Interface (TUI). The prior entered address (e.g., personal address) will not be retained.

Send a Pager Notification Each Time a Message Arrives

You can choose to be notified by pager when a new message arrives in your mailbox. When this feature is enabled and an incoming message matches your notification settings, a page will be sent to the designated pager.

≡>NOTE

Pager notification when messages are received will not work when an extended absence greeting is used and is set to not allow messages. To continue to receive pager notifications during your absence, be sure the extended absence greeting is set to allow messages to be recorded. For more information see **Greetings** on page **342**.

To set up your pager to inform you when a new message arrives in your mailbox, perform the following:

- Select the Send a pager notification when a new message arrives checkbox to be paged each time a new message arrives. You can also specify the type of message that triggers a pager notification. Choices include urgent voicemail messages, standard priority voicemail messages, and faxes.
- 2. If a pager has been set up, the pager name is displayed. If pager has *not* been set up, you must enter your pager information by following the instructions below under **Add a Pager**.

Add a Pager

1. To add the pager click the <u>add</u> link next to **Pager set up: no pager set up on file**. The **Pager Set Up** screen appears (Figure 24).

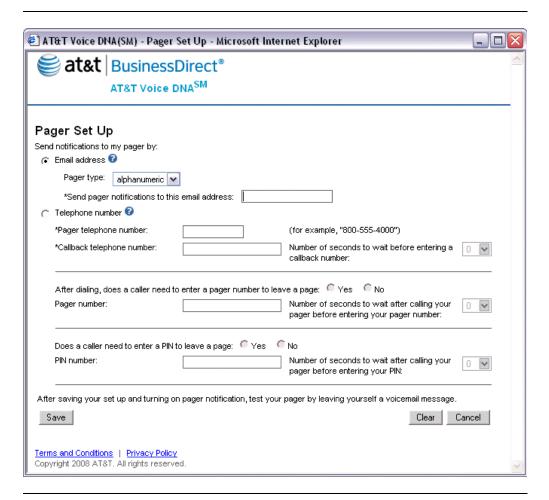


Figure 24. Pager Set Up screen

- If you want to send the page by email, click the Email address option button and then:
 - a. In the **Pager type** drop down menu, select the type of pager.
 - b. In the **Send pager notifications to this email address** field, enter the address where the email should be delivered.

When entering the email address for your pager, you may have the option of entering different domain names. These different domain names allow you to control which parts of the email are sent to the pager. For example, this can be used to specify that only the body of the email which contains the callback telephone number be sent to a numeric pager. You may also be able to have more control over what information is sent by using special options (switches) that are added to the Subject line of the email to provide specific information to the pager. Contact your paging service to determine what domain name or special options that you can use based on your paging service.

- 3. If the page needs to be a telephone number, click the **Telephone number** option button and then:
 - a. In the **Pager telephone number** field, enter the telephone number for the pager.
 - b. Callback telephone number This is the callback number that the subscriber wants to see on their pager when they are notified that a new message has arrived.
 - c. **Number of seconds to wait before entering a callback number —** Enter the number in seconds of the time to wait before entering a callback number.
 - d. After dialing does a call need to enter a pager number to leave a page
 If you select the Yes option button enter the following:
 - Enter the **Pager number** This is the pager number to dial after calling the paging provider.
 - Number of seconds to wait after calling your pager before entering your pager number — Enter the number in seconds of the time to wait after calling you pager before entering your pager number.
 - e. **Does your caller need to enter a PIN to leave a page** If you select the **Yes** option button, enter the following:
 - Enter the Pin number This is the pager PIN to dial after the pager number.
 - Number of seconds to wait after calling your pager before entering your PIN — Enter the number in seconds of the time to wait after calling you pager before entering your pin.
- 4. Click the **Save** button to save the changes.
- 5. Test your pager set up by calling yourself and leaving a message. If your settings are correct you should receive the page.

Change a Pager

If you need to change your pager information after it has been set up, perform the following.

- 1. Click on the My Settings link on the navigation bar menu.
- 2. Drag down to Mailbox Settings to view the Mailbox Settings screen.
- 3. Click on the change link in the New Message Notification section.
- 4. The **Pager Set Up** page (Figure 24) will be displayed with the pager's information. Make the desired changes to the information.
- 5. Click on the **Save** button to save the changes.

Disable Pager Notification

If you want to disable your pager notification, un-click the **Send a pager notification** when a new message arrives checkbox on the Mailbox Settings page (Figure 23) and click on the **Save** button. You can also toggle the setting On/Off from the TUI.

Add a Wireless Telephone to Your Mailbox

You can use your AT&T Voice DNA mailbox as the mailbox for up to two wireless telephones from AT&T.

■NOTE 1:

This feature is available only if you have the voicemail feature through AT&T Voice DNA. The wireless phone must be in the same local service area as the mailbox. Only wireless telephones from AT&T can be added to your AT&T Voice DNA mailbox. This service is not available in all areas. Your AT&T Voice DNA Administrator can contact AT&T Voice DNA Customer Care or your AT&T Account Representative to check on availability in your area.

■NOTE 2:

Please review and delete any existing wireless messages and greetings on your wireless phone before adding it to your mailbox. Existing messages and greetings are **deleted and not retrievable** upon adding the phone to your mailbox. See the **Greetings** section for more information on greetings.

To add a wireless telephone to your AT&T Voice DNA mailbox:

 On the Mailbox Setting page, in the AT&T Wireless Messages section, click the add link next to Wireless telephone number 1. A screen similar to Figure 25 is displayed.

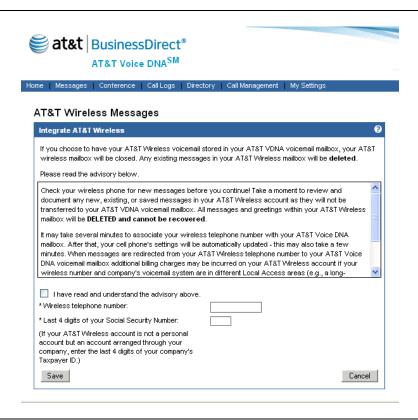


Figure 25. AT&T Wireless Integration Screen

- 2. Read the advisory on the AT&T Wireless Integration screen and check the box indicating I have read and understand the advisory above.
- 3. Enter the 10 digit telephone number of the wireless phone/device you want to integrate. This is a required field.
- Enter the last four digits of your Social Security number. This is a required field.

■NOTE 3:

The last four digits of the Social Security Number entered in Step #4 must match the digits associated with the wireless phone account being added.

5. Click **Save**. The system responds with a confirmation message similar to the one show in Figure 26.

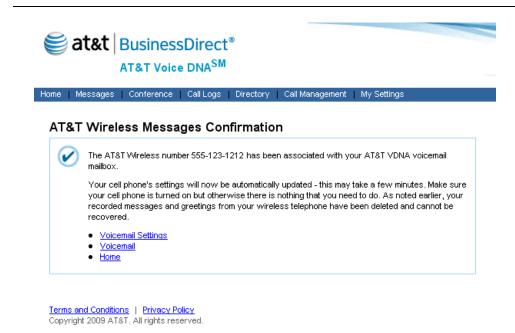


Figure 26. AT&T Wireless Integration Confirmation Screen

7. Your AT&T Voice DNA access number is now the mailbox access number for your wireless phone. Your access number will also be the call forwarding number for all busy/no answer calls to your wireless phone.

■NOTE 3:

It is recommended that you place a test call to your mailbox to ensure that set up was successful. If set up was not successful, repeat steps 0 through 7 or call Customer Care for assistance.

8. Once the account has been associated with your AT&T Voice DNA account, you will need to dial into the messaging system using the wireless phone to set up your PIN, record your name, and select a greeting to use. You can select the same greeting associated with your AT&T Voice DNA account or record another separate one to use. Once setup has been completed, the wireless phone message count will be in sync with the new message count on your AT&T Voice DNA phone. If you have chosen to use the Message Waiting Indicator (MWI), all new messages left for either phone will result in a MWI light on the VDNA phone and a Message Indicator on the associated wireless phone.

Removing a Wireless Telephone From Your Mailbox

To remove a wireless telephone to your AT&T Voice DNA mailbox:

1. On the **Mailbox Setting** page, in the **AT&T Wireless Messages** section, , click the **remove** link next to the **Wireless telephone number** you want to remove. See Figure 27

Neu		
ne-s	v Message Notification	
	Send an email alert when a new message arrives.	
	- Yolcomails - Crigoric orny	
	☐ Voicemails - Standard priority ("non-urgent")	
	Faxes (all)	
	Send this alert to:	
	Send a pager notification when a new message arr	rives
	Voicemails - Urgent only	- 124K
	☐ Voicemails - Standard priority ("non-urgent")	
	Faxes (all)	
	Pager set up: no pager set up is on file - add	
9	Forward the message to:	
AT8	LT Wireless Messages	
Use	your AT&T Voice DNA mailbox for your AT&T Wirele	ess messages.
	Mireless telephone number 1: 816-555-1313 (remove	ਸ਼੍ਰੇ ਬ
್ಗ	Vireless telephone number 2: (add)	
Cha	nge Mailbox PIN	
-	er the old mailbox PIN and the new PIN. PINs must be t	6-10 digits. A PIN with all zeros is not allowed.
ALC: NO.		
Ente *Old	Mailbox PIN:	
Ente *Old *Nev	Mailbox PIN: w Mailbox PIN: enter New Mailbox PIN:	

Figure 27. Mailbox Settings from Navigation Bar Menu with Wireless Integration

The system responds with a confirmation message similar to the one show in Figure 28.

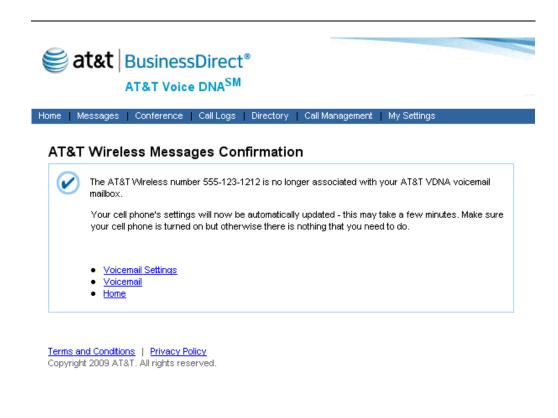


Figure 28. AT&T Wireless Integration Removal Confirmation Screen

Change Your Mailbox PIN

To change your Mailbox PIN:

- On the Mailbox Setting page, in the Change Mailbox PIN section, enter your current Mailbox PIN in the Old Mailbox PIN field.
- Enter your new Mailbox PIN in the New Mailbox PIN field. Your PIN must be a 6 to 10 numeric digit. A PIN with all zeros is not allowed.
- 3. Enter your new Mailbox PIN in the **Re-enter New Mailbox PIN** field. The system requires you to enter the PIN twice to ensure that you did not make a mistake when entering the PIN.
- 4. Click the Change Mailbox PIN button to save your changes.

≡>NOTE:

If you forget your Mailbox password, contact your company administrator who can reset your password.

Message Problems

If you are having problems with messages playing in Internet Explorer, you must have the browser set to play media in the page. To do this, select the Media icon from the tool bar (Internet Explorer 6.028 or earlier). This opens the windows media player on the left side of the browser. On the Media player, select **Media Options**Settings Play web media in the bar. After this is set up, the messages should play.

To listen to a message, your PC must have Windows Media Player along with speakers or a headset. If you don't have Windows Media Player, you will need to save the file to your PC and then open it with the media player you have on your PC.

Voicemail for Multiple Line Appearance (MLA)

When a single telephone number appears on multiple phones (i.e., MLA) and voicemail is also supported, the Message Waiting Indicator and retrievals of messages may be different between the two phones. In most cases, when a message has been stored, the Message Waiting Indicator will appear on both phones.

However, if one telephone has voicemail (e.g., the manager) and a second does not have voicemail (e.g., a secretary), even if MWI is showing on the second phone, pressing the voicemail button does not permit the secretary to retrieve the message (they must dial the mailbox access number).

Shared Mailbox

AT&T Voice DNA Users with Voicemail can have a common (shared) mailbox that provides messaging support for the primary Voice DNA telephone number and up to fifteen additional phone numbers. The Administrator enables this feature by setting a particular voicemail mailbox as the **Final Destination** for a hunt group and ACD, and, by setting other AT&T Voice DNA phones and external phones as Mailbox Members for a particular Shared Mailbox. The User can retrieve these voicemail and fax messages. If your voicemail mailbox has been set to receive voicemails, an information box located on the left side of your Personal Web Site states that this is a shared voicemail mailbox (see Figure 29).

■NOTE:

The Shared Mailbox *Owner* is the User assigned to the primary AT&T Voice DNA phone number. Shared Mailbox *Members* are those Users, Hunt Groups, Call Distribution queues, or fax numbers, whose calls are routed to this common (shared) mailbox. Members can include other Voice DNA numbers that do not have voice mail assigned in the COS, as well as external numbers that are assigned to the shared mailbox as a final destination.

if they have the locate feature available, they must remember to set the last leg to the VM number, if they to want to retrieve messages later

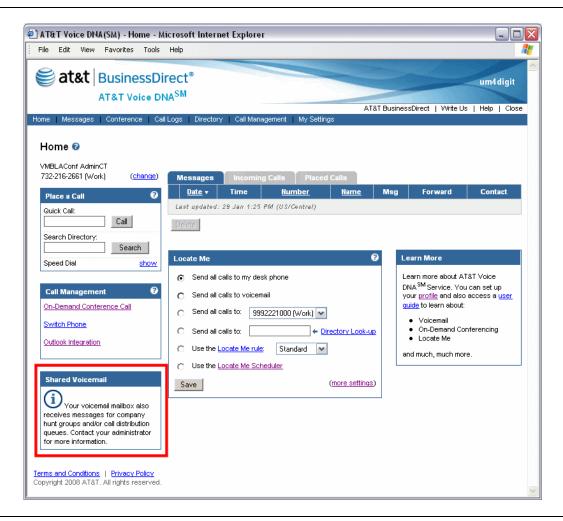


Figure 29. Example of a Shared Mailbox - Owner

The additional fifteen phone numbers connected to the shared mailbox can be other AT&T Voice DNA Users that do not have a mailbox or non-AT&T Voice DNA Users. For example: a Shared Mailbox can be configured to accept unanswered calls from the Mailbox Owner's primary AT&T Voice DNA telephone number, from a second non-AT&T Voice DNA telephone number associated with the User's home office, and from a third fax number.

IMPORTANT:

Call Forwarding must be enabled for each additional telephone number to send busy or unanswered calls to the Shared Mailbox.

The Shared Mailbox will accept voice messages (from busy or unanswered calls) and fax messages for all the telephone numbers associated with the mailbox. There is no indication of the telephone number that was called; however, icons differentiate voice and fax messages.

Shared Mailbox Message Retrieval

All AT&T Voice DNA phones associated with a Shared Mailbox have Message Waiting

Indication (MWI) and can access messages through the Telephone User Interface (TUI).

The Shared Mailbox Owner can access messages through the Messages tab on their User Personal Website. The Messages tab displays all messages in the Shared Mailbox. The Shared Mailbox Owner also has additional messaging notification options, such as email and paging.

Shared Mailbox Members cannot access the messages through their AT&T Voice DNA Personal Websites and must coordinate with the Shared Mailbox Owner if they want to retrieve messages using a method other than the TUI. A sample Shared Mailbox Member's mailbox is shown in Figure 30.

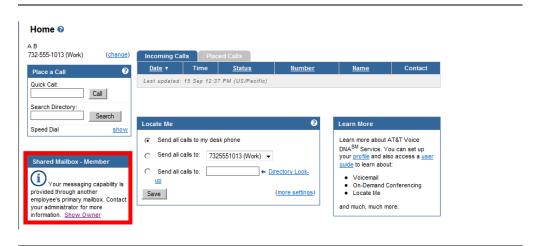


Figure 30. Shared Mailbox - Member

Shared Mailbox Member - Locate Me Settings

To ensure calls are forwarded to the Shared Mailbox telephone number, Shared Mailbox Members must set the **If busy, forward to** and

If no-answer telephone numbers to the Shared Mailbox access number on the Locate Me page.

■NOTE:

If you do not have access to the end user Portal, use the TUI to set the **if busy** and **if no answer** options. Refer to Table 13 on page 319 for further information.

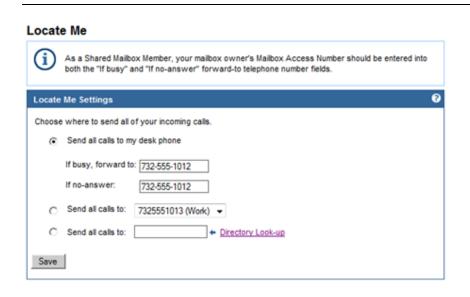


Figure 31. Shared Mailbox Member - Locate Me

Making a Conference Call

You can make conference calls with up to ten participants (including yourself). It is simply a matter of entering their numbers and starting the call by pressing the **Start Call** button.

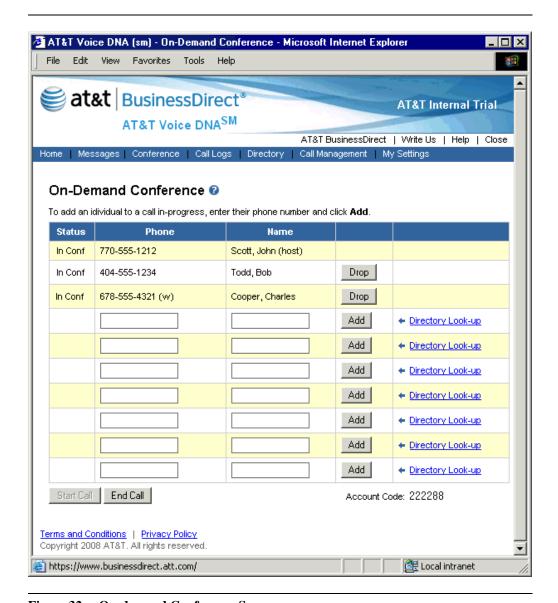


Figure 32. On-demand Conference Screen

■NOTE 1:

If you change the Phone Number of the host (this uses the Remote Click to Call feature), the new phone number must be provisioned in the profile of the host. When invoking Remote Click to Call (or Click to Conference), always check the originating TN to ensure that it reflects the contact number for the device at which you will initiate the call.

■NOTE 2:

Click to Conference to call 911 does not work correctly and should not be used.

■NOTE 3:

Click to Conference to an Auto Attendant CTN (Primary or Secondary) is not supported. That is, an AT&T Voice DNA user cannot use the Click to Conference feature to add an Auto Attendant CTN as a conference participant.

■>NOTE 4:

You must have a provisioned telephone to use this feature.

■>NOTE 5:

You can conference up to two callers plus yourself, by using your telephone (provided that you have the ability to perform a conference call). For more information on how perform a conference call from your telephone, see Table 6 or Table 7.

To begin a conference call:

1. Go to the **Conference** page by clicking **Conference** on the navigation bar menu.

■NOTE:

Status must be Idle for the initiator in order to initiate the conference.

2. Enter the phone numbers (mandatory) and names (optional) of the participants. You can use the **Directory Look-up** link if these participants are included in the Directory. To use the **Directory Look-up**, click on the **Directory Look-up** link. The system responds with Advanced Search window.



Figure 33. Advanced Search Window

Enter in the search criteria and click on the **Search** button. Note that, for each search string, you can specify **contains**, **exact**, **starts with**, or **ends with** (wildcard characters are not supported). The system responds with the results that match the search criteria. If you want to add a contact to the Conference Call from the results, select the option button to the left of their name and click on the **Select** button. Once you select the option button, this contact is added to the Conference Call page.

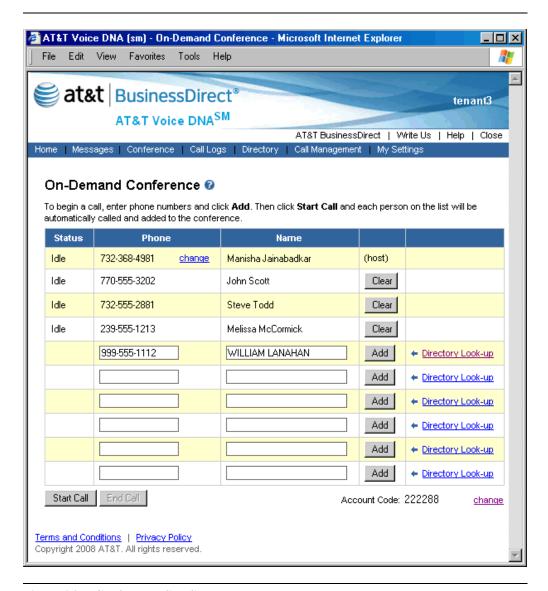
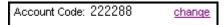


Figure 34. Conference Call Setup

- 3. Click on the Add button to add the contact to the conference list. If you manually entered the contact, click the Add button after entering the information to add each number and name to the conference list. If you have added a contact that you want to drop before the call is initiated, click on the Clear button located to the right of the contact's name.
- 4. Account Codes can be: not used, Optional, or Mandatory.
 - If Account Codes are not used (not enabled in your Class of Service template), you will not see the Account Code field located on the bottomright-hand side of the On-Demand Conference Screen (Figure 34).
 - If Account Codes are Optional, you will see the Account Code that has been selected and the <u>change</u> link to the right of the Account Code on the bottomright-hand side of the On-Demand Conference Screen (Figure 34).



This Account Code displayed on the screen will be used for this conference. This field can be blank since the Account Code is optional. You can change the Account Code by clicking on the **change** link. Clicking on the **change** link displays a screen similar to Figure 35.



Figure 35. Change Optional Account Code for Conference Call

This screen allows you the following choices:

 Do not use an account code. Selecting this option indicates that no Account Code should be associated with this conference. Selecting this option causes the Account Code field on the Click to Conference Screen to be blank.

Associate the calls for this conference with the selected account code. Selecting this option indicates that the selected Account Code should be associated with this conference. The selected Account Code will be displayed on the On-Demand Conference Screen. This Account Code will be associated with all of your conferences until you change the Account Code. To change the Account Code, select an Account Code from the drop down menu. Only Account Codes that exist in the pull down menu can be used. These codes must have been entered into the pull down menu by your Administrator.

After making any changes, click on the **Save** button to save the setting.

If Account Codes are Mandatory, an Account Code is required to be used for the conference. The Account Code that has been selected will be displayed and the <u>choose an account code</u> link will be located to the right of the Account Code.

Account Code: 222288 choose an account code

The displayed Account Code will be associated with this conference. This Account Code will be associated with all of your conferences until you change the Account Code. You can change the Account Code by clicking on the **choose an account code** link. Clicking on the **choose an account code** link displays a screen similar to Figure 36. Only Account Codes that exist in the pull down menu can be used. These codes must have been entered into the pull down menu by your Administrator.



Figure 36. Change Mandatory Account Code for Conference Call

To change the selected Account Code, select another Account Code that is displayed in the drop down menu and click on the **Save** button to save the setting.

5. When ready to start the conference call, click the Start Call button. Once the Start Call button is clicked, your (originator) phone rings first. When your phone is answered, then each person on the list is called simultaneously and if they answer, they are added to the conference. The caller and others on the call will not hear any ringing as each party is added. Anyone who does not answer within 16 seconds is dropped from the call (conference status for that leg will show Failed). Figure 37 shows a Conference Call in progress.

■NOTE:

If you receive a busy signal or reach a user's Voicemail, you can use the **Drop** button to drop the user from the call.

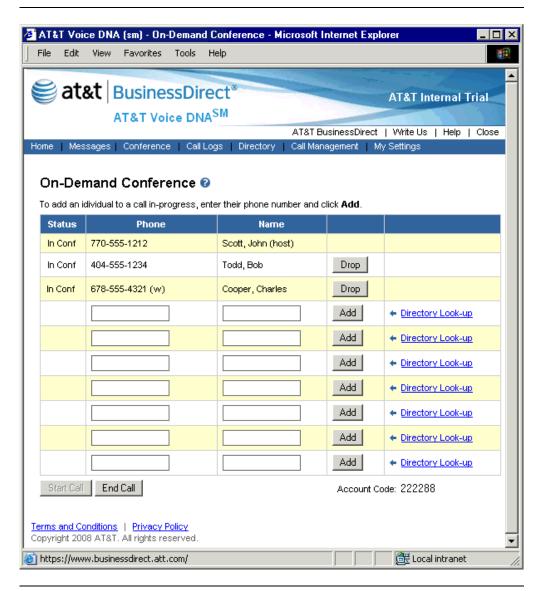


Figure 37. On-demand Conference Call in Progress Screen

■NOTE:

On a Conference Call, if a Host goes on Hold, then each party controls (toggle on/off) music independently of the other by depressing the # key. When a participant goes on Hold, then only the Host can toggle MOH on and off.

Adding a Caller to a Conference Call in Progress

- 1. Enter that person's phone number and/or name.
- 2. Click the Add button.

Drop a Caller From a Conference Call in Progress

To drop an attendee from an in-progress conference call, click on the **Drop** button next to the attendee's name.

■NOTE:

Dropping an attendee to the call does not end the call. You must click on the **End Call** button to end the call.

Ending a Conference Call

1. To end a conference call, click the **End Call** button.

■NOTE:

You should end the conference call by clicking on the **End Call** button before exiting the Personal Web Site (or changing screens). Exiting the Personal Web Site does not clear the conference state. For example, if your PC lost power the existing conference call would not be killed. If the host hangs up their phone, the conference call ends and the other attendees are disconnected.

Using Speed Dial

The **Speed Dial** item under the My Settings menu lets you create and delete personal Speed Dial numbers, and displays those assigned by your company administrator. You can use your Speed Dials for one-click dialing to your selected numbers.



Figure 38. Speed Dial from the Navigation Bar Menu

Speed Dialing

On the Personal Web Site, in the **Place a Call** box, locate the Speed Dial **show** link.



Figure 39. Speed Dial Link

- 1. Click **show** to display your Speed Dial list.
- 2. Click a Speed Dial number to dial that number.
- 3. Click a Speed Dial name to edit the Contact Information.

■NOTE:

The Speed Dial and Optional Account Codes feature will not work together.

Viewing your Speed Dial Lists

Use the drop down list at the top of the Speed Dial page to select the list you want to view. You can select Personal (1-10), Personal (11-20) or Company Assigned.

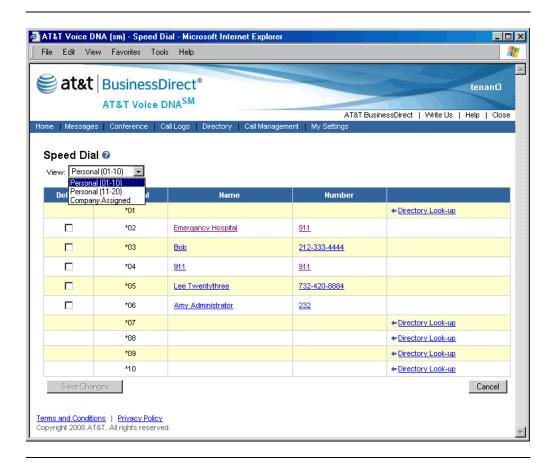


Figure 40. Speed Dial Drop Down List

Creating a Speed Dial Entry

The Speed Dial item under the **My Settings** menu lets you create and delete personal Speed Dial numbers, and displays those assigned by your company.

You can create a Speed Dial entry as follows:

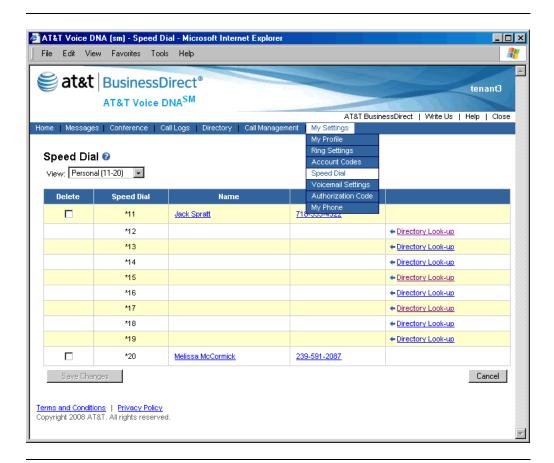


Figure 41. Speed Dial Entries

- 1. Go to the My Settings item on the navigation bar menu at the top of the page.
- 2. Select **Speed Dial** from the menu.
- 3. Use the **Directory Look-up** link to search the Directory for this person. Click on the option button next to the desired name and then click on the **Select** button to transfer the data to the Speed Dial screen. You cannot enter the contact's name or number directly into the Speed Dial screen. Do not enter 911 as a Speed Dial Code.
- **4.** Once the information is entered, click the **Save Changes** button to add this entry to your Speed Dial. Only 20 personal Speed Dial entries can be entered.

Deleting a Speed Dial Entry

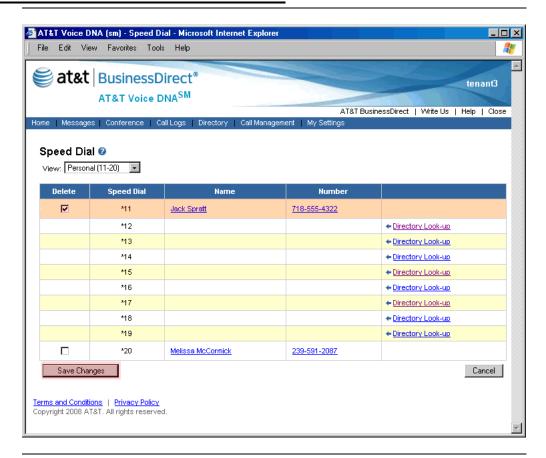


Figure 42. Deleting a Speed Dial Entry

- 1. Go to the **Speed Dial** page.
- 2. Click the checkbox in the **Delete** column next to the entry you wish to delete.
- 3. Click the **Save Changes** button to remove this Speed Dial entry.

Using the Directory

The **Directory** gives you an on-line phonebook, which includes your company directory plus your own entries and groups.

The Directory allows you to quickly locate the phone number of a contact and initiate a call automatically from the Directory page. When you use the Click to Call feature, when the call is started, your (originator) phone rings first. When your phone is answered, then their telephone rings, and the speaker, if available, becomes active as the call is connected.

The Directory is also the basis for creating groups of callers and a personal directory. For example, you might want to include a contact in your Family & Friends Groups so that you could apply special call treatment for this group of callers. The Locate Me feature uses the contact's Group to apply call treatments. For example, you could create a Locate Me rule that would forward all callers which are contained as contacts in your Family & Friends Group to be forwarded to your Cell phone, and if no answer then forward to your Home phone, and if still no answer then to your Voicemail.

All of the contact information is stored on the secure server by AT&T. As a result, you can do the following:

- Access the Contacts list from any Web-connected location so you can use the same list of contacts at home and at work.
- Have a company-wide Contacts list created by the company administrator as well as a personal Contacts list.
- Finally, you can export your directory or company contacts to a ".csv" file.

Viewing the Directory

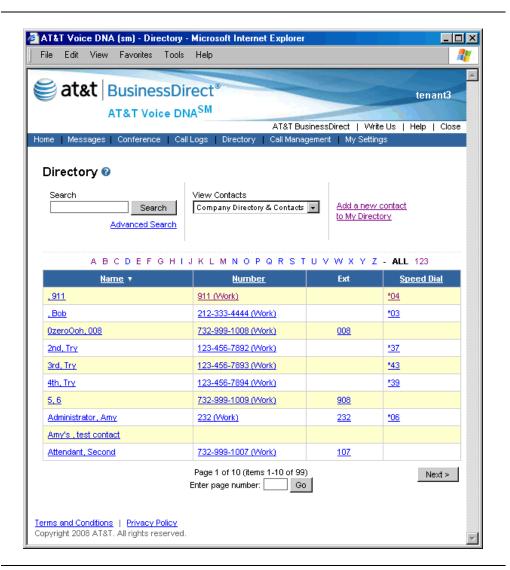


Figure 43. The Directory

To view the Directory page:

- 1. Select **Directory** from the navigation bar menu of the Personal Web Site.
- Select View Directory to display the Directory page, where you can View (specific groups), Search, Add New Contacts, select listings alphabetically, and page through the Directory listings. Groups are defined by the system. You can have up to 200 personal contacts included in the Directory.

Moving Through the Directory

To display different parts of a Directory listing, either:

Use the alphabetical jump links.



Figure 44. Alphabetical Links

■NOTE:

If there is no contact associated with a particular letter, the Directory displays the message **No Results**.

— Alternatively, use the **Previous** and **Next** buttons below the Directory table.

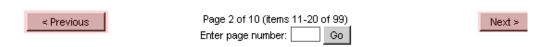


Figure 45. Previous & Next Buttons

 Alternatively, enter a page number and click the Go button below the Directory table.

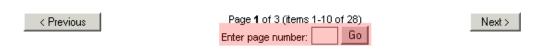


Figure 46. Page Number Field

Jumping Alphabetically

To jump to a letter of the alphabet in a Directory:

1. Click the A-Z jump links at the top of the Directory table.



Figure 47. Jump Links

Searching the Directory

The **Directory** page features **Search** and **Advanced Search** functions to help you quickly find someone in your **Directory** or the **Company Directory**. You can search on part or all of a contact's name, phone number or nickname.

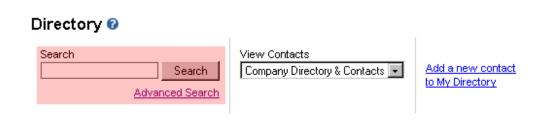


Figure 48. Directory Search All Contacts

To Search:

- 1. Enter all or a portion of the person's name in the **Search** field.
- 2. Click the Search button.
- 3. You can click the Advanced Search link to perform partial string searches on first name, last name, company, or phone number, or to limit your search to a specific group (VIPs, Co-workers, Family & Friends, or your Refuse Group). Note that, for each search string, you can specify contains, exact, starts with, or ends with (wildcard characters are not supported).



Figure 49. Directory Advanced Search Fields

When using the **Directory Search** → **Advanced Search** → searching on the **Phone Number** will only return Work TNs that match your search criteria. Other TNs that may be populated in the directory for users (e.g., cell, home, etc.) will not be found in this search. This is true even if an alternate TN, such as Cell TN is designated by the employee as their Primary TN.

View Contacts

The **View Contacts** drop down menu on the **Directory** page allows you to display all of the contacts associated with a particular group.



Figure 50. View Contacts

To View Contacts in a group:

- 1. Select a group from the View Contacts pull-down menu.
- Once you select a menu item, the system displays all contacts associated with the group. For example, if you selected My Directory, all contacts associated with the My Directory group will be displayed.

Making Calls to Directory Numbers

To call someone listed in the Directory:

- 1. Click the telephone number listed.
- 2. This gives you the **Phone Call** screen, from which you can click the **Call** button to call that number. Figure 51 shows an example of the Phone Call screen.



Figure 51. Phone Call Screen

Adding a New Contact

To add a new name/number to one of the Directories:

1. Click the Add a New Contact to My Directory link.

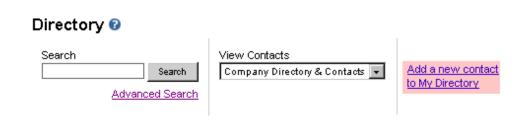


Figure 52. Add a New Contact Link

- 2. That gives you the Add Contact screen.
- 3. Enter the name and group, phone numbers, fax number, speed dial numbers, email addresses, and other contact information (Figure 53) by entering (typing) or selecting pre-defined values from drop down menus. An entry into an Other field will override the associated drop down menu. You must enter a first name and at least one phone number (other than a fax), and select a Primary Phone designation. A fax number cannot serve as a Primary Phone.

■NOTE 1:

If you do not have the Locate Me feature, you have more limited contact entry ability. Since you do not have the Locate Me Rules, you do not need or have the Groups (vip, refuse, etc.). You also can only add one phone number per contact.

■NOTE 2:`

Personal contacts that are assigned to the groups **Co-Workers** or **Family** cannot be used as directory search criteria via Outlook Integration. If you want to view these groups in Outlook Integration, select **Phone Directory** → **Advanced Find** → **Personal Contacts** → **Find**. The entire contents of your personal directory are displayed and can then be sorted by group.

■NOTE 3:

When editing a contact that is in the company directory, you can only make 2 changes:

- add the contact to a call group (for locate me treatments)
- assign a personal speed dial to the contact.

You cannot edit any of the corporate directory information for the contact. Only the administrator can add/edit corporate directory information.

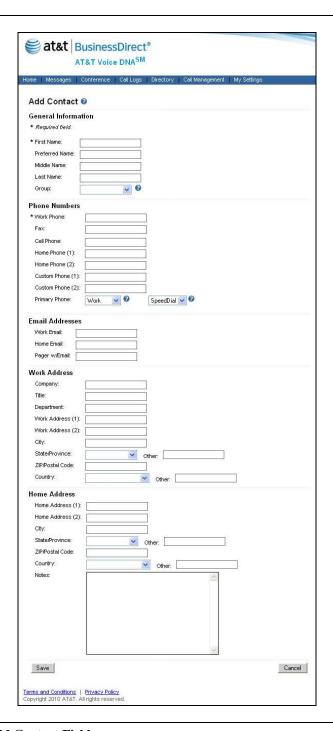


Figure 53. Add Contact Fields

4. Once the contact information has been entered, click on the **Save** button to save the information.

Editing Directory Entries

To edit the contact information for someone listed in a Directory:

1. Click their name.



Figure 54. Name Column

- 2. This gives you the **Edit Contact** screen for that person.
- **3.** Edit the name and group, phone and fax numbers, speed dial numbers, email addresses, or other contact information.
- 4. Click the **Save** button to save your changes.

Editing Directory Contact Information

You can edit the Contact information for a contact by clicking on the contact's name in the Directory or by clicking the Contact symbol for that entry line from one of the logs.

- Click the name of the contacting the Directory or by clicking the Contact symbol
 for that entry line from one of the logs.
- 2. This brings you the **Edit Contact** page for this person. If the contact is in the Company Directory, most of the fields are display only.
- **3.** Add or edit the general information, phone and fax numbers, email addresses, work address or home address for this person.

■NOTE:

When entering phone numbers:

- Enter the complete phone number, including the area code.
- Do not include a prefix, such as the number 1 for out-of-area calls or the number that is used to obtain an outside line (usually 9).
- If an international number is entered, the complete international number must be entered, including international access code and country code.
- A valid local extension, such as a 4 or 5-digit number can be entered.
 However, the extension number cannot start with the number that is used to obtain an outside line (usually 9).
- The number to obtain an outside line is required in order for Click to Call to work with an international number contained in the Directory.
- 4. Click the **Save** button to store those Directory changes.

Exporting a Directory and Contacts

You can export your Directory and Contact list to a ".csv" file, to open with other applications.

■NOTE:

The file can include a maximum of 2,000 records. If your list has greater than 2,000 records, decrease the file size by choosing **Company Directory only** or **Company Contacts only**. You can also export the file in sections by choosing **Selected Contacts**, typing a *start* letter in the **Last Name** field, and choosing **starts with** from the drop-down menu.

To export your Directory or Contacts:

- 1. Select **Directory** from the navigation bar menu of the Personal Web Site.
- 2. Select Export Directory and Contacts to view the Export Directory and Company Contacts screen.

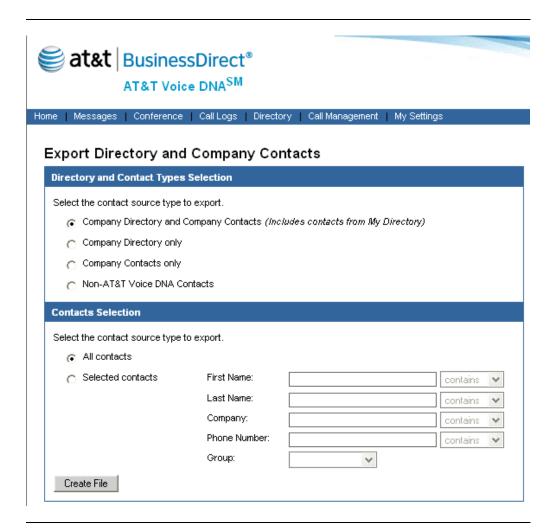


Figure 55. Export Directory and Contacts screen

- 3. Choose the type of contacts to export under **Directory and Contact Types Selection**.
- 4. Specify additional search criteria (if desired), by clicking Selected Contacts under Contacts Selection. Enter text in any of the fields, including First Name, Last Name, Company, Phone Number, and Group. To the right of each field, you can select additional search criteria, such as: contains, exact, starts with, and ends with.
- 5. Click the **Create File** button. The **Export Directory and Company Contacts Confirmation** screen appears.

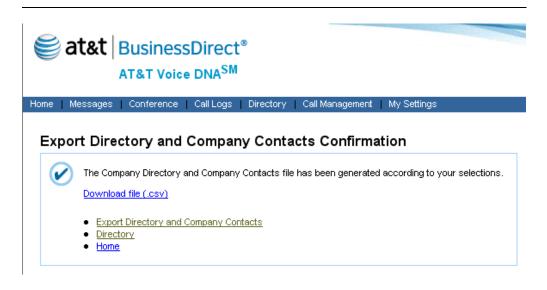


Figure 56. Export Directory and Company Contacts Confirmation screen

6. Click **Download file (.csv)**. You will be prompted to open or save the file.

Managing Calls



Figure 57. Call Management from the Navigation Bar Menu

You will find the Call Management features under **Call Management** on the navigation bar menu. Click an entry under **Call Management** to use that function. For example, pull down to **Locate Me** to forward your calls; or pull down to **Switch Phone** (Mid-Call Move) to transfer a call to another telephone number.

Using Locate Me

Call forwarding can be set dynamically using the option buttons in the **Locate Me** frame of the Personal Web Site or using the phone display. For example, you might want to forward all calls to your desk phone, or have all calls go directly to a mailbox if you do not want to be disturbed. Additionally, you could choose to have all calls go directly to a mobile telephone for a day when you are planning to be out of the office. If your unanswered calls are forwarded to a Shared Mailbox, you need to enter the proper Shared Mailbox access number in the **if busy** and **if no-answer** fields in the **Locate Me** frame.

≡>NOTE:

If you do not have access to the end user Portal, use the TUI to set the **if busy** and **if no answer** options. Refer to Table 13 on page 319 for further information.

The **Locate Me** feature gives you a variety of options for routing incoming calls to your desk phone, to your Voicemail, or to other telephone numbers. It also gives you a Locate Me Rule feature that allows you to route calls based on who they are coming from, and to route them along a sequential path to locate you at specified telephone numbers. You must be subscribed to this feature in order for it to be active. Note that the checkboxes under the option buttons are ignored by the system unless the option button is selected.

■NOTE:

The **Locate Me** frame on the **Home** page does not contain all of the options described below. To see all of the options from the **Home** page, click on the **more settings** link located in the **Locate Me** frame on the **Home** page.

Your options under Locate Me include:

- Send all calls to my desk phone
 - If busy, forward to this telephone number.
 - If no-answer, send to this telephone number.
- Send all calls to voicemail.
- Send all calls to a number selected from a list. This list is a drop down menu of the phone numbers that you have defined for yourself in My Profile.
 - If this number is busy or not answered, then send the call to voicemail.
- Send all calls to a new number. If you set this option from the Call Management
 → Locate Me menu item, you can use the Directory Link to look up and enter the desired number.
 - If this number is busy or not answered, then send the call to voicemail.
- Apply a Locate Me Rule based on where you are (Standard, AtLunch, GoneHome).
- Apply the Locate Me Scheduler which allows you to schedule how your calls will be forwarded for the week (Sunday through Saturday).

■NOTE 1:

The selected option button indicates your current setting. To select a different setting, simply click the associated option button.

■NOTE 2:

If Call Distribution or Hunt Group takes control of the call and if added as a destination in a Locate Me rule, they will override the remaining Locate Me steps (including bypassing the Locate Me rule final handling).

Setting Up Locate Me

To use the Locate Me feature:

1. Click the **Locate Me** link under Call Management.

■NOTE:

The **Locate Me** frame is also located on the Personal Web Site; however, it does not show the **My Profile Phone Numbers** frame.

2. This gives you the **Locate Me** page, which displays the **Locate Me Settings** and your **Profile Phone Numbers**.

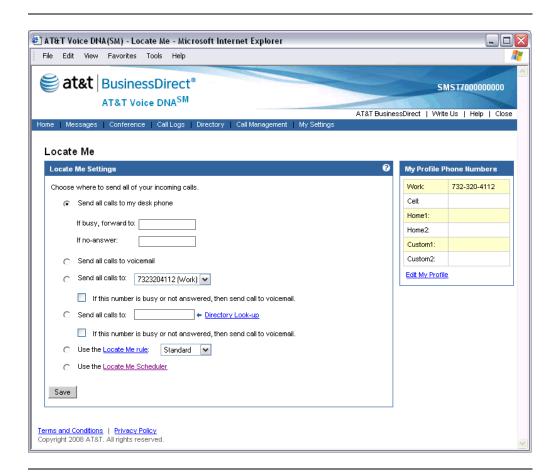


Figure 58. Locate Me Settings

Click an option button under Locate Me Settings to apply one of the listed options. Then select the desired checkbox or enter the desired choice into the entry box.

■NOTE:

If you are a Shared Mailbox Member, be sure to enter the Shared Mailbox access number in the **If busy, forward to** and **If no-answer** fields. Refer to **Shared Mailbox Member - Locate Me Settings** on page 45 for more information.

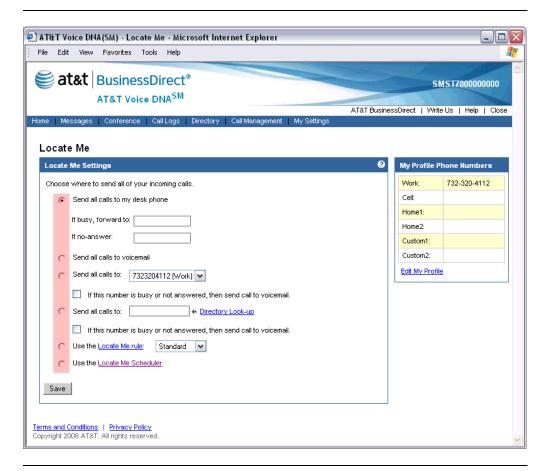


Figure 59. Locate Me Option Buttons

4. Then click the Save button to put that setting into effect.

■NOTE 1:

If the call forwarding setting is changed to send calls to voice mail or to be forwarded to another number, you must remember to change the call forwarding setting back to normal. Otherwise, calls continue to be forwarded to the specified location until the setting is changed.

■>NOTE 2:

If a Call Forward setup from the deskphone is forwarded back to the same phone, it will not follow the Locate Me rules for desk phone. You should not use *72 on your own Telephone Number if you want to disable forwarding. If you want to disable forwarding, do not forward to your own Telephone Number, enter *73.

■NOTE 3:

CFFV (Call Forward Fixed To Voicemail) and VM (Voicemail) must be enabled in your Class of Service template for you to have the option **Send all calls to voicemail** in Locate Me. Contact your Administrator to check if this feature is available for you.

■NOTE 4:

If you enter a telephone number in the second Send **All Calls to** field (located to the left of the **Directory Look-up** link) that is a number defined in your profile (e.g., your work number – 7705552121) and **Save** the information, Voice DNA recognizes the phone number and sets the option button in the first **Send All Calls to** field along with number [e.g., 7705552121 (work)] that was defined in your profile.

■NOTE 5:

If you want calls to ring your work phone number, it is recommended that when setting up Locate Me, you select **Send all calls to my desk phone** rather than **Send all calls to** your work phone number. This is recommended even though your work phone may be configured in your profile as the same as your desk phone. This ensures that you will continue to receive calls when your phone is moved to a new location and is 911 restricted, and that your calls will go to voicemail if your phone is not registered.

Using the Locate Me Rules

The Locate Me Rules page shows a table of the rule paths you have in effect for the following groups: VIPs, Co-workers, Family & Friends, All Others, and Refuse.

For example, if a contact calls and you have the rule active as shown in Figure 60 and that contact is part of your **Family & Friends Path** Group, the options would be:

- First option would be to forward to 7705551212.
- Second option would be to forward to 6785551111.
- Third option would be to forward to 4045552222.
- Final option would be to forward to your Voicemail.
- Phones would ring in order (one at a time).
- If not found, Work Email is notified.



Figure 60. Locate Me Rules

At the top of this table are buttons you can use to **Edit a Rule**, **Delete a Rule**, and **Add a New Rule**.



Figure 61. Edit, Delete, and Add New Buttons

When you add or edit a rule, the screen gives you fields for entering a phone number path for each of the groups — VIPs, Co-workers, Family & Friends, All Others, and Refuse.

■NOTE 1:

If a caller is not defined as being in a group (VIPs, Co-workers, Family & Friends, or Refuse), they are in the group All Others.

■NOTE 2:

You cannot delete the Standard, At Lunch, or Gone Home rules.

■NOTE 3:

There is a maximum of 10 locate me rules for a user. Once you reach the 10 rule limit the **Add New Rule** button will be grayed out (inactive).

ocate Me Rule: null			
Edit your Ring Lengths		Edit Ring	Lengths
IPs Path			Clicking on a link
Call 1st:	770-555-1212	← Work Cell Home 1 Home 2	populates the value
Call 2nd:	404-555-2637	← Work Cell Home 1 Home 2	into the field
Call 3rd:	678-555-1111	← Work Cell Home 1 Home 2	
Call Final:	Voicemail		– Final destination, if not foun
Ring my phones:		once 🗲	— Order to ring my phones
If not found, notify:	Notify at home email	—	🗕 If I am not found, what to do
If work phone busy, call:		← Cell Home 1 Home 2	
o-workers Path			
Call 1st:	770-555-1212	← Work Cell Home 1 Home 2	
Call 2nd:		← Work Cell Home 1 Home 2	
Call 3rd:		<u>Work Cell Home 1 Home 2</u>	
Call Final:	Voicemail		
Ring my phones:	⊙ In order ⊜ All-at	-once	
If not found, notify:	Notify at home email		
If work phone busy, call:		← Cell Home 1 Home 2	
amily & Friends P	ath		
Call 1st:	770-555-1212	← Work Cell Home 1 Home 2	
Call 2nd:			
Call 3rd:		₩Ork Cell Home 1 Home 2	
Call Final:	Voicemail		
Ring my phones:		-once	
If not found, notify:	Notify at home email		
If work phone busy, call:		← Cell Home 1 Home 2	
All Others Path			
Call 1st:	770-555-1212	₩Ork Cell Home 1 Home 2	
Call 2nd:	678-555-1111	← Work Cell Home 1 Home 2	
Call 3rd:	404-555-2637		
Call Final:	Voicemail		
Ring my phones:	In order C All-at-	-once	
If not found, notify:	Notify at home email	Ī	
If work phone busy, call:		← Cell Home 1 Home 2	
Refuse			
Final Only: Busy Signal	• ←	How to treat a Refuse	ed Call

Figure 62. Edit Rules Page

For each group, you can specify whether to ring your forwarding numbers in order (In Order), or all at once (All-at-Once). In addition, you can specify what to do if you are not found at one of these numbers. For example, you can receive a notification of the call at one of two email addresses (work and home) or at a pager number.

The Ring my phones option behaves as follows:

- If the All-at-Once option button is selected, incoming calls that match the selected call treatment are simultaneously forwarded to all numbers configured in the call treatment. If you answer the call at one of the numbers, the other numbers stop ringing. If you do not answer the call at any of the numbers, the call is forwarded to the final destination. In order for this feature to be available, the Simultaneous Ringing must be enabled in your Class of Service template. Contact your Administrator if you want to check if this feature is available for you.
- If the In order option button is selected, incoming calls that match the selected call treatment are forwarded sequentially. The number entered in the Call 1st field rings first. If you do not pick up the call at this number, the call is forwarded to the number entered in the Call 2nd field. If no answer, then to the number entered in the Call 3rd field. If no answer, then to the final destination as specified in the If not found, notify field. The amount of time that a phone will ring before a no answer is determined is set in the My Settings → Ring Settings menu.

■NOTE:

When configuring a Locate Me rule with Ring in Order, if any of the phones in your Locate Me rule is picked up by a voicemail system or answering machine, is declined or if the line is busy, the call is considered to be answered, and no other phone numbers in your Locate Me rule will be called. The calling party's voicemail message will not be left in your voicemail box but in the voicemail box or on the answering machine associated with the phone number in your Locate Me rule that picked up. To avoid this problem, all phones specified in your Locate Me rule should be configured with an adequate number of rings before going to voicemail. This ensures that all phones in your Locate Me rule are rung, and that the call is forwarded to your voicemail box if none of the phones answer.

You can also specify where a call should be forwarded if your work line is busy when a call from this group comes in—you can click a link to populate this field with your work number, cell number, or one of two home numbers.

Finally, for calls that do not find you along your Locate Me path, go to your Voicemail (if you are subscribed to Voicemail).

■NOTE:

If you plan to include your own TN as one of the Locate Me routes, you should place it in the 1st route to ensure ring-through in event that you are in the 911 Restricted state.

Forwarding Behavior

Call forwarding gives you the unique advantage of applying different call treatments to different groups of callers. For example, you can have all calls from your VIP group members forwarded from your desk phone, to your mobile phone, to your pager, and then to your mailbox. Alternatively, you can select to have all devices ring at the same time. However, call forwarding can only function correctly if all the prerequisites are met, and all the necessary information has been entered correctly.

■NOTE 1:

A call treatment is defined as the various numbers you want your incoming calls forwarded to, in a specified order. For example, you might want your incoming calls to ring your desk phone first, and then be forwarded to your lab phone, then to your cellular phone. Call treatments and forwarding to multiple numbers helps ensure that you never miss important calls. In addition, all of your messages can be left in your voicemail box. Now, you only need to give callers a single number and you only need to check one voicemail box.

The option button at the bottom of the **Locate Me** frame of the Home page lets you set your call forwarding treatment.

The following examples describe call forwarding behavior that occurs under specific conditions:

- If you do not enter any phone numbers for your call treatments and enable call forwarding, all incoming calls are treated as a normal call. This means, calls are forwarded to your normal setting, usually your desk phone.
- If you do not enter all of your personal contact numbers in your profile, you cannot select them directly when creating a call treatment. If you do not have a mailbox, your calls cannot be forwarded to a mailbox. For example, you do not have a mailbox, but you add a contact to your refuse list and configure the system to send the calls from this contact to the non-existent mailbox. In this case, your phone rings until you answer it, or until the caller disconnects the call.
- If you forward calls to an external number with a mailbox, and the external messaging system picks up the forwarded call, all subsequent call treatments are ignored. Since the call was answered by the external messaging system, the call is shown in your incoming calls list, not the missed calls list.
- Your call treatments are ignored when you are part of a hunt group and receive a call that has rolled to your extension from the hunt group.
- Call treatments are ignored if another user within the company intercoms you, unless you are currently on a call on the primary line.
- * Codes cannot be used in call treatments.
- The 911 emergency number cannot be used in call treatments.
- You cannot use a hunt group number in a call treatment. The execution of the treatment stops at the hunt group number because hunt groups have priority over call treatments.

- A call to an employee configured with Locate Me drops after the configured ring count on the final Locate Me target if the employees COS settings do not include VM <u>unless</u> the employee has specified another number to call if the called party is busy.
- If work phone busy, call: is set.— If this option is set, any time a call is made to the work phone when the work phone has an active call, the caller is transferred to the number set in the If work phone busy, call: field. This is also the case if the work number appears on different devices. For example, any calls to that number while that number has an active call results in the caller being transferred to the number set in the If work phone busy, call: field.

Adding a User Group/Speed Dial to a Company Directory Contact

You can add a user group to a Company Directory Contact in your personal directory. This allows you to apply call treatment settings for this caller (e.g., you want to have the same Locate Me call treatments for the group **VIPs** applied for calls from your supervisor). This user group and speed dial will only be available to you from your Personal Web Site. Other users will not see these entries.

The following describes how to add a user group and/or a Speed Dial to a Company Directory Contact.

- 1. Search the **Company Directory** for a contact.
- 2. Click on the contact's name. This will display the **Edit Contact** screen. As a user, you can only change the **Group** and the **Speed Dial**.
- To add a group, you need to select a value from the drop down menu in the Group field. The following values are available: VIPs, Co-workers, Family & Friends, and Refuse. Figure 63 shows an example of the Group menu choices.

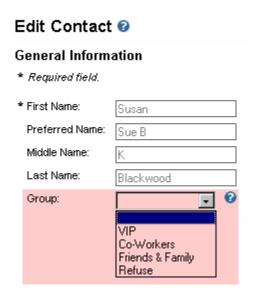


Figure 63. Edit Contact Group Field

4. To add a Speed Dial, select a personal speed dial from the drop down menu. Figure 64 shows an example of the Speed Dial menu choices.

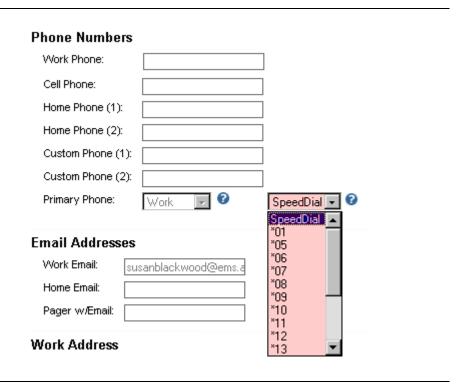


Figure 64. Edit Contact Speed Dial Field

5. Click on the **Save** button to save your changes.

Locate Me Scheduler

The **Locate Me Scheduler** allows you to schedule how your calls will be forwarded for the week (Sunday through Saturday). For example, you could schedule all your calls to ring your desk phone on Monday 9-5, all your calls to ring your cell phone all day on Tuesday, etc.

To configure and activate the **Locate Me Scheduler**, you need to:

- Set the default call treatment. This is where all calls will be forwarded for time slots that do not have a scheduled call treatment.
- 2. Create a schedule of call treatments.
- 3. Activate the **Locate Me Scheduler** on your Personal Web Site.

The Locate Me Scheduler option is located under the Call Management menu.



Figure 65. Locate Me Scheduler from the Navigation Bar Menu

Configure Locate Me Scheduler

The are two parts to configuring the **Locate Me Scheduler**:

- 1. Set the default number to which all calls will be forwarded if there are no call treatments for a time slot.
- 2. Add any call treatments to the scheduler.

Set the Default Call Treatment on the Locate Me Scheduler

The following describes how to set the default call treatment for the **Locate Me Scheduler**.

Select the Locate Me Scheduler option under the Call Management menu.
 The system responds with the Locate Me Scheduler page similar to Figure 66.

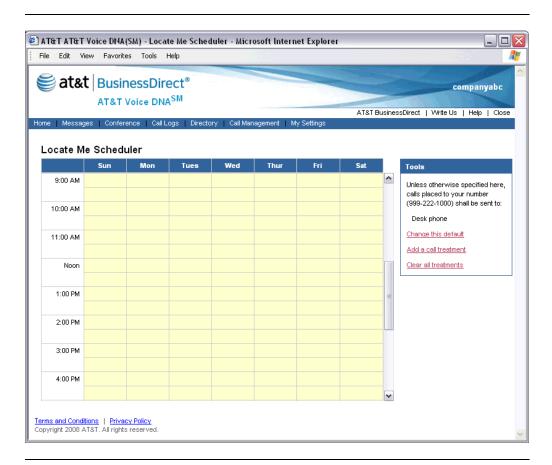


Figure 66. Blank Locate Me Scheduler

- 2. The Tools portion of Figure 66 indicates the default call treatment for calls made to your telephone number when the Locate Me Scheduler is active. This default call treatment is invoked when a call is received, Locate Me Scheduler is active, and a call treatment has not been specified for that time/day. In this example, any call made to 999-222-1000 is sent (by default) to your Desk Phone provided the call falls outside of the call treatment criteria (time/day) as indicated on the schedule.
- To change the default setting, click the Change this default link in the Tools frame. The system responds with a page similar to Figure 67.

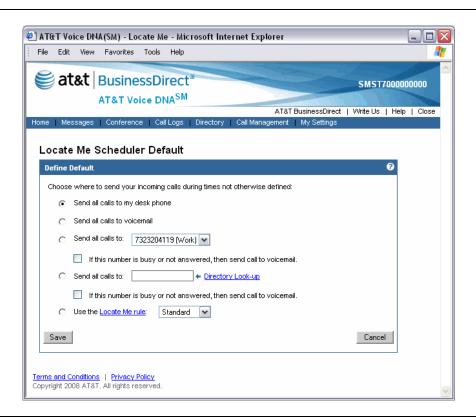


Figure 67. Locate Me Scheduler Default

- 4. You can change the default setting to one of the options listed on Figure 67. This default call treatment is invoked when a call is received, Locate Me Scheduler is active, and a call treatment has not been specified for that time/day.
- 5. After you have made any changes, click on the **Save** button to save the changes. The system responds with a confirmation page similar to Figure 68.



Figure 68. Change Default Confirmation

Add a Call Treatment

The following procedure describes how to add a call treatment to the schedule.

- Select the Locate Me Scheduler option under the Call Management menu.
 The system responds with the Locate Me Scheduler page similar to Figure 66.
 If there are existing call treatments, they will appear on the schedule.
- 2. To add a new call treatment, click the **Add a call treatment** link in the **Tools** frame. The system responds with a page similar to Figure 69.

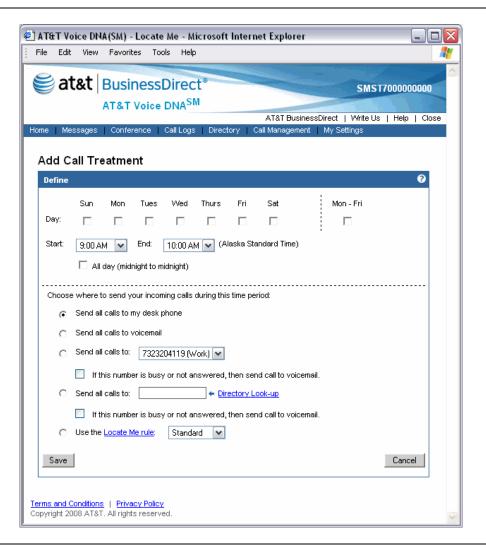


Figure 69. Add Call Treatment

- 3. Select the following on the Add Call Treatment page:
 - Day Indicates the days you want the call treatment to be active. Select the
 checkboxes for which days you want the call treatment to be active. You can
 select the checkboxes for individual days (e.g., Mon, Wed, etc.) or the
 checkbox for Mon-Fri.
 - Time Indicates the times you want the call treatment to be active. Select the Start time and the End time from the drop down menus. The times are displayed in your default time zone and will be indicated to the right of the End time. You can also select the All day (midnight to midnight) checkbox if you want the call treatment to be in effect for all 24 hours.
 - Locate Me Indicates the action the call treatment should initiate. For example, Send all calls to voicemail, Use the locate me rule, etc.

Example Scenario:

In this example, we want the call treatment to be active on:

- Mon, Wed, and Fri.
- 9:00 a.m. to 5:00 p.m.
- Send all calls to my desk phone

Figure 70 shows an example of how the completed page would appear.

Items to remember:

- Any times that have default background color (light yellow) will have the default call treatment applied. Times that have call treatments will have a white background.
- Call treatment times cannot overlap.
- Maximum allowed number of call treatments per day is 5.
- Maximum allowed number of call treatments per week is 35.
- Rings all registered phones and line appearances.
- Schedule retained in the database if the user de-selects the Locate Me
 Scheduler but is removed if the Administrator disables Locate Me.
- Maximum allowed call forwarding unconditional treatments is 15.
- The scheduler is not resumed when Call Forward (*72/*73, *62/*63, *92/*93) or DND (*78/*79 analog phones only) are turned off via Feature Codes. The default will be set back to Send all calls to my desk phone.
- No integration with:
 - Voice DNA Outlook Integration.
 - The user's Microsoft Outlook or any other desktop calendar.

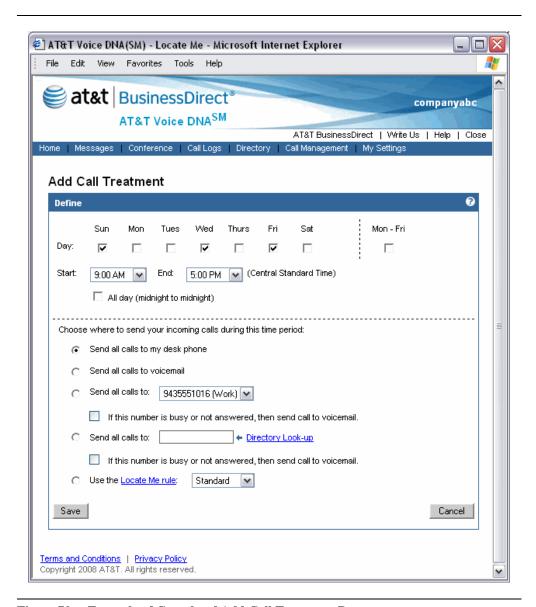


Figure 70. Example of Completed Add Call Treatment Page

4. Once you have entered the desired call treatment criteria, click on the **Save** button to save the information. The system responds with a confirmation page similar to Figure 71.



Figure 71. Add Call Treatment Confirmation

Edit a Call Treatment

The following procedure describes how to edit a call treatment on the scheduler.

Select the Locate Me Scheduler option under the Call Management menu.
 The system responds with the Locate Me Scheduler page similar to Figure 72.

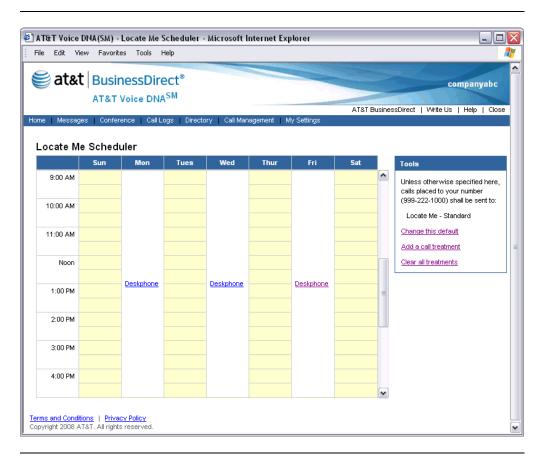


Figure 72. Locate Me Scheduler with Entries

2. Click on call treatment link (located in the body of the scheduler) that you want to edit. Clicking on a link inside the scheduler causes the Edit Call Treatment page to be displayed for the selected call treatment. For example, if you clicked on the <u>Deskphone</u> link for Fri in Figure 72, this would allow you to edit the call treatment named **Deskphone** for Friday. Figure 73 shows an example of the Edit Call Treatment page.

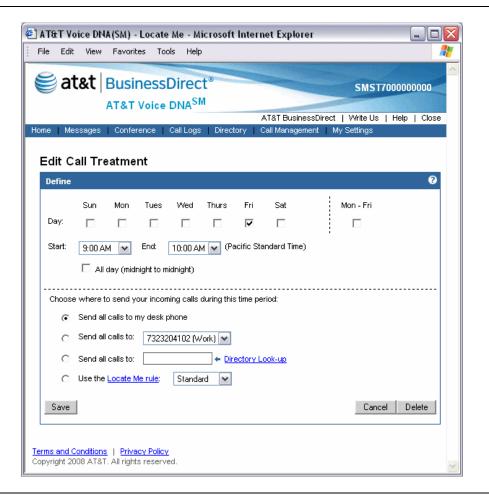


Figure 73. Edit Call Treatment

3. Make the desired changes and then click on the **Save** button to save the changes. The system will respond with a confirmation screen.

Delete a Call Treatment

The following procedure describes how to delete a call treatment on the schedule.

- Select the Locate Me Scheduler option under the Call Management menu.
 The system responds with the Locate Me Scheduler page similar to Figure 72.
- 2. Click on call treatment link (located in the body of the scheduler) that you want to delete. This causes the Edit Call Treatment page to be displayed for the selected call treatment. For example, if you clicked on the <u>Deskphone</u> link for Fri in Figure 72, this would allow you to delete the call treatment named <u>Deskphone</u> for Friday. Figure 73 shows an example of the Edit Call Treatment page.
- 3. Click on the **Delete** button to delete the call treatment. The system will respond with a confirmation page as shown in Figure 74.

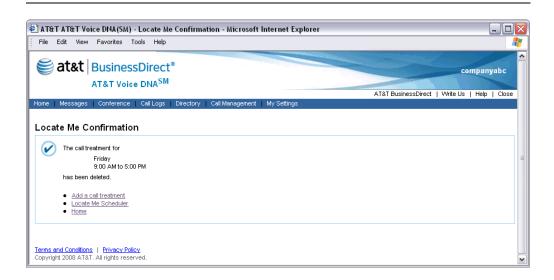


Figure 74. Delete Call Treatment Confirmation

Clear All Call Treatments

The following procedure describes how to clear all of the call treatments on the scheduler.

- Select the Locate Me Scheduler option under the Call Management menu.
 The system responds with the Locate Me Scheduler page similar to Figure 72.
- 2. Click on the **Clear all treatments** link in the **Tools** frame. The system responds with a warning message as shown in Figure 75.



Figure 75. Clear Call Treatment Warning

3. Click the **OK** button to clear all call treatments from the scheduler. The system responds with a confirmation message as shown in Figure 76.

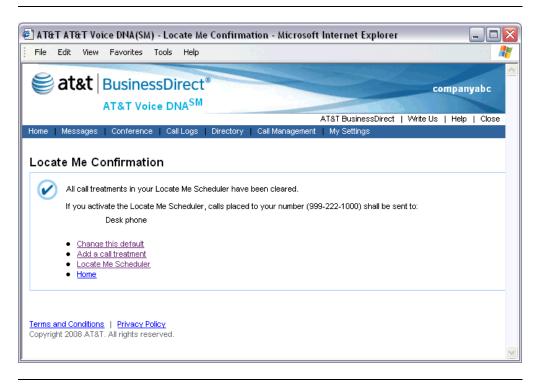


Figure 76. Clear Call Treatments Confirmation

Activate Locate Me Scheduler

The **Locate Me Scheduler** is activated from the **Home** page of the Personal Web Site. Select the option button **Use the <u>Locate Me Scheduler</u>** link in the **Locate Me** frame. Figure 77 highlights where the link is located.

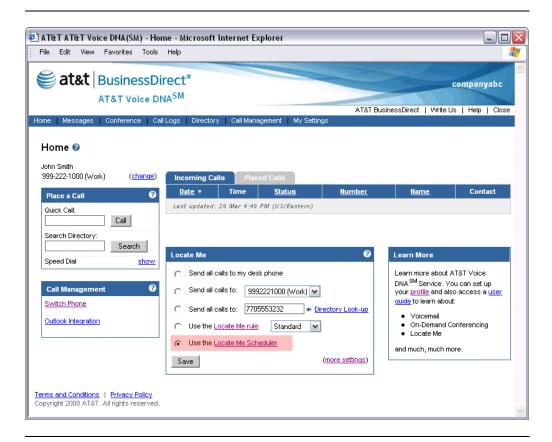


Figure 77. Home Page with Locate Me Scheduler Link Highlighted

Move a Call in Progress

To move a call in progress to another phone line, use the **Call Management** feature, **Switch Phone** (Mid-Call Move).



Figure 78. Switch Phone (Mid-Call Move) from the Navigation Bar Menu

On the **Switch Phone** (Mid-Call Move) page, you can transfer a call to another phone. Only calls that are dialed using the Personal Web Site calling function can be transferred to a different phone.

■NOTE 1:

Switch Phone can only be applied to calls that are in progress and have been initiated via a click to call.

■NOTE 2:

You can only move a call one time. For example, if you move a call from your work phone to your cell phone, you cannot move the call from your cell phone to another number.

■>NOTE 3:

Switch phone does not cut through the voice path until the called party answers, therefore; switch phone cannot be used to transfer the call to a phone number that goes through an 8YY call prompter application.



Figure 79. Switch Phone (Mid-Call Move) Page

To transfer your call to a different phone:

- On the Call Management menu, select Switch Phone (Mid-Call Move) from the drop down menu.
- 2. On the **Switch Phone** (Mid-Call Move) page, select the phone number to which you want to transfer this call, from the drop down menu. Alternatively, enter a number in the blank entry field below it.
- 3. Click the **Transfer** button.
- **4.** Answer the call on the other phone, and hang up the original phone.

Call Forking

The Administrator can assign more than one phone to your extension (Call Forking feature). The Call Forking feature sends an incoming call simultaneously to one or more SIP devices registered to the same extension. For example, if a user has three SIP devices registered to their extension, an incoming call rings all devices at the same time. When the user answers the call at one of the phones/devices, all other phones stop ringing.

Making a Phone Call

You can also use the **Call Management** menu to make a phone call.



Figure 80. Phone Call from the Navigation Bar Menu

- 1. Go to **Call Management** on the navigation bar menu.
- 2. Pull down to Phone Call.
- 3. Enter a number to call.
- 4. Click the Call button.

NOTE 1:

For the following types of incomplete call attempts, the user will hear a fast busy:

- Invalid number
- Network congestion
- Number changed
- Number not in service
- Number not assigned

■NOTE 2:

Switch phone does not cut through voice path until the called party answers, therefore; switch phone cannot be used to transfer the call to a phone number that goes through an 8YY call prompter application.

Viewing Call Logs

The Personal Web Site shows summary information, or **Call Logs**, of your Voicemail (including fax messages), Incoming Calls, and Placed Calls. You can page through these Call Logs using the **Previous** and **Next** buttons.



Figure 81. Call Logs Table, Messages Tab

A CAUTION:

If your SIP User ID does not match the 10-digit phone number associated with you, you will not see your messages on your Personal Web Site. You will need to use the Telephone User Interface (TUI) in order to get your messages. For more information, contact your Administrator.

≡>NOTE:

You will not be able to return calls (by clicking on the number) from the Call Logs for international calls due to current dial plan limitations.

≡>NOTE:

If your calls are routed to a Shared Mailbox, voice and fax messages are *not* listed on the **Messages** tab. These calls are routed to the Shared Mailbox Owner. See **Shared Mailbox Message Retrieval** on page 44 for more information.

Viewing the Messages Log

To view a list of messages:

- 1. Click the **Messages** tab at the top of the table (it appears by default).
- 2. This tab shows you a list of messages (voicemail, emails, and faxes) that have been received, the date and time of each message, the number, and the name of each person who sent the message (if available).

Making Calls to Messages Numbers

- 1. Click the telephone number listed.
- 2. This gives you the **Phone Call** screen.
- 3. Click the Call button to call that number.

Viewing the Incoming Call Log

To view a list of received calls:

1. Click the **Incoming Calls** tab at the top of the table.



Figure 82. Incoming Calls Tab

This shows you a list of the last 10 incoming calls, the date and time of each call, the number and name of each person who called, and the status of each call (Received or Missed).

■NOTE:

Any calls originated from the Voicemail service (Messages Call Back Now, Zero Out Attendant) will not have a respective entry in Placed Calls Call Logs. This is true whenever the call is originated from a location other than the owned phone (PSTN, other Users, etc.).

Making Calls to Incoming Call Numbers

To call someone on the Incoming Calls list:

- Click the telephone number listed.
- 2. This gives you the Phone Call screen.
- 3. Click the Call button to call that number.

■NOTE:

You will not be able to return calls from the Personal Web Site for international calls due to current dial plan limitations.

Editing Directory Contact Information for an Incoming Call Name

To edit the contact information for someone listed in the Incoming Call list:

- 1. Click the Contact symbol ¹ for that entry line.
- 2. This gives you the **Edit Contact** screen for that person.

Adding a Contact from the Incoming Call List

To add someone from the Incoming Call list to your Directory:

1. Click the Contact symbol ¹ for that entry line.

Viewing the Placed Call Log

To view a list of calls placed from this telephone number:

1. Click the Placed Calls tab at the top of the table.



Figure 83. Placed Calls Tab

2. This shows you a list of the last 10 calls placed, the date and time of each call, the number and name of the person who called, the status of each call (Placed), and the duration of each call in hours: minutes: seconds.

Making Calls to Placed Call Numbers

To make a call to a number on the Placed Calls log:

- 1. Click the telephone number listed.
- 2. This gives you the **Phone Call** screen.
- 3. Click the Call button to call that number.

■NOTE:

You will not be able to return calls from the Call Logs for international calls due to current dial plan limitations.

Editing Directory Contact Information for a Placed Call Name

To edit the contact information for someone listed in the **Placed Call** log:

- 1. Click their name of the contact.
- 2. This gives you the **Edit Contact** screen for that person.
- 3. Make the required changes to general information, phone numbers, etc.
- 4. Click the **Sav**e button to save your changes.

Adding a Contact from the Placed Call List

To add someone from the Placed Call list to your Directory:

1. Click the Contact symbol 1 for that entry line.



Figure 84. Add Contact Link

- 2. On the Add Contact screen, enter that person's information.
- 3. Click the **Save** button on the **Add Contact** screen to save your changes.

Another Way to View Call Logs

You can also work with your **Call Logs** by using the **Call Logs** item in the navigation bar menu.

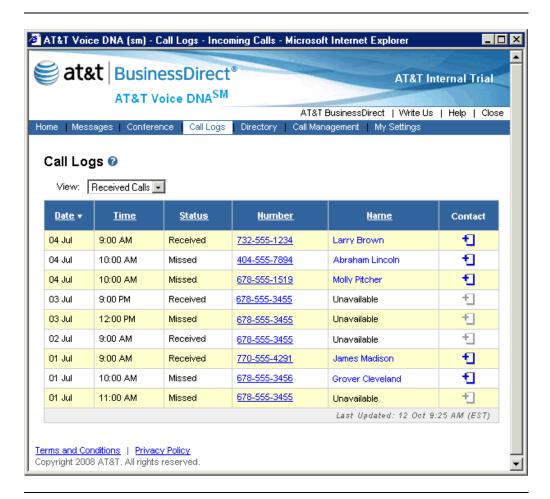


Figure 85. Call Logs from the Navigation Bar Menu

On the Call Logs page, use the drop down menu to view logs of **All Calls**, **Missed**, **Incoming**, or **Placed** Calls.

Time Zone for Call Logs

Call logs show a time and date for each call based on the time zone set in your User Profile as set by your administrator. If this time is not correct, contact your administrator with the correct time zone setting.

Changing Your Settings

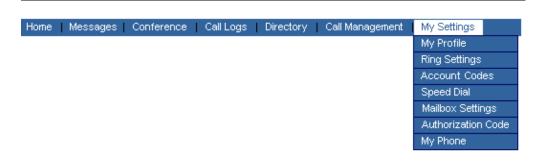


Figure 86. My Settings From the Navigation Bar Menu

The **My Settings** item on the navigation menu allows you to set or change your Profile, set Ring Lengths for all your phones, view your Account Codes, set up your Speed Dial numbers, change your Voicemail Password, change your Authorization Code, and personalize your Web-enabled desktop phone.

■NOTE:

The first time you log into the web site you should click **My Settings** in the navigation bar, verify the existing information, and add information to complete your profile. The telephone number information is used to help you easily set up your call forwarding.

Editing My Profile

The **My Profile** screen is where you set and change your contact information – your phone numbers, email addresses, and physical addresses. You can update all of the fields except the fields under the General Information section (**Title**, **Work Phone**, **Work Email**, etc.). These fields can only be updated by an administrator.



Figure 87. My Profile From the Navigation Bar Menu

Adding/U	ndating	your Phone	Numbers
Auuiliz/ U	puanng	YUUI I HUHE	Munipers

Work:	770-555-1212	
Fax:		
Cell:	678-555-1111	Publish Number in Directory
Home 1:	404-555-2222	Publish Number in Directory
Home 2:	404-555-2637	Publish Number in Directory
Custom 1:		Publish Number in Directory
Custom 2:		Publish Number in Directory

Figure 88. Phone Number Fields

1. On My Settings, select My Profile from the drop down menu.

- 2. Enter your phone numbers for your Fax, your Cell, your Home, and, if necessary, use the two additional **Custom** fields for extra numbers.
- 3. You can choose whether to publish the numbers in the Directory. Click the checkbox next to the number to have it included in the Directory. Other applications (e.g., Company Directory, etc.) will only have access to information that is set to be published.

■NOTE:

Fax numbers are automatically published to the directory, so the **Publish Number in Directory** checkbox is not available next to the **Fax** field.

4. Use the drop down menu to set one of these numbers as your primary number. The default primary number setting is Work. The primary number is used (default) when making outgoing calls from your Personal Web Site, and is listed in the Company Directory Main Search as your (primary) telephone number. This setting does not affect incoming calls to your Work TN (you would need to use the Locate Me settings if you wish to forward calls to the Primary TN).

■NOTE 1:

The Primary number is the number that displays in the main company directory for this employee. Therefore, if you can change your primary to your cell phone, anyone searching the directory will see that number first. You can still always click on an entry to see all published TNs (but no extension) associated with that user. Your Primary number is always published in the Directory. This includes any numbers that are set to **not** be published in the Directory. For example, if you set your Primary Number to your Cell Phone, your Cell Phone will be published in the Directory (regardless of the **Publish Number in Directory** settings under **My Profile**).

■NOTE 2:

For extension only users, if you wish to change the Primary TN back to your extension, select **Work** from the Primary TN drop down list on the My Profile Page. Do not use the **change** link on the Personal Web Site homepage.

Adding Email Addresses

mail Addresses	i	
Pager with email:		Publish Address in Directory
Home email:		Publish Address in Directory

Figure 89. Email Address Fields

1. Type your email addresses, as appropriate, into the Pager with Email and Home Email fields. Note that these addresses are in addition to your work email.

Adding your Home Address

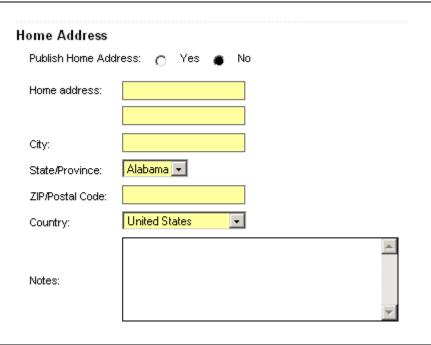


Figure 90. Home Address fields

- 1. Select the option button, **Yes** (to include) or **No** (do not include, default) to indicate whether you want your home address published in the Directory. Other applications (e.g., Company Directory, etc.) will only have access to information that is set to be published.
- 2. Enter your Street Address.
- 3. Enter your City.
- 4. Select your State from the drop down list.
- **5.** Enter your Zip or Postal Code.
- 6. Select your Country (United States is the default).
- 7. Add Notes, as necessary or desired.
- 8. Click the **Save** button to save your phone numbers, email addresses, and home address. Use the **Clear** button to clear the existing data and start over. Clicking on the **Cancel** button closes the window and returns you to the previous screen.



Figure 91. Save, Clear, and Cancel Buttons

Ring Settings

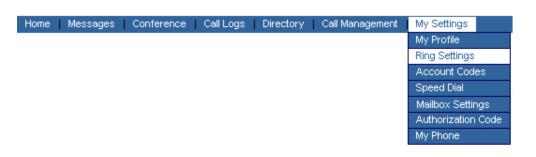


Figure 92. Ring Settings From the Navigation Bar Menu

The **Ring Settings** item under the **My Settings** menu lets you establish ring lengths, from 4 (about 1 ring) to 32 (about 8 rings) seconds, for each of your phones (Work, Cell, Home-1, etc.) and to enable/disable Distinctive Ringing.

Ring Lengths

Calls coming to you through the AT&T Voice DNA will ring according to your Ring Lengths setting, before moving along your path to your next **Locate Me** phone or option. You can speed up the call forwarding process by entering shorter ring lengths, and you can define longer ring lengths for calls to numbers that may take you longer to answer, such as your cell phone.

■NOTE:

If you set the Ring Lengths for a call option (e.g., Cell) to be less than the number of rings before that phone's Voicemail is activated, the call will skip to the next option since the phone was not answered.

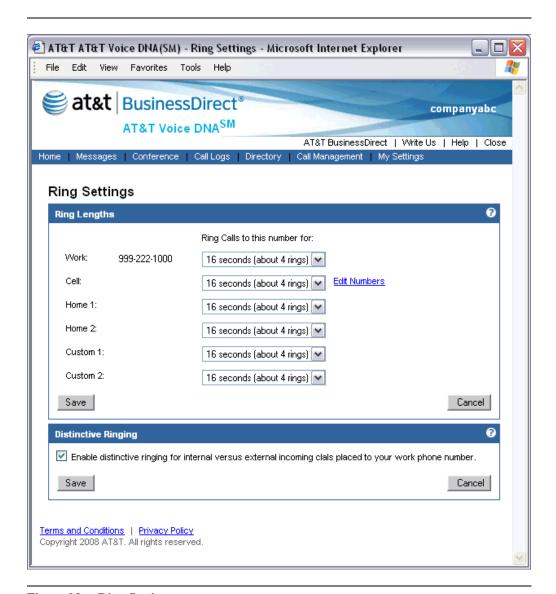


Figure 93. Ring Settings

For each phone:

Use the drop down menu to select the ring length, from 4 seconds (about 1 ring) to 32 seconds (about 8 rings). If set to **No Timer**, the phone will continue to ring until someone picks up or the caller hangs up. Click on the **Save** button located in the **Ring Lengths** section to save any changes to ring lengths.

■>NOTE 2:

If you are using the Reminders Messages feature of the **Telephone User Interface** (see page 349), be aware that the ring settings for the reach numbers might prevent the call from being delivered to all of the designated numbers when the in-order option is selected. If multiple numbers are used the ring setting needs to be set at or below 8 seconds or 2 rings.

Setting Up Distinctive Ringing

If you would like to hear a different ring for calls from inside your company as compared to the ring for calls from outside of your company, you can select this option on the **Ring Settings** page.

In the **Distinctive Ringing** section on the **Ring Settings** page, select the checkbox for distinctive ringing and then click the **Save** button located in the **Distinctive Ringing** section. Distinctive ringing will apply for calls to your work phone number when the checkbox is active. If you want to use the phone based ring choices, the Distinctive Ringing checkbox must be unchecked (disabled). The phone will need to be rebooted (power cycle) for the changes to take effect.

There are 3 aspects to Distinctive Ringing:

- 1. **Ring Choices by Line Appearance:** Configurable ring choice per line appearance, including silent ring (Visual only) option.
 - Configurable via the phone menu.
 - Number of ring choices available varies by phone type.
 - Repetitions of the same extension will share a ring treatment.
- 2. **Delayed Ring** Ability to select a ring tone that has visual only alert for first 2 rings, followed by an audible ring.
 - Configurable via the phone menu as a ring choice.
 - At least 2 Delayed Ring tones will be supported such that a user can differentiate between 2 lines requiring delayed ring on the same phone.
 - On the LG phones, the delayed ring is a separate option and can be applied in conjunction with any of the listed ring types.
- 3. Caller ID based Ring Ability to set distinctive rings for specified Caller ID.
 - Configurable via the phone menu, phone contact settings.

Important Notes for Distinctive Ringing

- This feature applies to Polycom and LG phones.
- Distinctive Ring must be disabled for the given extension for phone based ring choices to apply.
- If a ring choice is made for both a contact and a line and that contact calls in on that line, the contact choice takes precedence.
- Ring Choices will not apply if an alert info header is sent from the system (e.g., BLA).

Editing Numbers

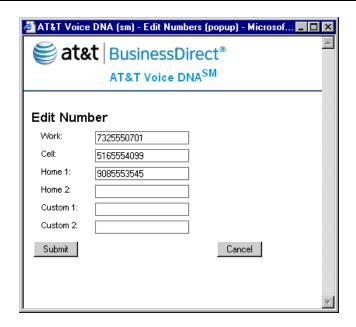


Figure 94. Phone Number Fields

Use the **Edit Numbers** link in the **Ring Lengths** box to add or update your telephone numbers for Work, Cell, Home-1, Home-2, Custom-1, and Custom-2.

- 1. Click the Edit Numbers link.
- 2. Type in the phone numbers you want to add or update.
- 3. Click the **Submit** button to save your changes.

Account Codes

≡>NOTE:

Account Codes are restricted to phone types that support them.



Figure 95. Account Codes From the Navigation Bar Menu

The Account Codes item under the **My Settings** menu shows the account codes that have been set up by your Administrator. You can use these codes to allocate costs to specific customers, efforts, or groups. You cannot edit these codes; they are displayed in the Account Codes table for your use.

Account Codes let employees assign specific codes to some or all of their outgoing calls. These codes can then be used to bill calls to clients, departments, or projects.

Once the administrator has created an account code, it is added to the list of available account codes in every employee's Personal Web Site. Employees can then select this code from the drop down menu when making a phone call. You can also enter the code on your phone before making external calls.

After account codes are added, it can be can specified as whether they are optional or required for each employee. If account codes are required, calls are not connected until a valid account code is entered. Account Codes can be entered when the placing a call from the Personal Web Site. For example, if you used the **Phone Call** menu item to make a call, you might see a screen similar to Figure 96. Note that the Account Code for this call would be **10000**. If no account code was required, this field would show **No account codes found** or **None**.

■NOTE 1:

The Administrator creates the Account Codes and sets the Account Codes to be optional or mandatory in the Class of Service template. However, it is up to you to select/enter the correct Account Code when making a call.

≡>NOTE 2:

At the moment, Speed Dial and Optional Account Codes will not work together.

■NOTE 3:

Blind Transfer and Mandatory Account Codes are not compatible. If you attempt to Blind Transfer a call from a device that subscribes to Mandatory Account Codes and the transfer attempt is to a location that is not intratenant, the call drops.



Figure 96. Example of Making a Phone Call that Contains a Account Code

Account Codes can also be entered directly on the desktop telephone (non-LG), as follows:

• If account codes are not required (optional), enter *50, followed by the account code, <send>, the offnet dialing code (e.g., 9, outside line), the telephone calling number, and then press <send>. The <send> at the end (after the account code) is optional. This <send>, eliminates the 4-second delay before the system initiates the call. So the format would be:

*50<Account Code><send><Offnet Dialing Code><TN><send>

(e.g., 50*88888<send>97325551313<send>)

If account codes are required, enter the offnet dialing code (e.g., 9, outside line), the telephone calling number, press <send>, the account code, and then press <send>. So the format would be:

<Offnet Dialing Code> <TN> <send> <Account Code> <send> (e.g., 97325551313<send>88888<send>)

■>NOTE 1:

For the mandatory account codes, the final <send> is required.

■>NOTE 2:

If you have a Cisco phone, you cannot directly enter an account code.

Account Codes are never required for calls to other internal extensions or to 911 emergency services.

Entering Optional Account Codes with LG Phones

To make a call using Optional Account Codes using an LG phone, perform the following:

- 1. Press *50, followed by the Account Code (e.g., 1234).
- 2. Wait for call to be made or press the [Call] softkey.
- 3. The LCD displays **[Enter Number]**. Enter the telephone number that you are trying to call and press the **[OK]** softkey.

Authorization Code

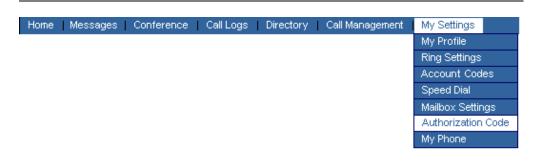


Figure 97. Authorization Code From the Navigation Bar Menu

The Authorization Code feature allows an employee to unlock a restricted phone using an authorization code to make an external call by dialing the authorization feature code *80 followed by their extension, the authorization code, and the number they want to call separated by *. For example, if the employee's extension is 2340, and their authorization code is 8888, to call the external number 770-555-4567, enter *80*2340*8888*7705554567. Employees can change their Authorization Code under the My Settings menu item on their Personal Web page. The authorization code can consist of 4 to 6 digits.

The Authorization Code assigned to an employee is listed on the **Authorization Code** page (Figure 98). This page allows you to enter your Authorization Code.

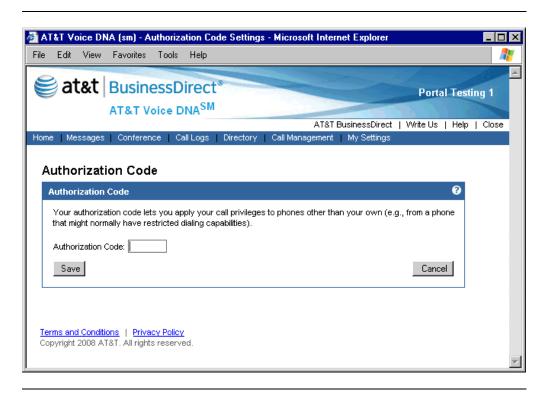


Figure 98. Authorization Code Page

My Phone

Phone(s) assigned to an employee's extension are listed on the **My Phone** page. Figure 99 shows an example. Figure 100 shows an example with a fixed destination.

You can have multiple IP phones registered to the same extension. The maximum number of IP devices that can be registered to the same extension depends on the configuration AT&T created for your company. Your Administrator sets the limit for each employee individually using the *Max. Forked Extensions* field in the Administrator Tool. However, the limit per user cannot exceed the general company registration limit that was set by AT&T.

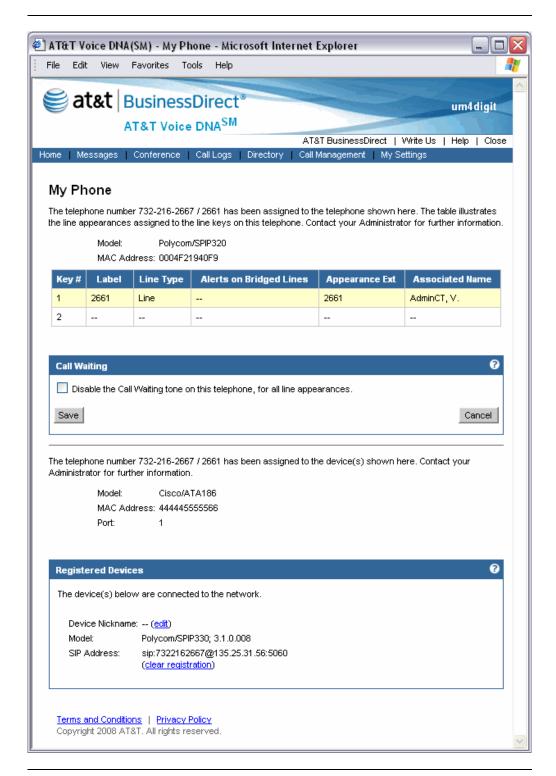


Figure 99. Example of My Phone

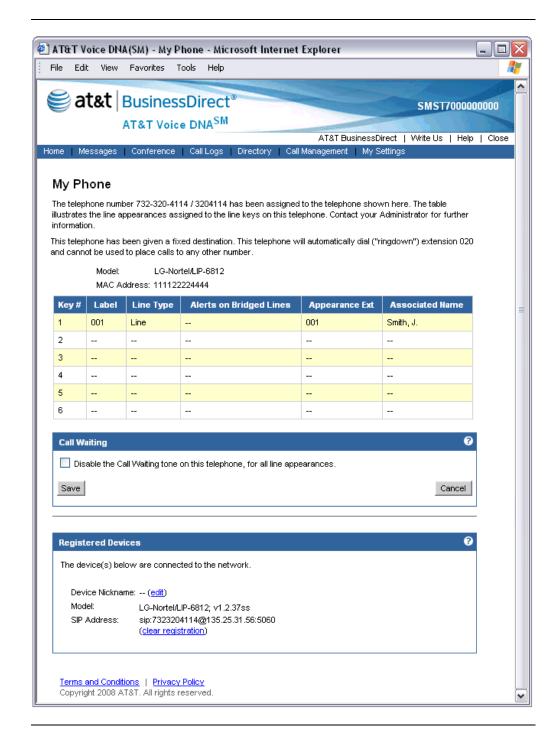


Figure 100. Example of My Phone with a Fixed Destination

Table 2 describes the information on the My Phone page.

Table 2. My Phone Page Information

Section	Data	Description
Text	Fixed Destination Message	Indicates if phone has been given a fixed destination. If this text is displayed, the phone will automatically dial the assigned number.
Line	Key#	Key number in the display.
Appearances	Label	If the phone supports labels, any label that has been entered for this line will be displayed. If the phone does not support labels, this column will not be displayed.
	Line Type	Line or Bridged
	Alerts on Bridged Lines	Audio & Visual, Visual Only, or blank.
	Appearance Ext	The appearance extension.
	Associated Name	Name associated with the line.
Call Waiting	checkbox	If you have Call Waiting tone enabled, whenever you are on a call and receive a second call to your extension, you will hear a Call Waiting tone. You may disable the Call Waiting tone on this telephone by selecting this checkbox. Disabling the Call Waiting Tone If you have the Call Waiting feature, when you are on a call and a new call comes in to the line, you will hear a tone to alert you that a call is waiting. If you prefer, you can disable the call waiting tone. If the call waiting tone is disabled, when you are on the phone and a second calls comes in, there is still a visual indicator on the phone to signal that a call is waiting. To disable the call waiting tone, on the My Phone page, find the Call Waiting section. Click the checkbox labeled Disable the Call Waiting tone on this telephone, for all line appearances. and then click Save.
Registered Devices	Device Nickname	The optional nickname for this phone. To enter/edit a nickname for a phone, click the <u>edit</u> link located in the Device Nickname field to open the My Phone Nickname page. You can then modify the nickname.
	Model	Indicates the type of phone.
	SIP Address	The port information for this phone. Port information is displayed differently, depending on the type of phone, as follows: Adapter ports appear as port#@serialnumber, for example d006@0001-2332-1667-7771. IP phones appear as sip: <sip id="" user="">@<ip address="">:<port>, for example sip:4086263075@172.16.1.96:5060.</port></ip></sip>
	clear registration	To clear a phone registration click on the <u>clear</u> registration link of the phone experiencing problems with call handling.

My Phone Nickname

To edit the nickname for a phone, click the <u>edit</u> link located in the **Device Nickname** field to open the **My Phone Nickname** page. You can then add or modify the nickname. Once you have made the desired changes, click the **Save** button. An example of the My Phone Nickname screen is shown in Figure 101.

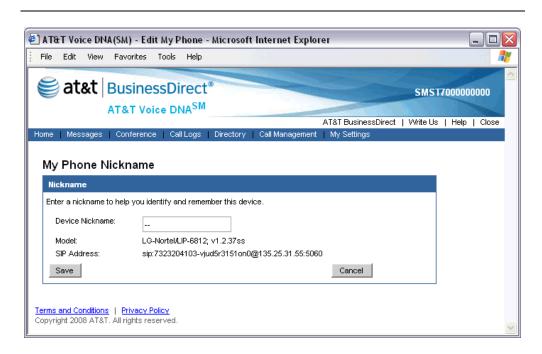


Figure 101. Example My Phone Nickname

Disable Call Waiting

If you have Call Waiting tone **enabled**, whenever you are on a call and receive a second call to your extension, you will hear a Call Waiting tone.

To disable the Call Waiting tone on a telephone (all lines):

- Select the checkbox associated with the text Disable the Call Waiting tone on this telephone, for all line appearances located within the Call Waiting section for the telephone.
- 2. Click the **Save** button to save the change.
- 3. After saving the change, wait five minutes and then unplug the telephone and then reconnect the telephone. This resets the telephone. Note that the telephone may take several minutes to restart.

If this telephone has additional line appearances (for other extensions), whenever you are on a call and any extension receives a second incoming call (i.e., that extension is already busy on another call), and if you have Call Waiting tone **enabled**, you will hear a Call Waiting tone.

If this telephone has additional line appearances (for other extensions) and any extension receives an incoming call (and that extension is <u>not</u> already busy on another call), your telephone will ring. The Call Waiting tone setting (enabled or disabled) will not change this.

Clear a Phone Registration

If the device behavior is not as expected, it may be necessary to clear the device and reset. The following procedure describes how to clear the registration then allow the device to re-register.

■>NOTE:

Registrations for other phones associated with the User TN in a forking arrangement are legitimate, and do not need to be deleted. This is only used for deleting phones registrations with problems.

 To clear a phone registration, under My Settings → My Phone, in the Registered Devices section (Figure 99) click on the <u>clear registration</u> link of the phone experiencing problems with call handling. The My Phone Clear Registration page is displayed (Figure 102).

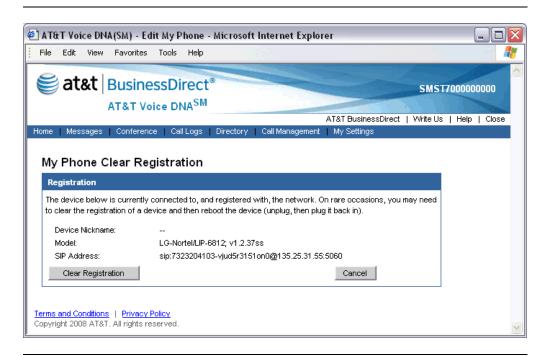


Figure 102. My Phone Clear Registration

- 2. On the **My Phone Clear Registration** screen (Figure 102), click on the **Clear Registration** button. The system responds with a confirmation page. This is not deleting the phone, but only clearing the phone registration. Click on the **Cancel** button to cancel the operation.
- 3. Once the phone registration for the phone type have been cleared, wait five minutes and then reboot the phone (power disconnect). This can be done by unplugging the device and reconnecting the device. Allow the device to reregister. Once the device has re-registered, call your extension. If the correct device rings, you are done.

Outlook Integration Software

The Outlook Integration software can be downloaded from the Administrator Tool under the Admin Services section or the Personal Web Site (if they are subscribed to the feature).

Required Information for the User

Before you begin installing Outlook Integration, you need the following information:

- User name (your BusinessDirect user name)
- Password (your BusinessDirect password)

PC Requirements for Outlook Integration

The following are system requirements for using Outlook Integration:

- Windows 2000 or later, XP Operating System.
- Outlook 2000 or later must be contained on the PC.
- Administrative privileges are required to install Outlook Integration plug-in.
- User must be authorized for Locate Me and Call Forwarding capabilities.

Download and Install Outlook Integration

Before you begin installing Outlook Integration, you need the following information:

- User name (your BusinessDirect user name)
- Password (your BusinessDirect password)

■NOTE:

The **Outlook Integration** link used to download the software only appears on a user's Personal Web Site if they are subscribed to this feature.

To install the AT&T Outlook Integration software:

- 1. Close Microsoft Outlook® if it is running.
- Go to the Outlook Integration installation page by clicking on the Outlook
 Integration link. You can download the software from the Personal Web Site under Call Management or on the Administrator Tool under Admin Services.

■NOTE:

Do not change the telecom services URL provided in the download. The system automatically provides the correct URL.

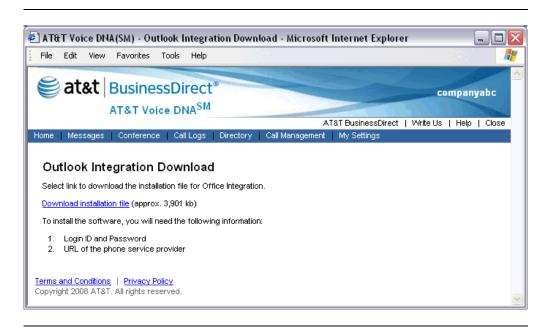


Figure 103. Outlook Integration Download Screen

3. Click on the **Download installation file** link. A window similar to the following will be displayed:



Figure 104. Download Installation File Screen

4. Click on the **Save** button. At the prompt, specify a location on your PC where you want to save the **attoutlook.exe** file, for example C:\. You can save the file to the location of your choice. The window will appear similar to the following:

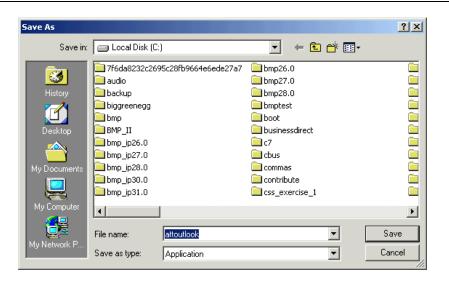


Figure 105. Save As Screen

5. After the file has been downloaded to your PC, on the Download complete page (Figure 106), click on the **Run** button to execute the **attoutlook.exe** file and install Outlook Integration on your PC.

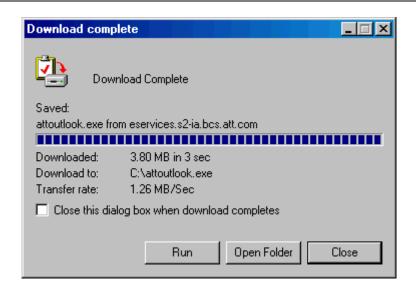


Figure 106. Download Complete Page

If you choose not to install Outlook Integration at this time (or have the checkbox Close this dialog box when download completes selected), you will need to locate the attoutlook.exe file and double click on the file to start the Outlook Integration installation. When the installation starts, a few windows will be displayed briefly. After these windows disappear, the Outlook Integration Setup Wizard starts as shown in the following window.

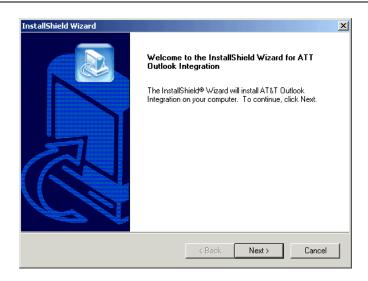


Figure 107. InstallShield Wizard Screen

Click on the Next button. If Microsoft Outlook is running, you will see a window similar to the following that states you must close Microsoft Outlook before proceeding.

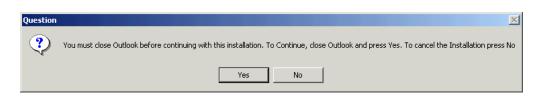


Figure 108. Microsoft Outlook Warning Screen

If you see this window, close Microsoft Outlook before clicking on the **Yes** button. If Microsoft Outlook was not running or when you close Microsoft Outlook, the system responds with a window similar to the following.

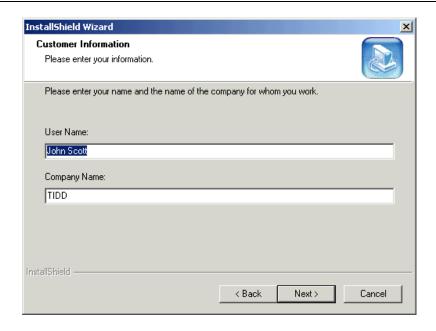


Figure 109. Customer Information Screen

7. Enter your User Name and Company Name. This User Name does not have to match the BusinessDirect User Name. Once you are finished entering this information, click on the **Next** button. The system starts to install the application. You should see a window similar to the following while the application is being installed.

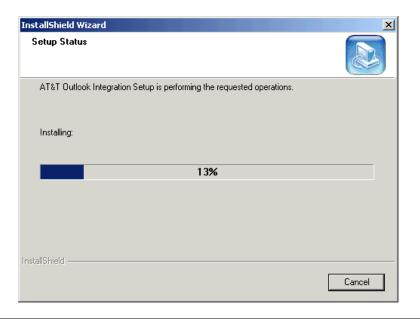


Figure 110. Setup Status Screen

8. Once the application has installed, the system responds with a window similar to the following.



Figure 111. Wizard Completion Screen

- 9. Click on the **Finish** button to complete the installation.
- 10. Restart Microsoft Outlook to complete your settings.
- **11.** For information on your setup, refer to the Outlook Integration documentation. To locate this documentation:
 - a. Start up Microsoft Outlook if not running.
 - b. Click on the ATT Voice DNA menu.
 - c. Click on the ATT Outlook Integration Help menu item located on the AT&T Voice DNA menu. The following shows an example of how the menu will appear. The documentation will appear in a separate pop-up window.

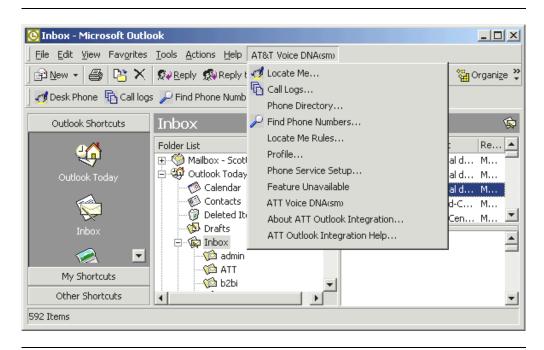


Figure 112. AT&T Outlook Integration Help

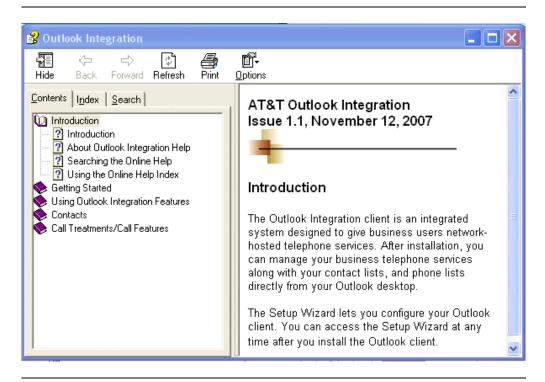


Figure 113. Outlook Integration Help Window

Emergency Services

The Voice DNA is capable of sending the 911 calls you dial to the Public Service Answering Point (PSAP) that provides emergency services to your area, provided that your telephone number is within the geographic area normally associated with your actual location and your telephone information has been sent to the PSAP and placed in its records. If your Voice DNA is being delivered to you at your home address, or at a location remote from the normal corporate offices, please contact your administrator and ascertain what emergency services are available at that location, then follow any cautions or advice your administrator gives you. Please note that you will not be able to reach 911 operators in any situation in which the data link that serves your location is inoperative, or if electricity is lost or turned off to your computer. Additionally, in any situation in which you access your Voice DNA from a location other than your normal office location, such as from your home, from another location when traveling, or away from the office, you will not be able to reach an emergency operator who services the actual place from where you are calling. Whenever you need to contact an emergency operator from a location other than your normal work location, or in situations in which your Voice DNA is inoperable for any reason, do not attempt to use your Voice DNA, but instead always use a traditional telephone at that location or a cell phone. For more information about emergency services and your Voice DNA, please consult the Voice DNA Service Guide.

■NOTE 1:

Transferring a caller to 911 is not recommended. When Call Transfer is invoked, the calling party TN for the 911 call is the Voice DNA subscriber and the caller is connected to the Voice DNA user's PSAP with the Voice DNA user's information (Voice DNA user's TN and associated address information). Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly.

■>NOTE 2:

You should not use Click to Call when calling 911.

■NOTE 3:

Subscriber calls are restricted when 911 restrictions are applied. Subscriber call restriction is 2-way (e.g., impacts inbound and outbound calls for a subscriber). So, when Subscriber calls are restricted, the user cannot make or receive intratenant calls and all user initiated calls (Locate Me, Hunt Group, Call Distribution) are also restricted.

≡>NOTE 4:

Inbound calls are allowed to terminate to the phone when the user in the restricted state, with the exception of Locate Me Rules Call Forwarding where the external TNs are ignored entirely, and internal TNs are ignored for secondary routes. The call will go to voicemail if no answer, or the routes will be ignored.

911 Move Detection and Restriction

This feature detects device movement and takes action based on the resulting device locations for the user so that the appropriate outbound calling number will be used for a 911 call (to support routing to the responsible PSAP) and/or restrict the user's dialing capabilities if their devices are in a non-supported configuration. In this case, the user is restricted and cannot make Intra-tenant calls both inbound and outbound. This also means that you will not be able to receive calls as part of a Call Distribution queue and you will not be able to initiate calls from the Personal Web Site. Inbound PSTN calls are allowed.

■>NOTE:

Inbound calls are allowed to terminate to the phone when the user in the restricted state, with the exception of Locate Me Rules Call Forwarding where the external TNs are ignored entirely, and internal TNs are ignored for secondary routes. The call will go to voicemail if no answer, or the routes will be ignored.

Voice DNA Site User

If you are a normal user and your phone has been restricted, you will see a warning message on the Home Page as shown in the following examples.

If a user is restricted and is a regular Voice DNA (not remote worker) user, the user's phones must be returned to a valid configuration. The restriction will automatically be lifted when this is done. This means:

- The user who is restricted answers Yes to the restricted warning question.
- The user who is restricted could move the **phone causing the restriction** back to a tenant site, so that all phones are at the same tenant location.
- The administrator could delete the phone causing the restriction from the user in the case when the user has multiple phones and one of them has been moved causing the configuration to be invalid.
- Once one of the preceding items has occurred, the restriction status is recalculated and if the remaining phones are in a valid configuration, then the restriction would be lifted.

A valid configuration is one where all phones are at the same tenant location (location served by the tenant's managed router). Phones served by a tenant location with a managed router will all have an IP address associated with the managed router.

Example 1: This is an example of the message that you might receive if you have an interruption in your network connection. If you answer **Yes**, your phone will be unrestricted. You can determine the current 911 Emergency Registered Location for your telephone number by contacting the AT&T 911 Center at 800-356-2972.

Warning: Your AT&T Voice DNA service has been restricted. The service has detected a recent interruption in your network connection. Please indicate the location of your telephone by clicking either the Yes or No link below:

 $\underline{\underline{Yes}}$ My telephone is located at its current 911 Emergency registered location No My telephone is at a different location

You can determine the current 911 Emergency registered location for your telephone number by contacting the AT&T 911 Center at 800-356-2972.

Example 2: If you answered **No** in Example 1, you will receive a message similar to the following. This message will tell you how to change your address to clear the restriction or to move the phone. You can determine the current 911 Emergency Registered Location for your telephone number by contacting the AT&T 911 Center at 800-356-2972.

Warning:

You have indicated that your telephone is not at its 911 Emergency registered location. Your service will remain restricted until you either:

- -return the phone to its registered location
- -update the 911 Emergency registered location to correspond to your new location by contacting the AT&T 911 Center at 800-356-2972.

Example 3: This example shows a message where the user moves the phone to the home location and has a device that has been nomadic. You can determine the current 911 Emergency Registered Location for your telephone number by contacting the AT&T 911 Center at 800-356-2972.

Warning: Your AT&T Voice DNA service has been restricted. The service has detected that your phone was recently at a nomadic location and has since returned to its primary location. Please indicate if the current 911 Emergency registered location is set to your primary location by clicking either the Yes or No link below:

> Yes My 911 Emergency registered location is set to my primary location No My 911 Emergency registered location is NOT set to my primary location You can determine the current 911 Emergency registered location for your telephone number by contacting the AT&T 911 Center at 800-356-2972.

Remote Worker or Voice DNA Nomadic User

If you are a Remote Worker or Voice DNA Nomadic User and your phone has been restricted, you will see a warning message on the Home Page as shown in following warning message:

Warning: Your AT&T Voice DNA service has been restricted. The service has detected a recent interruption in your network connection. Please indicate the location of your telephone by clicking either the Yes or No link below:

 $\underline{\underline{Yes}}$ My telephone is located at its current 911 Emergency registered location $\underline{\underline{No}}$ My telephone is at a different location

You can determine the current 911 Emergency registered location for your telephone number by contacting the AT&T 911 Center at 800-356-2972.

■NOTE:

If you have a standard feature package, you will need to call your Administrator to clear the restriction for you. If you have enhanced or premium feature package, you can go to Personal web portal to clear the restriction or call your Administrator.

For a Remote Worker or Voice DNA Nomadic User, valid configurations are nomadic and all devices are at the same tenant location. Each time a Remote Worker or Voice DNA Nomadic User phone is reset from a nomadic location, the primary extension is restricted. If all the User's devices are at the same non-tenant location, then the User can answer **Yes** to the question on their Personal Web Site to clear the restriction. If they are not in a valid configuration (say one is nomadic and the other is at a tenant location), then the User will not be able to remove the restriction through the Personal Web Site. To clear the restriction from the Personal Web Site, the phones would have to be moved to a valid location. An alternative is available; the Administrator can delete a phone, resulting in the remaining phones being in a configuration that will allow the restriction to be cleared.

FAX User

When a FAX port has been restricted:

- Since a FAX User does not have a BusinessDirect login (a BusinessDirect login is needed to access to the AT&T Voice DNA Personal Web Site), the User needs to contact their Administrator to inform them that the TN has been restricted. Only the Administrator can clear the 911 restriction for a FAX.
- The 2 ports on Cisco ATA 186 act as independent devices, hence the analog devices (phone or FAX) plugged into the ATA need to be cleared independently.

For example: If an ATA 186 has one voice TN and one FAX TN, only the voice TN can be cleared through the AT&T Voice DNA Personal Web Site, if the voice TN has an Enhanced or Premium package. The FAX TN must be cleared by the Administrator. If an ATA 186 has 2 voice TNs, each TN must be cleared through its AT&T Voice DNA Personal Web Site independently.

Restricted Call Treatment

When a call is blocked due to TN being in restricted mode, the caller experience will hear an intercept message.

Registering Temporary Locations (for Remote Workers)

Remote Workers can register a *temporary* nomadic location, such as a hotel room, so that emergency calls can be directed to the appropriate Public Safety Answering Point (PSAP) while the worker is at the temporary location.

Remote Workers may register a temporary location in one of two ways:

■ Change 911 Address tool - Some Remote Workers can log in to AT&T BusinessDirect and use the Change 911 Address tool.

■NOTE:

Only Remote Workers whose user profiles have been appropriately provisioned can use the online tool. To find out of you are provisioned for the tool, contact your AT&T Voice DNA company administrator.

■ AT&T 911 Center - The toll-free number for AT&T 911 Center is listed in the AT&T Voice DNA User Guide (800-356-2972).

During the registration process, whether via the **Change 911 Address** tool or the AT&T 911 Center, AT&T will check that AT&T Voice DNA service is available at the temporary location, and will only register the temporary location if service is available or permitted. If E911 service is not available at the location, AT&T Voice DNA service is not available from that location, and the Remote Worker is not permitted to use the service at that location. The Remote Worker will be advised if E911 service is not available at the location.

When the phone is *returned* to the *primary* remote work location, the remote worker must either access the **Change 911 Address** or *call back* the AT&T 911 Center and request that the Registered Location be reverted to the primary work location. A remote worker can have only *one* Registered Location at any time.

Change 911 Address Tool

The **Change 911 Address** tool allows a remote worker to register a temporary 911 address for one or more phone numbers. The number of phone numbers a Remote Worker can work with depends on the permissions selected in the User's profile. Depending on your permission level as the User, you are authorized to register temporary 911 addresses for one of the following:

- One phone number authorized to work with your only your own phone number. This is the most common configuration.
- 2-20 phone numbers authorized to work with a set of between 2 and 20 company phone numbers.

 All phone numbers associated with an MCN/GRC/SOC account – authorized to work with a set of phone numbers defined by the MCN/GRC/SOC number(s) for your company.

Using the Change 911 Address Tool

 To access the Change 911 Address tool, log in to AT&T BusinessDirect and select Change 911 Address as shown in Figure 114. For instructions on logging in to AT&T BusinessDirect, see Logon Procedures.

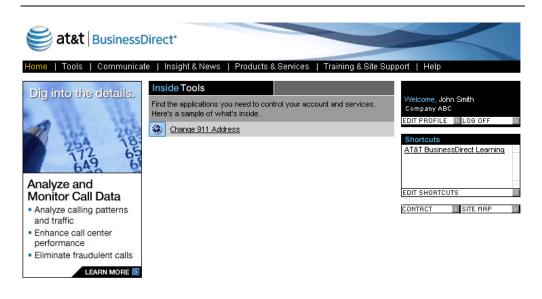


Figure 114. Business Direct Window with Change 911 Address Tool

2. The 911 Address Change Home page is displayed. The fields displayed on your home page will depend on your permission level for such a change. Figure 115 is an example of a home page for a User with permission to work with only one phone number. Click Next to proceed.



Figure 115. 911 Address Change Home Page (One Phone Number)

■NOTE:

While you are using any of the functions in the **911 Change Address** tool, do not use any browser function such as, **Back**, **Forward**, **Stop**, or **Refresh**. If you do so, it will interrupt your request and cause inaccurate information to be displayed.

Figure 116 shows a home page for a User with permission to work with between 2 and 20 phone numbers. Figure 117 shows a home page for an employee who has permissions to work with all of the phone numbers within an MCN/GRC/SOC account. While, as a User, your permissions may allow you to work with more than one phone number, you can change the address for only *one phone number at a time*. Therefore, to begin, you must indicate which phone number you want to work with. In either case make a search selection and click **Search**. Use the search results to select the phone number you want to work with and proceed to **Step 3**.

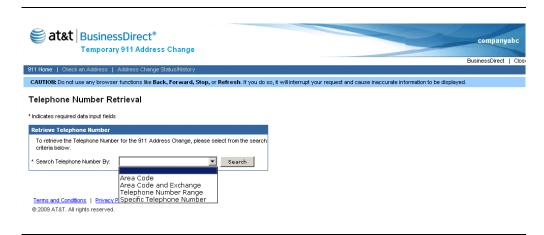


Figure 116. Telephone Number Retrieval (2 to 20 Phone Numbers)

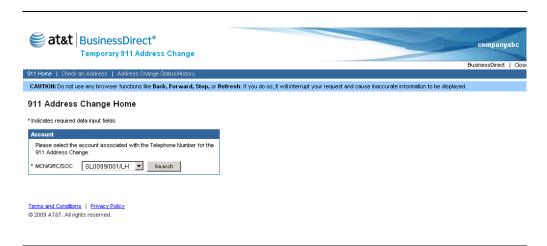


Figure 117. 911 Address Change Home Page (MCN/GRC/SOC)

After selecting the phone number, you will choose whether you want to enter a
new temporary address or select an address from a list of temporary addresses
you have previously registered.

As shown in Figure 118, click the radio button corresponding to your selection.

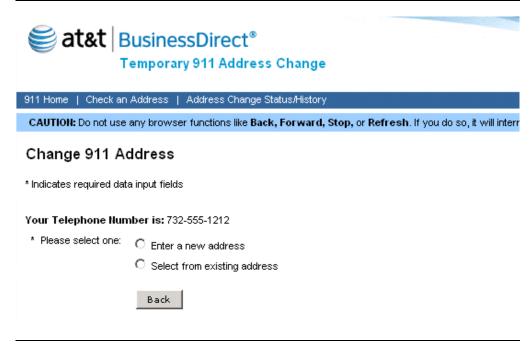


Figure 118. Select a New or Existing 911 Address

4. The screen changes depending on the selection you made in the previous step. If you selected Enter a new address, a screen similar to Figure 119 is displayed. Enter the required information and click Submit.

■NOTE:

The following message appears at the bottom of the page:

If applicable, please enter your Room/Floor/Apt and/or Building/Wing/Pier information as it will better direct emergency services to your location. Please note that, while the majority of Public Service Answering Points can receive this information, some do receive Room/Floor/Apt or Building/Wing/Pier information.

If you selected **Select from existing address** in the previous step, a screen similar to Figure 120 is displayed. Select a previously registered address from the drop down menu. If the address you selected displays correctly, as shown in Figure 121, click **Submit**.

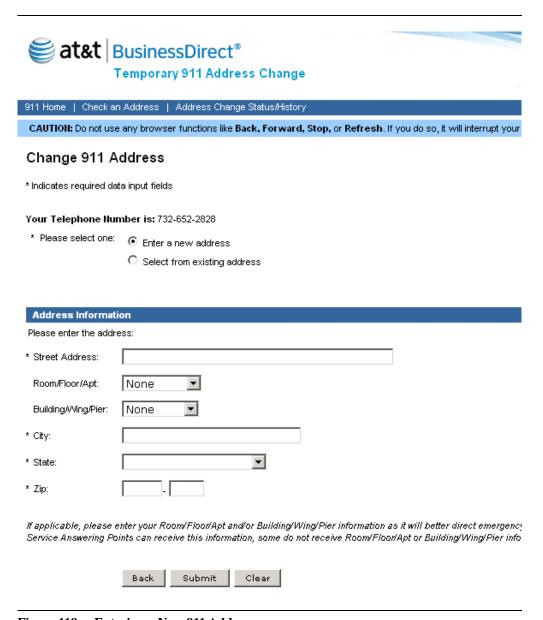


Figure 119. Entering a New 911 Address

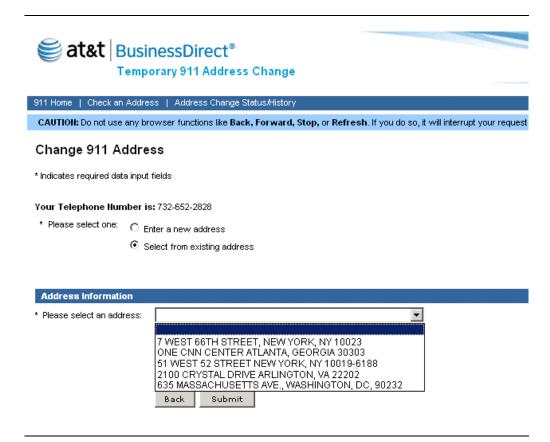


Figure 120. Selecting an Existing Address

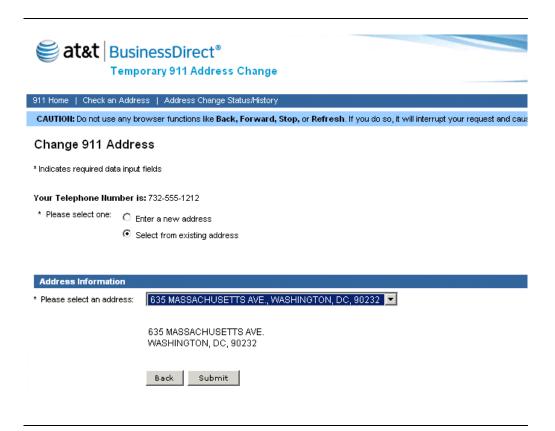


Figure 121. Selected Existing Address

5. The Authorization page, as shown in Figure 122, is displayed with the information you've entered along with the authorization statement. If the information is correct and you agree with the statement, you can proceed with authorizing the 911 address change by clicking I authorize AT&T to change my 911 address.

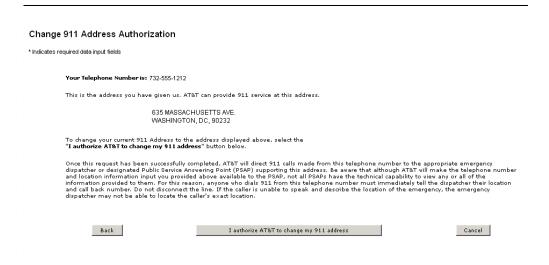


Figure 122. Authorization Page

6. A confirmation screen, similar to Figure 123, is displayed. It states that your request for a temporary 911 address change has been successfully submitted and that a change confirmation email, similar to Figure 124 will be sent to you.

≡>NOTE:

The User is *not* authorized to move the phone until he or she has received a second email message from AT&T confirming that the temporary address change is *completed* as shown in Figure 125.

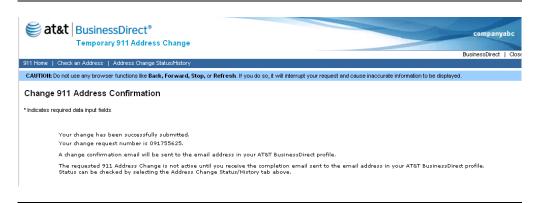


Figure 123. Change 911 Address Confirmation

```
DATE: Tuesday, January 27, 2009
TO: John Smith
FROM: AT&T E911 Temporary Address Change

The E911 Temporary Address Change requested on Tuesday, January 27, 2009 via AT&T BusinessDirect has been placed. Please
NOTE: You are NOT authorized to use your phone at the requested location until you receive a completion email.

Request Number: 091755625
Date Requested: 01-27-2009
Activity Request: Change E911 Primary Address to Temporary Address
Telephone Number: 732-555-1212
```

You will receive a completion email once your request has been completed. Until you receive the completion email, your present E911 address information will remain in effect. You are NOT authorized to use your service at the new location until that completion email is issued. Once the completion email is received, your new address will become effective, and will remain the E911 address for your telephone number until you request a change.

You may view details of your request by logging into AT&T BusinessDirect at www.businessdirect.att.com, accessing the E911 Temporary Address Change Tool from the home page and then clicking on the Address Change/History tab on the top of the screen.

If you did not submit this request, please call our Customer Care Center immediately at 1-800-356-2972 to speak to a representative. It is critical that you do this immediately to ensure your safety in an emergency situation.

Please do not respond to this email, as it is an unattended mailbox.

Thank you for using AT&T.

Figure 124. Change Confirmation Email

```
DATE: Friday, January 27, 2009
TO: John Smith
FROM: AT&T E911 Temporary Address Change
The E911 Temporary Address Change requested on Friday, January 27, 2009 via AT&T BusinessDirect has completed successfully.

This activity request has set the E911 address associated with telephone number 732-555-1212 to the following address:
635 MASSACHUSETTS AVENUE
WASHINGTON, DC 90232

In the event you dial 911 from 732-555-1212 emergency services will be dispatched to the address above. This address will remain in effect until you submit another E911 Temporary Address Change request. Accordingly, when you leave the above address and use your AT&T service at another location, it is very important that you register your new address, so that emergency calls will be directed to the correct location. Please note that not all emergency operators receive the complete address information for the caller. Accordingly, it is important to provide your exact name, address and call back number to the emergency operators.

You may view details of your request by logging into AT&T BusinessDirect at www.businessdirect.att.com, accessing the E911 Temporary Address Change tool from the home page and then clicking on the Address Change/History tab on the top of the screen.

If you did not submit this request, please call our Customer Care Center immediately at 1-800-356-2972 to speak to a representative. It is critical that you do this immediately to ensure your safety in an emergency situation. Please do not respond to this email, as it is an unattended mailbox.

Thank you for using AT&T.
```

Figure 125. Change Completion Email

Checking the Status of a 911 Address Change Request

You can check the status of your 911 address change requests.

1. Select Address Change Status/History from the navigation bar. A screen similar to Figure 126 is displayed.



* Indicates required data input fields

Please select the request number to view the detail of the request.

Date	Request No.	Telephone Number	Status	Submitted Via
01/27/2009	091755625	732-555-1212 Completed		Web
01/27/2009	091755572	732-555-1212	Completed	Web
01/27/2009	091755372	732-555-1212	Completed	Web
01/27/2009	091755288	732-555-1212	Completed	Web
01/27/2009	091754923	732-555-1212	Completed	Web
01/27/2009	091754907	732-555-1212	-1212 Completed	
01/19/2009	091746493	732-555-1212	555-1212 Completed	
11/06/2008	<u>081681701</u>	732-555-1212	Completed	Web
11/05/2008	081680483	732-555-1212	Completed	Web
10/24/2008	081666604	732-555-1212	Completed	Web
08/25/2008	<u>081588651</u>	732-555-1212	Cancelled	Web
08/14/2008	081575224	732-555-1212	Completed	Web
08/06/2008	081567500	732-555-1212	Completed	Web
07/29/2008	081557882	732-555-1212	Completed	Web
07/28/2008	081556680	732-555-1212	Completed	Web
Back				

Figure 126. Change Status/History

2. To view more detail on a request, click the request number. The details are displayed in a screen similar to Figure 127.



Figure 127. Change Status/History Detail

Checking the Availability of an Address

You can check if 911 service is available at an address.

- 1. Select Check an Address from the navigation bar.
- 2. The Check an Address screen, similar to Figure 128, is displayed.
- 3. Enter the required information into the fields and click **Submit**.

■NOTE:

When you request a temporary 911 address change, whether via the **Change 911 Address** tool or the AT&T 911 Center, AT&T will check that AT&T Voice DNA service is available at the temporary location and will only register the temporary location if service is available. The Remote Worker will be advised if E911 service is not available at the temporary location, and, in that case the Remote Worker is not permitted to use AT&T Voice DNA service at that location.

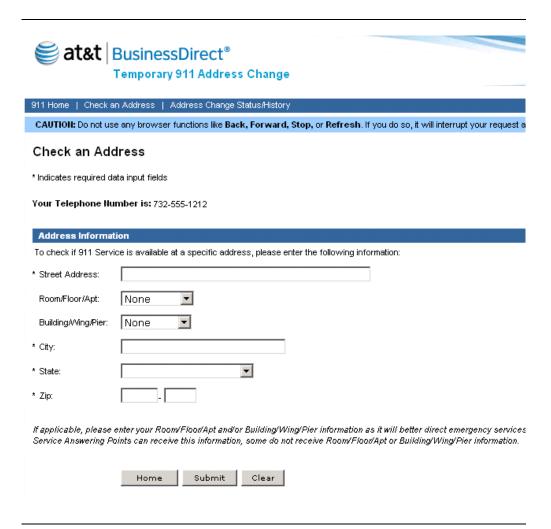


Figure 128. Check an Address

Returning to Your Primary 911 Address

When the phone is *returned* to the *primary* remote work location, the Remote Worker must either access the **Change 911 Address** tool (see **Change 911 Address Tool)** or *call back* the AT&T 911 Center (800-356-2972) and request that the Registered Location be reverted to the primary work location. A Remote Worker can have only *one* Registered Location at any time.

Restricted MLA and BLA Scenarios

Each individual BLA (Bridged Line Appearance)/MLA (Multiple Line Appearance) appearance is restricted based on the status of the TN owner.

Scenario 1:

For BLA, the behavior is as follows:

- 1. **911 restriction:** The Manager's line appearance on the assistant's phone is consistent with the Manager's restriction. Therefore, if the manager moves their device to an unknown location, the manager's appearance on the assistant's phone is also restricted (but the assistant's main line is unrestricted).
- 2. TN used as calling party number (non-911 calls): For general outbound calls, if the assistant uses the Manager's line appearance on their phone, the assistant's TN is used as the CPN (Calling Party Number).

Scenario 2:

The extension on the first line of the phone in Scenario 1 shows up as a multi-line appearance (MLA) on another phone. If the extension becomes restricted, then it will be restricted on the 1st phone and on the phone where it shows up as a multi-line appearance.

E911 Restriction Clearing

Restricted users can unrestrict their line:

- by the user on the Personal Web Site
- by the Administrator on the Administrator Tool.

Additionally, restricted users attempting to make an outbound call (other than 911, 8YY, or VM TN) will now hear an announcement:

- Informing them of their restricted status,
- Providing reason for the restriction,
- Prompting them for confirmation of their location in certain scenarios, and
- Providing directions to remove the restriction

If a line of an adapter is cleared from restriction, all clearable lines associated with that adapter are un-restricted.

The Personal Web Site and the announcement allow a line restriction to be cleared if the user is:

- A remote worker whose line location status is nomadic,
- A non-remote worker whose line location status is nomadic AND
- Who is assigned a public TN that is not the DCN or customer specified 911 destination, AND
- A non-remote worker whose line location status is home.

Otherwise, the Personal Web Site and the announcement will convey the reason for restriction, but will not allow clearing.

When a line is restricted, outbound calls (other than 911 and 8YY) are restricted when placed from any appearance of that line, including MLA or BLA appearances on other phones. With the Interactive Voice Response (IVR) Announcement clearing, any line appearance of a restricted line who attempts to make an outbound call will receive the IVR Announcement and have the ability to clear the E911 restriction (assuming the line is in a **clearable** state).

Clearing Restriction from the Telephone Keypad

If your service is restricted and you attempt to place a call, you will hear a warning message regarding the service restriction. The message will tell you the reason for the restriction and describe steps for restoring your service.

The steps for restoring your service will depend on the reason for the service restriction. For example, in some cases, if your phone is at a registered AT&T Voice DNA location, you may be able to lift the restriction using the keypad on your phone. Always follow the instructions in the recorded message to restore your service.

Clearing the Restriction from the Home Page

If your service is restricted and you log in to your AT&T Voice DNA Personal Web Site, you will see a warning message displayed on your home page.

If you are at an authorized AT&T Voice DNA service location for your company or the current registered 911 emergency location for the phone, you may click **Yes**, and your service will be restored.

If you are at a different location, you must click **No**. In order for your service to be restored, you must either move the phone back to an authorized location or register your new location with the AT&T 911 Center.

Remember, if you are an extension-only user or if your phone number is the default calling number or the 911 routing destination for your location, you are not permitted to move your phone, so if your service is restricted, you must move your phone back to its original location.

BLA Scenario:

Manager's phone appears as a BLA on a secretary's phone. Manager moves his phone to an unknown location (nomadic) and is restricted. If the secretary selects the manager's line and attempts to make an outbound call, the secretary will get the IVR announcement and can clear the restriction by pressing 1. Same is true for the manager making an outbound call from his own line. Note that the secretary should clear the restriction unless she can confirm that the manager's phone is at his service address of record.

MLA Scenario:

Bill has a TN which also appears on Mary's phone as an MLA appearance. If Bill's line is restricted and if Mary selects Bill's line and attempts to make an outbound call, she will get the IVR announcement. Mary can clear the restriction by pressing 1.

This is the end of the Personal Web Site documentation. The remaining section of the document is Reference Materials for your phone.

Call Distribution

■NOTE:

You must be subscribed to the Call Distribution Feature and have an Aastra, Polycom or LG phone that supports Call Distribution in order to use this feature.

The Call Distribution feature lets you create/manage multiple queues where calls are queued for answering by associates. These queues are created and maintained by the AT&T Voice DNA Administrator. When a call is made to a Call Distribution number, the caller enters a queue, and the phone of the next available Call Distribution associate rings. If no associate is currently available, the caller is held in the queue until an associate becomes available or until the Call Distribution timer for that queue expires. If the queue timer expires, the caller is connected to a final destination number entered for the queue. If no associate is logged into the queue, the caller is connected directly to the final destination number for the queue. While on hold, the caller will hear music and a message. This message can be a default message or may be customized announcement. Callers to the queue can press zero (0) at any time to exit the queue and be connected directly to the final destination number for the queue (note that voicemail is not supported behind a Call Distribution Queue).

■NOTE:

When a call is received in the queue, the call is forwarded to the associate who has been logged in and idle the longest period of time. For example, if there are 6 associates and associates 1 to 5 are on the phone, associate 1 ends their call, the next call goes to associate 6.

An associate assigned to the role of Queue's **Final Destination** should not use the **reject** button on the phone to dismiss a call. Using the **reject** button causes the customer to hear Ringing and then a Fast Busy (which is an undesirable customer experience). If the associate identified in the **Final Destination** is unable to take the call, they should allow it to continue to ring until the queue no longer offers the call to that associate.

The end result is that you have a call-answering group that ensures your customers get the customer support they need. For example, Call Distribution could be used for customer support, reservations, or appointments. Customers could then be provided with a specific number to call depending on what they wish to do.

■NOTE 1:

It is recommended that end-users log-off from the Call Distribution queue at the end of day or if away from their phone for an extended period.

■NOTE 2:

If Call Distribution or Hunt Group takes control of the call and if added as a destination in a Locate Me rule, they will override the remaining Locate Me steps (including bypassing the Locate Me rule final handling).

Answering a Call Distribution Call

≡>NOTE:

You can only use Aastra, Polycom or LG phones with a Call Distribution Queue.

What the Administrator Must Do

The following describes what the Administrator must do to provision an associate to use Call Distribution after the queue has been created.

- 1. Enable the associate's Class of Service (COS) template with the Call Distribution feature. Office Administrators define an employee as an Call Distribution associate by assigning the employee the Call Distribution feature in their COS template and by assigning a phone model that supports Call Distribution. This causes the Call Distribution Login soft button (Login/Logout) to appear on the employee's phone once the phone is rebooted. Employees defined as associates can then use the Login soft button to login into the Call Distribution queue and then subsequently use other soft buttons to make themselves unavailable or available (Available/Unavailable) to take calls from the queue and to log out of the queue.
- Provide the associate with a User ID (same as the Extension for the specific Queue) and Password (if required) as specified when the Call Distribution Queue was created.

What the Associate Must Do

When an associate is set up as Call Distribution Enabled, two buttons appear on their phone. These buttons are the **Login/Logout** button and the **Available/Unavailable** button. These buttons are the only way that associates can make themselves available to take calls from the Call Distribution queue. However, associates must have access (extension and/or passcode) to a queue in order to use these buttons.

In order for an associate to answer a call, the associate must be logged into the queue. Associates log themselves in and out of the queue using the **Login/Logout** and **Available/Unavailable** buttons.

■NOTE 1:

The associate must be subscribed to the Call Distribution Feature and have an Aastra, Polycom or LG phone that supports Call Distribution.

■NOTE 2:

For Aastra phones: When an associate is on an active call or misses a call, the phone/associate is put in the **unavailable** mode. The associate must manually make themselves available again through the use of the **Available** softkey.

For Polycom phones: After hanging up from a call the phone/associate is put in the **unavailable** mode. The associate must manually make themselves available again through the use of the **Available** button.

For LG phones: After hanging up from a call the phone/associate is automatically put back in the **available** mode after about 10 seconds.

The following procedure describes the steps that you must complete (except when using a Polycom 320/321/330/331 phone. See **Logging into a Queue with a Polycom 320/321/330**/331 phone) to log into a queue:

- 1. Your phone should show a **Login** soft button.
- Press the Login soft button. You are now prompted for the User ID (extension of the Call Distribution Queue. This is not your phone extension.) and Password (if Passcode was set by Administrator).
- Enter the User ID and Password and press the Login soft button again. The phone should show an Unavailable button at the bottom of the LCD. This indicates that the phone is now available to receive calls.
- 4. When a call is received, the Unavailable button changes to Available. When you receive a call, you are unavailable. Once the call ends, you remain unavailable for calls until you manually click the Available button to make yourself available to receive calls again.

≡>NOTE:

For Aastra phones: When an associate is on an active call or misses a call, the phone/associate is put in the **unavailable mode.** The associate must manually make themselves available again through the use of the **Available** softkey.

For Polycom phones: After hanging up from a call the phone/associate is put in the **unavailable** mode. The associate must make manually themselves available again through the use of the **Available** button.

For LG phones: After hanging up from a call the phone/associate is automatically put back in the **available** mode after about 10 seconds.

5. If you need to make yourself unavailable for calls, click on the **Unavailable** button. These buttons are the only way that you can make yourself available to take calls from the Call Distribution queue.

■NOTE 1:

The human head icon next to the phone label on a Call Distribution configured phone changes under various conditions:

- Configured but not logged in outline of a head.
- Logged in and available a portion of the head turns black.
- Logged in but not available the head toggles to a bold X.

≡>NOTE 2:

It is recommended that associates log-off from the Call Distribution queue at the end of day or if away from their phone for an extended period.

Table 3 describes how Call Distribution behaves given specific events. Note that the **Final Destination** is a destination for calls that are not answered by any members of your queue, such as a Customer Service number. An associate assigned to the role of Queue's **Final Destination** should not use the **reject** button on the phone to dismiss a call. Using the **reject** button causes the customer to hear Ringing and then

a Fast Busy (which is an undesirable customer experience). If the associate identified in the **Final Destination** is unable to take the call, they should allow it to continue to ring until the queue no longer offers the call to that associate.

Logging into a Queue with a Polycom 320/321/330/331 phone

- 1. Press Menu.
- 2. Select 1. Features.
- 3. Select 10. ACD Login/Logout.
- 4. Select Login.
- 5. Press $\sqrt{.}$
- 6. Enter the ACD login and press the **Ok** softkey.
- 7. Press the **Down Arrow** to enter the password.
- 8. Press √.
- 9. Enter the ACD password and press the **Ok** softkey.
- 10. Press the Left Arrow.
- 11. Press the **Yes** softkey to confirm your input and login.
- 12. Press the **Left Arrow** continuously to return to the main screen.

Table 3. Call Distribution Behavior

	Event	Call Distribution Behavior		
1.	A call comes into the queue, and one or more associates are logged into the queue and are available to take the call.	Call Distribution sends the call to the associate who has been available the longest length of time.		
2.	A call comes into the queue and no associate is logged into the queue.	Call Distribution forwards the call directly to final destination.		
3.	A call comes into the queue and one or more associates are logged into the queue but all associates are unavailable.	If no associate is available, Call Distribution places the call in the queue. As soon as an associate becomes available, the call is sent to the associate's extension.		
		If no associate becomes available within the time limit for queuing a call, Call Distribution forwards the call to the final destination after the time limit for queuing a call is exceeded.		
4.	A call comes into the queue, and associates are logged into the queue. However, the first available associate is not at their desk to take the call and they forgot to make themselves unavailable via the soft phone buttons.	Call Distribution sends the call to the first available associate. If the associate does not answer their phone, Call Distribution changes the status of the associate's extension to "unavailable" and forwards the call to the next available associate. If all other associates are unavailable, the call is queued. The call is forwarded to the extension of the next available associate or to the queue if all associates are unavailable.		
5.	An associate is logged into queue and available to take a call. They first receive a direct call to their extension, and then receive a call from the queue while they are still talking to the first caller.	Since the first call did not go through the queue, it is not affected by Call Distribution, and the associate is still "available". If the associate is next in line for a call from the queue, and their phone has two lines, an incoming queue call is sent to the associate's second line. If the associate's phone has only one line, Call Distribution changes the status of the associate's extension to "unavailable", and forwards the call to the next available associate. If no other associate is available, the call is queued.		

Table 3. Call Distribution Behavior

	Event	Call Distribution Behavior		
6.	An associate is logged into the queue and activates the Do Not Disturb feature on their desk phone.	The status of the associate is changed to "unavailable" for all calls, including calls from the queue.		
		The call is forwarded to the extension of the next available associate or to the queue if all associates are unavailable.		
		Any call made directly to the associate's TN (not received via the queue) will get the Do Not Disturb treatment.		
7.	A number of calls are still in the queue when the last associate logs out of the queue.	Call Distribution forwards all remaining calls in the queue to the final destination.		
8.	The associate receives a call directly to their extension while they are handling a call from the queue.	Since the second call did not go through the queue, it is not affected by Call Distribution.		
		The call is sent to the extension of the associate. If the associate has a second line, the direct call comes in on the second line. If the associate has only one line, the second caller either hears a busy signal or is forwarded to the associate's mailbox if they have one.		
9.	The associate receives a call from the queue while the associate is logged into	Associate is put into the unavailable state.		
	the queue. They press the reject button for the call.	Associate with a Polycom phone must press the Available button to receive calls from the distribution queue.		
		LG Phones will automatically return to the available state in about 10 seconds.		
		Using the reject button causes the customer to hear Ringing and then a Fast Busy (which is an undesirable customer experience).		

Call Distribution Queue Problems

Unavailable to Queue

If you are a Call Distribution agent and you haven't received a queue call for a period of time, check your phone status to make sure you are still available to the queue. Under certain circumstances (such as a caller hang up while routing the call to the agent), you may be marked unavailable to the queue even though you haven't recently handled a queue call. If this happens, you will need to make yourself available to the queue again.

Clear the Device

If any difficulty is encountered with receiving calls or logging into Call Distribution Queues, check the user's **My Phone** tab and registrations. If the device behavior is not as expected, it may be necessary to clear the device and reset. To do this, delete the registration and allow the device to re-register. The following steps describe this procedure.

≡>NOTE:

Registrations for other phones associated with the User TN in a forking arrangement are legitimate, and do not need to be deleted. This is only used for deleting phones registrations with problems.

- To delete a phone registration, on the Personal Web Site for the user, under My Settings →My Phone, click the phone's hyperlink in the Phone Name column for the 1st entry of the phone type experiencing problems with call handling.
- 2. On the Edit My Phone screen, click on the Delete button. The system responds with a pop-up window asking if you want to delete the device. Clicking on the OK button deletes only that one registration for the phone. This is not deleting the phone but only deleting the phone registration. Click on the Cancel button to cancel the operation.
- 3. Repeat Steps 1 through 2 to delete other registration(s) for the same phone type.
- 4. Once all of the phone registrations for the same phone type have been deleted, reboot the phone (power disconnect). Allow the devices to re-register. You can control this by unplugging all devices and plugging them back in sequentially. Once all devices have re-registered, call your extension. If the correct device rings, you are done.

Call Distribution Queue Status Issue

Voice DNA service is dependent on good connectivity from the customer LAN and continuous power supply. Call Distribution Queue status is especially sensitive to fluctuations in the network/power supply. If a user experiences problems with the queue, it is a good practice to instruct the agent to make themselves unavailable, log out, and then re-enter the queue (log into the queue and mark themselves available again).

In addition, if the user has difficulty receiving calls, it could be the result of a **stale** device registration. The user should be directed to delete all registrations on the **My Phone** tab and allow all devices to re-register. To delete a registration, click on the phone name hyperlink. A window will then appear that will give you the option to delete the phone. Choose the **Delete** button. Repeat this for all registrations on the **My Phone** tab.

Allow the devices to re-register. You can control this by unplugging all devices and plugging them back in sequentially. Once all devices have re-registered, call your extension. If the correct device rings, you are done.

Call Distribution vs. Hunt Group

Call Distribution has the following advantages over hunt groups:

- Call Distribution provides queuing, whereas hunt groups do not.
- Associates can log themselves in and out of the queue using the programmable feature buttons on their telephone.

■NOTE:

Call Distribution buttons on user's phones only function when the phone is on-hook and there is no activity or the user puts a caller on hold.

Members of a queue can make themselves unavailable at any time. For example, Call Distribution associates can make themselves unavailable when they go to lunch. They do not need to constantly log into and out of the queue. Office administrators can create multiple queues. When multiple queues have been created, associates log themselves into a specific queue by entering the extension for the queue and an optional passcode if one is specified. Office Administrators can also specify the number of callers allowed in the queue, the number of associates who can be logged into the queue at one time, and the announcement/music file that should be played for callers to the specified queue while they are waiting for an associate to answer their call.

Line Appearances

Line Appearances allows the Administrator to configure a phone with multiple independent extensions (MLA) or to authorize a user's phone with secondary lines of other phone numbers (BLA).

This feature allows you to set up:

- 1. Duplicate appearances of your own TN on the your SIP phone.
- 2. MLAs of other employees' TNs as well as BLAs of other employees' TNs on the same SIP Phone.
- 3. Duplicate MLAs of another employees' TNs as well as duplicate BLAs of other employees' TNs on the same SIP phone.
- 4. Any combination of the preceding cases.

Line Appearance Types

AT&T Voice DNA IP phones may support one or more of the following types of Line Appearances:

 Multiple Line Appearances- single extension (MLA) - Lets you have more than one phone line (of the same extension/DID) appear on your IP phone i.e. repetitions of phone owner's extension. Example: Your extension shows up on 3 buttons on your phone.

This feature is supported on all phones certified with AT&T Voice DNA service, such Polycom, LG, Cisco and Aastra phones (not on soft phones).

- Multiple Appearances, Different Extensions (MLA) You can also have a Line Appearance for a different extension or phone number allowing you to answer an incoming call to another employee's extension. These simple line appearances, also referred to as Multi-line Appearances (MLAs) allow you to have:
- Line appearances of other users' extensions/TNs on your SIP phone.
- Duplicate appearances of your own extensions/TN or other user's extensions/TNs on your SIP phone. Note that duplicate appearances only generate a single registration.

When a call comes into an MLA extension, all phones with the MLA appearances will ring. Once one line is answered, the call is no longer available to other devices (e.g., you cannot barge into the call or resume from a different handset). There is no presence status displayed on the other devices.

The actual number of multiple extensions you can assign to a phone depends on the model of the phone. Additionally, the maximum numbers of devices on which an extension can be displayed is 32.

This feature is supported on all phones certified with AT&T Voice DNA service, such Polycom, LG, Cisco and Aastra phones (not on soft phones).

Bridged Line Appearances (BLA)— A Bridged Line Appearance, similar to MLA, is an appearance of one employee's extension on another employee's phone. You can use the Bridged Line Appearance to pick up a call to the other extension, but bridged lines also have additional functions to allow employees to more closely coordinate call activity on a given line. All phones configured with a BLA of an extension, including that of the owner of the extension, are said to belong to a BLA group.

When a call comes into a BLA extension, all phones with the BLA appearances will ring. Once the call is answered by one user of the BLA group, the call status is reflected via an LED light on all phones with that line appearance and the call is still available to other devices (e.g., if the call is put on hold by the person that answered the call, any other user in the BLA group can pick up the call).

As with MLAs, repetitions of BLAs are also supported.

This feature is supported on the following phone models certified with AT&T Voice DNA service: Polycom, LG, and Aastra phones (NOT on Cisco or soft phones).

Table 4. Supported Line Appearances by Phone Model							
	Aastra 6757i/6757iCT	Cisco 7940, 7960	LG 6812, 6830	Polycom IP 320/321/330/331, 601, 650, 560	Polycom IP 4000/6000	Counterpath Eyebeam Softphone	
Repetition of own Extension (Max=5)	Yes	Yes	Yes	Yes	No*	No	
MLA	Yes	Yes	Yes	Yes	No**	No	
MLA Repetitions (Max=5)	Yes	Yes	Yes	Yes	No*	No	
BLA	Yes	No	Yes	Yes	No**	No	
BLA Repetitions (Max=5)	Yes	No	Yes	Yes	No*	No	
MLA/BLA on the same phone	Yes	No	Yes	Yes	No*	No	

^{*}Single line phone

Call Load

In order to maximize performance on the AT&T Voice DNA IP phones, due consideration must be given to the estimated Call Load (i.e. the number of calls in a given time period) on a phone, a group of phones in a BLA group or all phones at a location. Your AT&T Voice DNA administrator is provided with guidelines on how many line appearances can be configured on your phone based on the expected call load for your location.

Additionally, your AT&T Voice DNA administrator has the ability to set the number of calls per line for your Polycom IP phones.

- For Polycom 301/320/321/330/331, this value can be edited to any value 1-4.
- For Polycom 601/650/560, this value can be edited to any value 1-24.
- For Polycom 4000/6000, this value can be edited to any value 1-8.
- For LG phones, the number of calls allowed per line key is always 2 calls (cannot be edited).
- For Aastra phones, the calls allowed per line key is always 1 call per key (cannot be edited).

^{**}A Polycom 4000 or Polycom 6000 can be the monitored device but cannot be the monitoring device because they only support a single line appearance.

Line Appearances on an Aastra Phone

Nine line appearances can be provisioned, including repetitions on the Aastra 57i and 57iCT phones. The following are some important considerations when provisioning line appearances:

- Only one call for each line appearance can occur at a time (no call stacking).
- If only one line is provisioned on a phone, the first four line keys will be provisioned with repetitions of Line 1.
- It is recommended that the primary line of the phone is provisioned on at least two line keys.
- If there is only one line appearance provisioned for a telephone number and that line already has a call (either active or on hold) then a second incoming call will automatically go to voicemail. (If the phone does not have voicemail, the calling party will hear a busy signal).
 - Given that a maximum of one call for each line appearance can occur at a time (no call stacking), it is recommended that a BLA line appearance is provisioned with at least two line appearances.

The maximum number of calls per BLA line in ONE across all phones that have that BLA line – there is NO call stacking. However, if there are BLA repetitions, then the number of calls per BLA that can be supported is equal to the number of BLA appearances on a phone (e.g., 2 appearances equal the ability to manage 2 calls for the BLA lines; 3 appearances equal the ability to manage three calls for the BLA lines). This is done on a per phone basis, so that if Phone A has three BLA line appearances of extension x1234 and Phone B has only one line appearance of extension x1234, only Phone A is able to answer a second incoming call. For this reason, it is recommended that all phones sharing a BLA are provisioned with the same number of repetitions of that BLA.

■ If you want to share a call on the line with a BLA group (allow another phone to pick up the call), you need to press the **Hold** button before sharing the call with the group. For example, if line 1 is configured for BLA and you pick up a call on line 1, you must press the **Hold** button to share the call with the BLA group. If you pick up a call on line 1 configured for BLA, and another call comes in on line 2, you can pick up line 2 without putting line 1 on hold. The line 1 call will be on hold automatically; however it is on hold locally only. That is, the line 1 call cannot be shared with the BLA group. Again, if line 1 is configured for BLA and you put the active call on line on hold (by pressing the **Hold** button), the call on line 1 will be shared with the BLA group (another phone in the BLA group will be able to pick up the call).

Auto Attendant

Auto Attendant is an optional feature for Voice DNA. This feature provides call prompter capability that allows customers to route their call using a Touch-Tone® phone through customized voice menus. When a caller makes a selection from the voice menu, the call is routed to extensions, departments, or lines that are forwarded to mailboxes, or to prerecorded information or announcements.

Auto Attendant Setup

For each Auto Attendant, the Administrator must designate the Voice DNA telephone numbers that are to be used as the Call Tree Numbers (CTN). For each Auto Attendant, you can designate up to 16 telephone numbers to be used with Auto Attendant. One of these CTNs must be designated as the **Primary CTN** while the other 15 are designated as **Secondary CTNs**. These numbers (**Primary CTN** and **Secondary CTNs**) must be provided to the AT&T Voice DNA Order Manager for provisioning in the Auto Attendant. Provisioning these numbers allows them to be used for transferring and forwarding within the Auto Attendant.

The **Primary CTN** is the unique identifier for the Auto Attendant. The **Primary CTN** is used to forward calls to the AT&T-assigned Call Tree Routing Number (CTRN). The CTRN is an AT&T provided number used to route Call Tree numbers to Auto Attendant. The CTRN is provided by your AT&T Voice DNA Order Manager in the AT&T Voice DNA Customer Confirmation Document (CCD).

All other CTNs associated with the Auto Attendant are called **Secondary CTNs**. These are additional numbers, which can be used to access the Auto Attendant. Secondary CTNs can be Direct Answered (answered by Auto Attendant) when the call is forwarded to the CTRN, or they can be Live Answered (answered by an agent). Both Primary and Secondary CTNs can be **Direct Answered** (answered by Auto Attendant) when the call is forwarded to the CTRN, or they can be **Live Answered** (answered by an agent).

Auto Attendant Transfer Destinations are numbers to which a call would be transferred in Auto Attendant (e.g., Accounting, Sales, Service, etc.). These destinations are configured as part of the Auto Attendant design.

CTN Configurations

A CTN may be configured in the Voice DNA application either as:

 A virtual user that is assigned an Enhanced or Premium feature package, with calls forwarded to the CTRN

OR

 A Hunt Group, with no members, designating the CTRN as the Final Destination so as to forward calls coming into the Primary CTN directly to the CTRN

■NOTE:

One of the Hunt Group CTNs MUST be designated as a Primary CTN and the rest will be designated as Secondary CTNs.

Items to Remember

- The CTNs as well as the TNs behind the prompts (also known as transfer destinations) must all belong to the Voice DNA range of TNs.
- 8YY# TNs cannot be ordered as part of the Voice DNA range of TNs. However, a customer can order an 8YY number separately and route it to the Voice DNA Auto Attendant CTN.
- Dial by Name and Dial by Extension:
 - Extensions in the Dial by Extension option on an Auto Attendant MUST be the last 3-7 digits of the user's publicly dialable TNs.

AND

Extension populated in Voice DNA must match the extension assigned to the user in the Auto Attendant Dial by Extension directory.

- Users added to Dial by Extension directory in an Auto Attendant MUST be assigned a 10-digit publicly dialable TN.
- Click to Conference to an Auto Attendant CTN (Primary or Secondary) is not supported. That is, an AT&T Voice DNA user cannot use the Click to Conference feature to add an Auto Attendant CTN as a conference participant.

Configuring Direct and Live Answer Calls

The following examples describe how you would configure the Call Forwarding on the Personal Web Site for Direct and Live Answer calls.

Direct Answer Scenario

To have calls Direct Answered by the Auto Attendant:

- 1. Create a Hunt Group in Voice DNA.
- 2. Assign the CTN as the Lead Number of the hunt group.
- 3. Assign the CTRN as the Final Destination of the hunt group.
- 4. Do **NOT** add any members to the hunt group.

In the following example, 732-555-3652 was assigned as the Lead Number for the hunt group and 908-555-3345 as the Final destination.

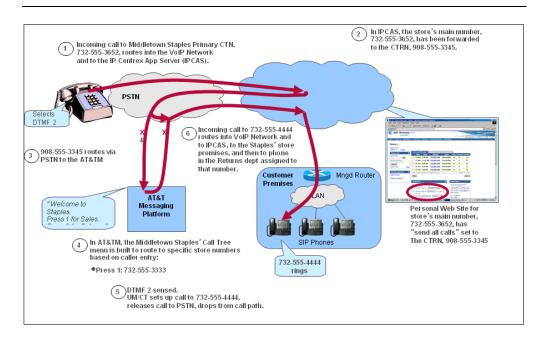


Figure 129. Direct Answer Scenario – End-to-End Call Flow

Live Answer Scenario

In the following example, the Voice DNA user called **Second Attendant** has been provisioned and assigned a phone number of 732-555-1007. This user and phone number are assumed to be a Secondary CTN that expects to answer calls live. In this case, the user sets the calls to be sent to their desk phone. However, they could have chosen different call forwarding options in the Locate Me frame.

If a call is received that the user wants to transfer to the Auto Attendant, they would transfer the call to the Primary CTN (in our example, 732-555-3652). In this scenario, it is **REQUIRED** that a Primary CTN be established that is always set to be direct answered by the Auto Attendant. This is accomplished by setting up the Primary CTN as a lead number of a Hunt Group, as described in the **Direct Answer Scenario** section. In our example, that number would be 732-555-3652. The Primary CTN has been set up as a Lead number of a Hunt Group that forwards calls to the CTRN.

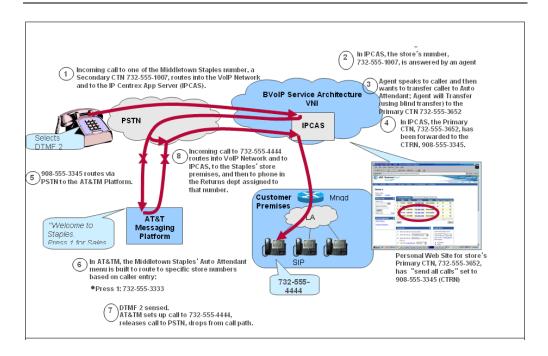


Figure 130. Live Answer Scenario – End-to-End Call Flow

≡>NOTE:

A Secondary CTN user must never set **call forwarding** to the PRIMARY CTN. They can only forward to the CTRN. They can **transfer** to the Primary CTN – since the Primary CTN has been set up in AT&T Voice DNA to always forward to the CTRN or the Primary CTN has been set up as the lead number of a Hunt Group, with no members, and CTRN as the final destination.

If the user does not want to live answer calls, they could send all the calls to the Auto Attendant, as described in the Direct Answer Scenario (i.e., Send all Calls to the CTRN) or pick a different call forwarding option in the Locate Me section.

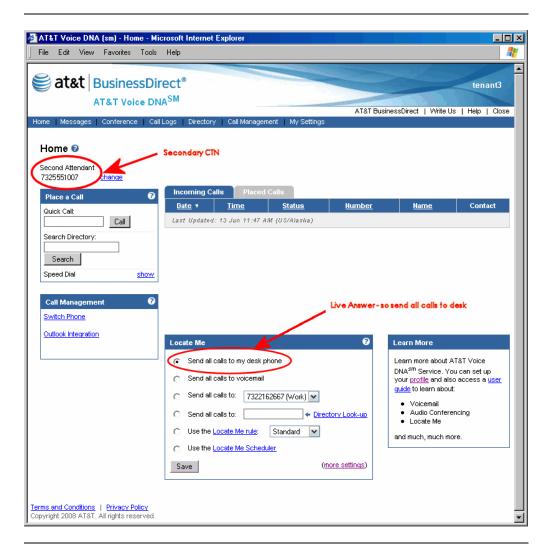


Figure 131. Live Answer Scenario

■NOTE:

The way to set up a Secondary Call Tree number in a Live answer scenario is:

- In the **Locate Me** section (home page for the Secondary CTN user), click on the **more settings** link located in the bottom right-hand corner.
- The Locate Me page will be displayed. Click on Send all calls to my desk phone and enter the CTRN in the If Busy, forward to and If No answer fields.

Configuring Auto Attendant with Hunt Groups

Hunt Groups can be used in conjunction with the Auto Attendant feature in different scenarios.

Scenario 1: Hunt Group sits behind an Auto Attendant

- Call comes into a CTN and flows through Auto Attendant prompts.
- Caller selects a prompt.
- Call flows through a Hunt group sequence.

Hunt Group Configuration:

- a. Auto Attendant could be Direct Answer or Live Answer.
- b. Transfer Destination pertaining to the Auto Attendant prompt must be the Lead Number of Hunt Group.
- c. Final Destination of Hunt Group can be any VDNA TN chosen by the customer (other than Voicemail TN)

Scenario 2: Hunt Group ends in an Auto Attendant (same as Direct Answer scenario described earlier)

Call comes into a published TN and hunts various members of a Hunt Group, ending in an Auto Attendant.

Hunt Group Configuration:

- a. Lead Number of Hunt Group must be a CTN. This CTN cannot be set up as a user in Voice DNA.
- b. Final Destination of Hunt Group must be the CTRN.

Configuring Auto Attendant with Call Distribution Queues

Scenario 1: Call Distribution Queue sits behind an Auto Attendant

- Call comes into a CTN and flows through Auto Attendant prompts.
- Caller selects a prompt.
- Caller placed in a queue until agent answers or transferred to the Final destination TN.

Call Distribution Queue Configuration:

- a. Auto Attendant could be Direct Answer or Live Answer.
- b. Transfer Destination pertaining to the Auto Attendant prompt must be the Lead Number of Queue.
- c. Final Destination of Queue can be any Voice DNA TN chosen by Customer (other than Voicemail TN).

Scenario 2: Call Distribution Queue ends in an Auto Attendant

Call comes into a published TN and caller is placed in Queue; if no agent is logged in or if all agents are busy, caller transferred to an Auto Attendant

Call Distribution Queue Configuration:

- a. Lead Number of Queue must be a CTN. This CTN cannot be set up as a **user** in Voice DNA.
- b. Final Destination of Queue must be the CTRN.

Emergency Call Log

Users whose TN is a DCN or a customer-specified 911 routing number, can access the **Emergency Call Log Report** from their Personal Web Site. For these users, the link to the **Emergency Call Log Report** is located in the **Call Management** frame. Administrators can access this report on the **Administrator Tool** under the **Reports** option.

The **Emergency Call Log Report** allows you to track calls that were placed to 911. You can view the **Emergency Call Log Report** by:

 Click the Emergency Call Log Report link located in the Call Management frame. The system responds with Emergency Call Log Report page as shown in Figure 132.

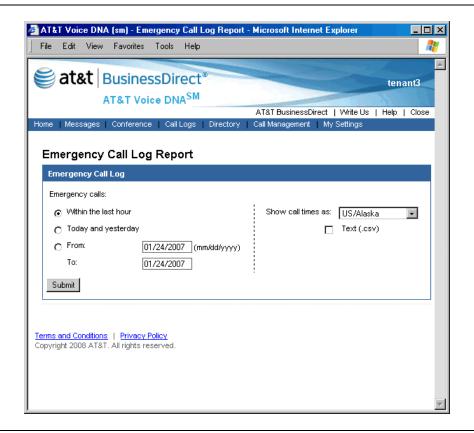


Figure 132. Emergency Call Log Report Criteria

- 2. Select the time frame of the report by selecting of the option buttons:
 - Within the last hour
 - Today and yesterday
 - From If you select From, then you need to enter the start date in the From: field and the end date in the To: field. The dates must be in the same calendar month. For example, 11/05/06 to 11/24/06.
- 3. In the **Show call times as:** field, select the time zone from the pulldown menu. This will display the call time in the appropriate time format.
- 4. If you want this report in a text file, select the **Text (.csv)** checkbox.
- 5. Click the **Submit** button to generate the report. Figure 133 shows an example of the Emergency Call Log Report.

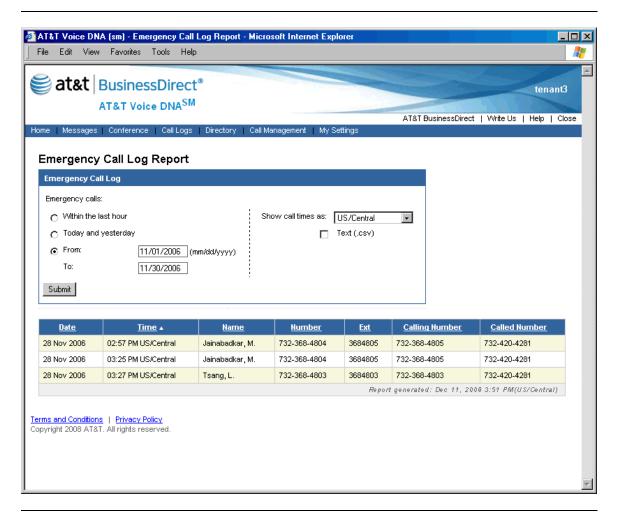


Figure 133. Emergency Call Log Report

Data Fields:

- Date Date of the emergency call.
- **Time** Time of the emergency call converted to the specified time zone.
- Name Last name and first initial of user associated with the extension.
- Number Public TN assigned to the user.
- Ext Extension associated with the call. In the case where the calling TN is a DCN, this allows you to identify the extension associated with the line key from which the call was made.
- Calling Number Calling party TN signaled on the call. This number will be either a DCN or other public TN associated with the tenant. The DCN may appear when the line from which the call is being made is an extension-only line, or is a public TN primary line on a phone that has been moved from its home tenant location.
- Called Number Emergency number to which the call was being made (911 or some other number designated in Voice DNA as an emergency call destination).

Reference Documents

Phone Configuration

The following information provides details about IP phones, the soft phone, and the IAD analog adapter used to support analog phones and faxes. For more information, see the documentation that was supplied with your phone.

■NOTE:

You may need to take additional manual steps to configure your phone to connect to the Voice DNA service when not at your corporate site. Please see the Remote Worker Customer Premises Equipment Guide.

IP Phones and Adapters

AT&T will utilize the following IP phones and soft phone for the AT&T Voice DNA.

■ Polycom IP Phones

- IP301
- IP320/321/330/331
- IP600
- IP601
- IP650
- IP560
- IP4000/6000

For more information about your Polycom phone, see the documentation that was supplied with your phone or the <u>Polycom web site</u>.

Aastra IP Phones

- 6757i (57i)
- 6757iCT (57iCT)

For more information about your Aastra phone, see the documentation that was supplied with your phone or the Aastra web site.

■ Cisco IP Phones

- 7960
- 7940G

For more information about your Cisco phone, see the documentation that was supplied with your phone or the Cisco web site.

Nortel LG Phones

- LIP-6812
- LIP-6830

For more information about the LG phones, contact your AT&T Voice DNA Administrator as they will have access to the latest guides.

■ CounterPath™ eyeBeam softphone

For more information about your CounterPath eyeBeam softphone, see the documentation that was supplied with your phone or the CounterPath Softphone Store for AT&T Voice DNASM at:

https://secure.counterpath.com/Store/Att/Default.aspx.

■NOTE:

The Eyebeam softphone should be used in its audio-only mode (i.e., no video).

- EdgeMarc EM-200EW-IAD (Integrated Access Device) Adapter for analog phones.
- Cisco IAD VG224 Cisco IAD or Cisco ATA 186 Adapters for analog phones.

For more information about your adapter, see the documentation that was supplied with your adapter.

■NOTE:

It is recommended to use a surge suppressor with the phone.

SIP Phone – Voice DNA Feature Support and Invocation

Voice DNA Features (Aastra)

Table 5. Voice DNA Features (Aastra)

		Voic	e DNA Feature Su	pport and Invocati	on
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
1.	Account Codes - Mandatory	Requires users to enter a code each time a call is made.	To place a call, dial Offnet code, dial phone number, press #, enter Account code, press # (again), then press the Dial softkey.	To place a call, dial phone number, press #, enter Account code, press # (again), then press the Call button (sometimes called the Talk/Hold key).	For outside calls only. NOTE- YOU DO NOT NEED TO DIAL AN ACCOUNT CODE FOR 911 External Transfer must be enabled by your Voice DNA Administrator For more information about Account Codes, refer to the section <u>Using</u> <u>Account Codes</u> .
2.	Account Codes - Optional	Allows users to optionally enter a code when making a call.	To place a call, press *50, followed by Account Code, press the Dial softkey, offnet code, phone number, and optional Dial softkey.	To place a call, dial phone number, press #, enter Account code, press # (again), then press the Call button (sometimes called the Talk/Hold key).	For outside calls only. NOTE- YOU DO NOT NEED TO DIAL AN ACCOUNT CODE FOR 911 External Transfer must be enabled by your Voice DNA Administrator For more information about Account Codes, refer to the section Using Account Codes.
3.	Call Forward	Lets user forward all incoming calls to another number. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls and press the Dial softkey . To turn off Call Forwarding, press *73 and press Dial softkey .	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls and press the Call button (sometimes called the Talk/Hold key). To turn off Call Forwarding, press *73 and press Call button.	

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation						
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>		
4.	Call Forward Busy	Lets user forward all incoming calls when busy to another number. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls and press the Dial softkey . To turn off Call Forwarding, press *63 and press Dial softkey .	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls and press the Call button (sometimes called the Talk/Hold key). To turn off Call Forwarding, press *63 and press Call button.			
5.	Call Forward No Answer	Lets the user redirect all calls to another telephone number when the user does not answer. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls and press the Dial softkey . To turn off Call Forwarding, press *93 and press Dial softkey .	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls and press the Call button (sometimes called the Talk/Hold key). To turn off Call Forwarding, press *93 and press Call button.			
6.	Call Hold	Lets the user place an active call on hold using their business phone. Analog Phone Sets: Lets the user place a call on hold by pressing flashhook.	During a call, press the HOLD button or soft key to put a call on hold; Press the HOLD button or PICKUP soft key to take the call off hold or select the line appearance.	During a call, press the Call button (sometimes called the Talk/Hold key) to put a call on hold. Since you have multiple lines available on the cordless handset you need to select one of the lines to take the call off hold (press the Features Button, use the down arrow or up arrow button to scroll through the list (note that the Features Button doubles as the up arrow button and the Menu Button doubles as the down arrow button)	To place a second call while the first call is still on hold, press a line key and dial the telephone number.		

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation						
Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>			
7. Call Park	Lets users place an active call in a 'hold' state, where it can then be retrieved (picked-up) by another user. For analog phones, the feature is accessible via feature codes. The parked call is retrieved using Call Pick-up).	Press the Park softkey (you may need to Press the More softkey to see the Park softkey). Enter extension to which call will be parked. Press the Dial softkey. This phone will be disconnected from the call. The other caller will hear Music on Hold. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the Call Park/Pickup softkeys.	Press F key on handset. Scroll to the Park option. Press the Select softkey. Dial the extension + the # key. e.g. 4212# Press the Xfer softkey. This phone will be disconnected from the call. The other caller will hear Music on Hold.	Call Park (*98) should always be followed by a destination. In the absence of a destination, users of Polycom phones will be able to reconnect with the calling party by pressing Resume soft key. Users of Cisco phones experience a dropped call (the calling party will also see a disconnect). Phone who originated the call park activity has 120 seconds to pick up the call. Subsequent to 120 seconds lapsing, a ringback to the parking phone will be attempted, after which the external party will then be redirected to the parking subscriber's voicemail. The timeout ringback handling for BLA/MLA will be: If the BLA appearance Parks the call, the appearance rings but the main phone does not. If the main phones Parks, the main as well as the appearance phones ring. For the MLA, regardless of who Parks the call, both the main phone as well as the appearance ring. Not supported for 7960G, or with ATA 186. The user should use the star codes (e.g., *98) or memorize the shortcuts for the soft key/menu items [e.g., press Menu hard key; press 1. Features, press 6 (for Park) or press 7 (pickup) or press 8 (pickup directed) or 9 (group)].			

Table 5. Voice DNA Features (Aastra)

		Voic	e DNA Feature Su	pport and Invocati	on
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
8.	Call Pick-up	Lets users retrieve a call that has been parked against an extension. This feature is accessible via the LCD on IP or IP-enabled digital phones or feature codes for analog phones.	Pick up a parked call by going off hook, pressing *99 followed by the extension where the call was parked and press Dial softkey. This phone will be connected with the other caller, who will no longer hear Music on Hold. If Call Park and Call Pickup are enabled in the user's COS template, you can use the Call Park/Pickup softkeys. While off-hook enter extension where the call was parked. Press the Pickup softkey (you may need to Press the More softkey). You should now be connected to the parked call, who will no longer hear the Music on Hold.	Press F key on handset. Scroll to the Pickup option. Press the Select softkey. Dial the extension + the # key. e.g. 4212# While Call Pickup softkeys are available on the Aastra Cordless handset phones users find it easier to follow the steps to use the *99 code.	Not supported for 7960G, or with ATA 186. The handset must be in the cradle for the pick-up function to work properly.

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation							
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>			
9.	Call Pickup - Directed	Lets the user pick up an incoming call ringing at an extension within the group by entering the feature code and the extension.	*53 <ext> <dial softkey=""> - The call can be picked up by anyone in the call group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call. If Call Park and Call Pickup are enabled in the user's COS template, you can use the Call Park/Pickup softkeys. While off-hook press the Drt Pickup softkey (you may need to Press the More softkey to see the Pickup softkey). Enter extension to which call is ringing. Press the Dial softkey. You should now be connected to the call.</dial></ext>	*53 <ext> <call (sometimes="" button="" called="" hold="" key="" talk="" the=""> The call can be picked up by anyone in the call group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</call></ext>				
10.	Call Pickup - Group	Lets the user pickup the most recent incoming call currently ringing at any phone in the user's call group.	*54 < Dial softkey > - The most recent call not answered in the group is picked up by the user in the group. If the call group is set to the default of none , anyone in the tenant with the call group of none can pick up the call.	*54 < Call button (sometimes called the Talk/Hold key)> The most recent call not answered in the group is picked up by the user in the group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.				

Table 5. Voice DNA Features (Aastra)

		Voic	e DNA Feature Su	pport and Invocati	on
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
11.	Call Transfer - Blind	Lets a user transfer an active call to another extension through a series of keystrokes on their business phone or using flashhook and transfer on analog sets. Reason display: Trnsfr <calling name=""></calling>	Answer an incoming call and press Xfer softkey followed by pressing the extension and then pressing Xfer softkey again. Call is released from phone.	Answer the incoming call. Press the F key on the handset. Scroll to the Xfer option. Press the Select softkey. Dial the extension. Press the Xfer softkey. Call is released from phone.	Transferring a caller to 911 is not recommended. Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly. For a Cisco 186 ATA: Answer an incoming call and perform a flashhook. Dial the TN to which you want to transfer. When the 3rd party phone starts to ring, hang up. Blind Transfer and Mandatory Billing Codes are not compatible. If you attempt to Blind Transfer a call from a device that subscribes to Mandatory Billing Codes and the transfer attempt is to a location that is not intratenant, the call drops. If you want to transfer a caller to a user's voicemail, enter the user's 10 digit number or use the *90 feature.
12.	Call Transfer -Consultative	Lets a user converse with a 3rd party and then transfer the call to that party through a series of keystrokes on their business phone. From analog sets, the user performs a flashhook, dial, converse and hangup to perform transfer. Reason display: Trnsfr <calling name=""></calling>	Answer the incoming call, then press the Xfer softkey followed by extension and hit Dial; Wait for the Transfer recipient to answer, then press the Xfer softkey.	Answer the incoming call. Press the F key on the handset. Scroll to the Xfer option. Press the Select softkey. Dial the extension. Wait for the Transfer recipient to answer, then press the Xfer softkey. Call is released from phone.	Transferring a caller to 911 is not recommended. Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly. NOTE: Since 3 lines are necessary to complete a consultative transfer, if one of the 3 available BLA lines is being used and a call comes in on that same line, a consultative transfer cannot be completed.

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation							
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>			
13.	Call Waiting – Analog Phones only	Notifies a user on an active call that there is a second incoming call. The user can switch between the two incoming calls using the Hold/Flash feature button on their business phone.	Not Available	Not Available	Analog Phones only			
14.	Call Waiting Turn off on Analog Phones	Dialing *70 turns off Call Waiting on analog phones for the current call only. When activated, a second incoming call is automatically transferred to Voicemail.	Not Available	Not Available	Analog Phones only			

Table 5. Voice DNA Features (Aastra)

	Voic	e DNA Feature Su	pport and Invocati	on
Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
15. Conferencing (via SIP phones up to 3 call legs)	Lets the user add up to two other parties to a call for support of Centralized Conferencing via the phone.	To join 2 incoming calls: - With the first incoming call on hold, answer the second call. - Press the Conf softkey. You will hear a dial tone. - Select the first call (on hold) and press the Conf softkey again. All 3 calls are in conference. To join 1 incoming and 1 outgoing call: - Place the incoming call on hold. - Place the outgoing call is answered, press the Conf softkey. You will hear a dial tone. - Select the call that is on hold and press the Conf softkey again. All 3 calls are in conference.	To join 1 incoming and 1 outgoing call: - While on an active call with the 1st party, press the "F" key on the handset. - Scroll to the Conf option. - Press the Select softkey. - Dial the extension for the 2nd party. - To connect all parties together, press the Conf softkey.	If external transfer is enabled for a user's COS settings, and a 3-way conference call is initiated from the phone, if the initiator drops out of the conference call, the other 2 parties can continue talking. If the desire is to disconnect the phone initiated conference call when the initiator drops, the external transfer setting must be disabled for that user (your Administrator can determine your external transfer status). Note that disabling external transfer will prevent the user from transferring calls outside of the Voice DNA group (a.k.a. tenant). If a AT&T Voice DNA subscriber conferences two calls and subsequently puts these callers on hold, the parties on hold will not be able to talk with each other while on hold For Example: Bob at 7325555555 (PSTN) calls you at 7325553189 (AT&T Voice DNA) and you conference in Greg at 7325553192 (AT&T Voice DNA). Since you put the callers on Hold, Bob and Greg cannot talk with each other while on hold.

Table 5. Voice DNA Features (Aastra)

		Voic	e DNA Feature Su	pport and Invocation	on
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	Comments
16.	Do Not Disturb – IP Phones	Lets the user specify 'do not ring this phone' from the personal communications portal. From the phone, the call treatments will remain in effect. Based on the specified call treatments calls will be forwarded to voicemail, if purchased; busy, if no voicemail.	Press DND softkey. The LED next to DND softkey turns red to indicate that Do Not Disturb is on. To disable the feature, press the DND softkey again (the LED will no longer be lit).		
17.	Do Not Disturb – Analog Phones	Lets the user send all calls to Voicemail	Not Available	Not Available	

Table 5. Voice DNA Features (Aastra)

	Voic	e DNA Feature Su	pport and Invocation	on
Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
18. Intercom Calling	Lets authorized callers, such as attendants, place a call to another user where the user's phone rings and the phone's microphone and speaker are automatically activated allowing the user to speak hands free with the caller. The intercom feature can also be programmed onto a speed dial button for easy access. The intercom feature is available within a specified user environment (tenant or group). Intercom can be enabled for an entire tenant. Check with your Voice DNA administrator to check whether this feature is enabled for all.	AT&T VDNA Intercom Calling: Lift the handset, press *96 followed by the extension then the Dial key. The called party's phone will ring the High Double Trill ring tone, the phone goes off- hook and the microphone is activated. *96 EXT SEND or *96 ONNET EXT SEND	AT&T VDNA Intercom Calling: not available on from the cordless handset.	The calling phone must have the Intercom COS setting enabled. The called phone does not need the Intercom COS setting enabled. If the Called phone does not support Intercom, then the phone will ring rather than go offhook. Intercom only works if the target device is a Polycom or LG phone (e.g., Cisco can call Polycom/LG but when a Polycom/LG calls Cisco the Cisco phone will ring rather than go offhook). You cannot invoke Intercom to a forked device (e.g., MLA). If forking active (e.g., 2 registered devices for the destination TN), intercom will not work. If you intercom to the BLA manager (manager/secretary scenario), the manager registration will go offhook.
19. Last Call Return	Lets the user return the last incoming call where caller id was available by dialing a feature code.	Dial *69 and press the Dial softkey to return the last incoming call.	Dial *69 and press the Call button (sometimes called the Talk/Hold key) to return the last incoming call.	

Table 5. Voice DNA Features (Aastra)

		Voic	e DNA Feature Su	pport and Invocation	I
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
20.	Last Number Redial	Lets the user automatically redial the last dialed number by pressing a single button on the business telephone, or accessible via Feature Access Code.	Press Redial button or dial *00, then Dial softkey.	Enter *00, then Call button (sometimes called the Talk/Hold key).	
21.	Message Waiting	For IP phones, a phone lamp notifies the user if a message has been left in the mailbox and a message button acts as a speed dial key to the mailbox. Analog phone users receive interrupted dial tone when a message is waiting. The customer must order voicemail in order for this feature to be enabled.	Flashing Message Waiting Indicator LED on the front of the phone. Press Voicemail softkey.		
22.	Bridged / Multiple Line Appearances	Lets the user have more than one phone line appear on their business phone set.	See Table 10	See Table 10	

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation						
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>		
23.	Speed Dialing	Corporate - 28 speed dial codes (programmed by your Voice DNA administrator) and are available at the company level. Personal - Lets the user program up to 20 entries (from their Personal Web Site) accessible either by their phone keys or through a feature code.	For codes provisioned via the Voice DNA Personal Web Site, dial the relevant speed dial code (*01 - *49) followed by Dial softkey .	For codes provisioned via the Voice DNA Personal Web Site, dial the relevant speed dial code (*01 - *49) followed by Call button (sometimes called the Talk/Hold key).	Do not enter 911 as a Speed Dial Code.		
24.	Transfer Call to VM	This feature lets user transfer a call directly to their Voicemail	While phone is ringing hit the Ignore softkey and call will take next treatment if Locate Me assigned or will forward to VM.				

Table 5. Voice DNA Features (Aastra)

		Voic	e DNA Feature Su	pport and Invocati	on
	Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>
25.	Transfer caller directly to VM	Transfer a caller directly to another employee's voicemail box	Select Xfer Enter *90xxxx (where xxxx is Employee 2's extension). Press Dial softkey or Xfer softkey	Select Xfer on the Features menu (press the Features Button, use the down arrow or up arrow button to scroll through the list (note that the Features Button doubles as the up arrow button and the Menu Button doubles as the down arrow button) Enter *90xxxx (where xxxx is Employee 2's extension). Press Call button (sometimes called the Talk/Hold key).	The COS setting Call Forward Fixed to Voicemail must be enabled. For an ATA supported device, do a flashhook on the phone connected the ATA, dial *90 <ext> # and then hang up. The call should complete the called extension's VM.</ext>
26.	Reject or Ignore softkey	SIP devices have a reject soft key that appears when an incoming call is offered to the phone. If the user presses reject to dismiss the incoming call, the caller will be sent to Voicemail if the Voice DNA user is subscribed to Voicemail. If the Voice DNA user is not subscribed to voicemail, the caller will hear ringing followed by a busy signal.	Press the Ignore soft key to dismiss the incoming call, the caller is sent to Voicemail if you are subscribed to Voicemail. If you are not subscribed to voicemail, the caller hears ringing followed by a busy signal.		

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation					
Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>		
27. Caller ID Block / Unblock	There are two modes of Caller ID Block/unblock, If the admin enables this feature for a user. Mode 1: Caller ID for that user is blocked on all calls outside the business (can be unblocked on a call by call basis using a star code). Mode 2: Caller ID is typically sent. The user may block the caller ID outside the business on a call by call basis by dialing a star code.			*67 (caller id blocking on a call-by-call basis) - turns on caller id blocking on a call (requires that the Administrator has enabled this feature for the user). *82 (caller id unblocking on a call-by-call basis) - turns caller id blocking off on a call for a user who has the automatic caller ID blocking feature enabled. If you have CBLK enabled (always block caller id, extra-tenant calls) and you make an intra-tenant call that rolls over to VM. The user experience (for the party receiving the VM) is: The call log shows the tn/name as a missed call. The VM log shows Anonymous for tn/name for the corresponding VM message.		

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation					
Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>		
28. Auto Call Back Busy	Auto Call Back Busy allows you to automatically call back a previously called busy line when the line becomes idle. Whenever the called number becomes idle, the call is connected to the original caller.	To activate: After busy signal is heard, end call (e.g., hang up or press Goodbye button). Press *66, then press the Dial softkey. Voice prompt will confirm activation.	To activate: After busy signal is heard, end call (e.g., hang up or press Release button). Press *66, then press the Call button (sometimes called the Talk/Hold key) Voice prompt will confirm	*66 to activate *86 to cancel		
		When the other line becomes free, you will receive ring tone. Answer the phone and you will be connected to called party. To de-activate:	activation. When the other line becomes free, you will receive ring tone. Answer the phone and you will be connected to called party.			
		Press *86, then press Dial softkey. Voice prompt will confirm activation.	To de-activate: Press *86, then press Call button (sometimes called the Talk/Hold key) Voice prompt will confirm activation.			

Table 5. Voice DNA Features (Aastra)

	Voice DNA Feature Support and Invocation					
Feature	Feature-Description	Aastra 57i/6757i 57iCT/6757iCT Base	Aastra 57i/6757i CT Handset	<u>Comments</u>		
29. Executive Busy Override (EBO)	This feature allows the user insert them self into an existing two-party call. This feature is often called barge in. The initiator must have the correct class of service settings in order to use these features. 3-way — The feature initiator enters an existing two-way call, turning it into a 3-way conference. Override — The feature initiator can disconnect an ongoing call on the target's phone and have themselves connected to the target. If the target is not on an active call, the caller can bypass all call forwarding and do not disturb settings on the target's phone.	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension> The EBO modes are: DigitMode 1 3-Way 4 Override Note 1: *77 EBO will work to a BLA extension. Note 2: Polycom and LG phones allow the barger to pull another line into a conference. Cisco phones do not allow this function. Note 3: Call Hold and Call Transfer are not allowed for the barger. Note 4: EBO is not supported for initiators behind an ATA. If *77 is dialed from behind an ATA with the expectation of EBO service, the initiator will now have anonymous calls rejected. *87 should be dialed to clear the condition.</extension></mode>		

Voice DNA Features (Polycom)

Table 6. Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation					
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	Comments		
Account Codes - Mandatory	Requires users to enter a code each time a call is made.	To place a call, dial Offnet code, phone number, Dial , Account Code followed by Dial .	To place a call, dial Off- net code, phone number, press the Dial button , Account Code followed by pressing Dial .	For outside calls only. NOTE- YOU DO NOT NEED TO DIAL AN ACCOUNT CODE FOR 911 External Transfer must be enabled by your Voice DNA Administrator For more information about Account Codes, refer to the section Using Account Codes.		
Account Codes - Optional	Allows users to optionally enter a code when making a call.	To place a call, press *50, followed by Account Code, Dial , offnet code, phone number, and optional Dial .	To place a call, press *50, followed by Account Code, press the Dial hard key, offnet code, phone number, and optional Dial button .	For outside calls only. NOTE- YOU DO NOT NEED TO DIAL AN ACCOUNT CODE FOR 911 External Transfer must be enabled by your Voice DNA Administrator For more information about Account Codes, refer to the section Using Account Codes.		

 Table 6.
 Voice DNA Features (Polycom)

	Voic	e DNA Feature Su	pport and Invocati	on
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>
Auto Call Back Busy	Auto Call Back Busy allows you to automatically call back a previously called busy line when the line becomes idle. Whenever the called number becomes idle, the call is connected to the original caller.	To activate: After busy signal is heard, end call (e.g., hang up or press end call button). Press *66, then press the Dial softkey. Voice prompt will confirm activation. When the other line becomes free, you will receive ring tone. Answer the phone and you will be connected to called party. To de-activate: Press *86, then press Dial soft key. Voice prompt will confirm activation.	To activate: After busy signal is heard, end call (e.g., hang up or press end call button). Press *66, then press the Dial softkey. Voice prompt will confirm activation. When the other line becomes free, you will receive ring tone. Answer the phone and you will be connected to called party. To de-activate: Press *86, then press Dial soft key. Voice prompt will confirm activation.	*86 to cancel
Bridged / Multiple Line Appearances	Lets the user have more than one phone line appear on their business phone set.	See Table 10	See Table 10	
Call Forward	Lets user forward all incoming calls to another number. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls and press Dial . To turn off Call Forwarding, press *73 and press Dial .	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls and press Dial button . To turn off Call Forwarding, press *73 and press Dial .	For Cisco/Polycom phones and the softphone, the confirmation is a fast busy tone.

 Table 6.
 Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation				
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>	
Call Forward Busy	Lets user forward all incoming calls when busy to another number. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls and press Dial . To turn off Call Forwarding, press *63 and press Dial .	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls and press Dial button . To turn off Call Forwarding, press *63 and press Dial .	For Cisco and Polycom phones, the confirmation is a fast busy tone.	
Call Forward No Answer	Lets the user redirect all calls to another telephone number when the user does not answer. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls and press Dial . To turn off Call Forwarding, press *93 and press Dial .	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls and press Dial button To turn off Call Forwarding, press *93 and press Dial .	For Cisco and Polycom phones, the confirmation is a fast busy tone.	
Call Hold	Lets the user place an active call on hold using their business phone. Analog Phone Sets: Lets the user place a call on hold by pressing flashhook.	During a call, press the HOLD button or soft key to put a call on hold; Press the RESUME soft key to take the call off hold or click on the line appearance.	During a call, press the HOLD button to put a call on hold; Press the RESUME soft key to take the call off hold.		

Table 6. Voice DNA Features (Polycom)

Voice DNA Feature Support and Invocation					
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>	
Call Park	Lets users place an active call in a 'hold' state, where it can then be retrieved (picked-up) by another user. For analog phones, the feature is accessible via feature codes. The parked call is retrieved using Call Pick-up.	Press the right More softkey. Press the Park softkey. Enter extension to which call will be parked. Press the Park softkey (again). This phone will be disconnected from the call. The other caller will hear Music on Hold. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the Call Park/Pickup softkeys.	Answer incoming call, then the Trnsfer button followed by Blind soft key and then input *98 and extension the call needs to be parked at and hit Dial . This phone will be disconnected from the call. The other caller will hear Music on Hold. While Call Pickup softkeys are available on the Polycom 320/321/330/331 phones users find it easier to follow the steps to use *98 code.	Call Park (*98) should always be followed by a destination. In the absence of a destination, users of Polycom phones will be able to reconnect with the calling party by pressing Resume soft key. Users of Cisco phones experience a dropped call (the calling party will also see a disconnect). Phone who originated the call park activity has 120 seconds to pick up the call. Subsequent to 120 seconds lapsing, a ringback to the parking phone will be attempted, after which the external party will then be redirected to the parking subscriber's voicemail. The timeout ringback handling for BLA/MLA will be: - If the BLA appearance Parks the call, the appearance rings but the main phone does not. If the main phones Parks, the main as well as the appearance phones ring. - For the MLA, regardless of who Parks the call, both the main phone as well as the appearance ring. Not supported for 7960G, or with ATA 186. The user should use the star codes (e.g., *98) or memorize the shortcuts for the soft key/menu items [e.g., press Menu hard key; press 1. Features, press 6 (for Park) or press 7 (pickup) or press 8 (pickup directed or 9 (group)]. The 320/321/330/331 interface is tedious to navigate the menu items (when selecting the menu items rather than the star codes, the user needs to be quick to enter the park/pickup extension).	

Table 6. Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation					
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>		
Call Pick-up	Lets users retrieve a call that has been parked against an extension. This feature is accessible via the LCD on IP or IP-enabled digital phones or feature codes for analog phones.	Pick up a parked call by going off hook, pressing *99 followed by the extension where the call was parked and press Send. This phone will be connected with the other caller, who will no longer hear Music on Hold. If Call Park and Call Pickup are enabled in the user's COS template, you can use the Call Park/Pickup softkeys. While off-hook, press the Pickup softkey, enter the extension of where the call was parked and press the Retrieve softkey. You should now be joined to the parked call. Pickup the handset and press the Pickup softkey. Enter the extension of where the call was parked. Press the Rtrieve softkey. This phone will be connected with the other caller, who will no longer hear the Music on Hold.	Pick up a parked call by going off hook, pressing *99 followed by the extension where the call was parked. While Call Pickup softkeys are available on the Polycom 320/321/330 /331 phones users find it easier to follow the steps to use the *99 code.	Not supported for 7960G, or with ATA 186.		
Call Pickup - Directed	Lets the user pick up an incoming call ringing at an extension within the group by entering the feature code and the extension.	*53 <ext> <send> - The call can be picked up by anyone in the call group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</send></ext>	*53 <ext> <dial> - The call can be picked up by anyone in the call group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</dial></ext>			

Table 6. Voice DNA Features (Polycom)

	Voic	e DNA Feature Su	pport and Invocati	on
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>
Call Pickup - Group	Lets the user pickup the most recent incoming call currently ringing at any phone in the user's call group.	*54 <send> - The most recent call not answered in the group is picked up by the user in the group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</send>	*54 <dial> - The most recent call not answered in the group is picked up by the user in the group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</dial>	
Call Transfer - Blind	Lets a user transfer an active call to another extension through a series of keystrokes on their business phone or using flashhook and transfer on analog sets. Reason display: Trnsfr <calling name=""></calling>	Answer an incoming call and press Trnsfer button or soft key followed by pressing Blind soft key followed by extension and press Send . Call is released from phone. A = PSTN, B, C = Polycoms A calls B, B blind transfers to C When C rings, the display shows B's id, once C answers, C's display shows A's callerid.	Answer an incoming call and press Trnsfer button or soft key followed by pressing Blind soft key followed by extension and press Dial . Call is released from phone. A = PSTN, B, C = Polycoms A calls B, B blind transfers to C When C rings, the display shows B's id, once C answers, C's display shows A's callerid.	Transferring a caller to 911 is not recommended. Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly. For a Cisco 186 ATA: Answer an incoming call and perform a flashhook. Dial the TN to which you want to transfer. When the 3rd party phone starts to ring, hang up. Blind Transfer and Mandatory Account Codes are not compatible. If you attempt to Blind Transfer a call from a device that subscribes to Mandatory Account Codes and the transfer attempt is to a location that is not intratenant, the call drops. If you want to transfer a caller to a user's voicemail, enter the user's 10 digit number or use the *90 feature.

 Table 6.
 Voice DNA Features (Polycom)

	Voic	e DNA Feature Su	pport and Invocati	on
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	Comments
Call Transfer - Consultative	Lets a user converse with a 3rd party and then transfer the call to that party through a series of keystrokes on their business phone. From analog sets, the user performs a flashhook, dial, converse and hangup to perform transfer. Reason display: Trnsfr <calling name=""></calling>	Answer the incoming call, then press the Trnsfer button or soft key followed by extension and hit Send; Wait for the Transfer recipient to answer, then select More soft key followed by Trnsfer button. A = PSTN, B, C = Polycoms A calls B, B does a consultative transfer to C and then B drops off the call. When C rings, the display shows B's id, B & C talk. After B hits transfer and drops, C's display shows A's callerid.	Answer the incoming call, then press the Trnsfer soft key followed by extension and press Dial; Wait for the Transfer recipient to answer, then press Trnsfer soft key. A = PSTN, B, C = Polycoms A calls B, B does a consultative transfer to C and then B drops off the call. When C rings, the display shows B's id, B & C talk. After B hits transfer and drops, C's display shows A's callerid.	Transferring a caller to 911 is not recommended. Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly.
Call Waiting – Analog Phones only	Notifies a user on an active call that there is a second incoming call. The user can switch between the two incoming calls using the Hold/Flash feature button on their business phone.	Not available	Not available	Analog Phones only
Call Waiting Turn off on Analog Phones	Dialing *70 turns off Call Waiting on analog phones for the current call only. When activated, a second incoming call is automatically transferred to Voicemail.	Not available	Not available	Analog Phones only

 Table 6.
 Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation						
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>			
Caller ID Block / Unblock	There are two modes of Caller ID Block/unblock, If the admin enables this feature for a user. Mode 1: Caller ID for that user is blocked on all calls outside the business (can be unblocked on a call by call basis using a star code). Mode 2: Caller ID is typically sent. The user may block the caller ID outside the business on a call by call basis by dialing a star code.			*67 (caller id blocking on a call-by-call basis) - turns on caller id blocking on a call (requires that the Administrator has enabled this feature for the user). *82 (caller id unblocking on a call-by-call basis) - turns caller id blocking off on a call for a user who has the automatic caller ID blocking feature enabled. If you have CBLK enabled (always block caller id, extra-tenant calls) and you make an intra-tenant call that rolls over to VM. The user experience (for the party receiving the VM) is: The call log shows the tn/name as a missed call. The VM log shows Anonymous for tn/name for the corresponding VM message.			

Table 6. Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation							
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	Comments				
Conferencing (via SIP phones up to 3 call legs)	Lets the user add up to two other parties to a call for support of Centralized Conferencing via the phone.	Place a call to the first party, after they answer, press the More button and then Conf button (or Confrnc soft key) and dial the second party's number. When the second party answers, press the More button and then Conf (or Confrnc soft key) again.	Place a call to the first party, after they answer, press the Conf soft key and dial the second party's number. When the second party answers, press the Conf soft key again.	If external transfer is enabled for a user's COS settings, and a 3-way conference call is initiated from the phone, if the initiator drops out of the conference call, the other 2 parties can continue talking. If the desire is to disconnect the phone initiated conference call when the initiator drops, the external transfer setting must be disabled for that user (your Administrator can determine your external transfer status). Note that disabling external transfer will prevent the user from transferring calls outside of the Voice DNA group (a.k.a. tenant). If a AT&T Voice DNA subscriber conferences two calls and subsequently puts these callers on hold, the parties on hold will not be able to talk with each other while on hold. For Example: Bob at 7325555555 (PSTN) calls you at 7325553189 (AT&T Voice DNA) and you conference in Greg at 7325553192 (AT&T Voice DNA). Since you put the callers on Hold, Bob and Greg cannot talk with each other while on hold.				
Do Not Disturb – Analog Phones	Lets the user send all calls to Voicemail	Not Available	Not Available					

 Table 6.
 Voice DNA Features (Polycom)

Voice DNA Feature Support and Invocation						
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>		
Do Not Disturb – IP Phones	Lets the user specify 'do not ring this phone' from the personal communications portal. From the phone, the call treatments will remain in effect. Based on the specified call treatments calls will be forwarded to voicemail, if purchased; busy, if no voicemail.	Press Do Not Disturb button. A flashing icon and text on the display indicates that Do Not Disturb is on. To disable the feature, press the Do Not Disturb button again.	Press the Menu key, press 1. Features and then press 2. Do Not Disturb To turn off Do Not Disturb, Press the Menu key, press 1. Features and then press 2. Do Not Disturb There is no DND hard button for the 320/321/330/331 phone.			

Table 6. Voice DNA Features (Polycom)

Voice DNA Feature Support and Invocation							
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>			
Executive Busy Override (EBO)	This feature allows the user insert them self into an existing two-party call. This feature is often called barge in. The initiator must have the correct class of service settings in order to use these features. 3-way — The feature initiator enters an existing two-way call, turning it into a 3-way conference. Override — The feature initiator can disconnect an ongoing call on the target's phone and have themselves connected to the target. If the target is not on an active call, the caller can bypass all call forwarding and do not disturb settings on the target's phone.	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension> The EBO modes are: DigitMode 1 3-Way 4 Override Note 1: *77 EBO will work to a BLA extension. Note 2: Polycom and LG phones allow the barger to pull another line into a conference. Cisco phones do not allow this function. Note 3: Call Hold and Call Transfer are not allowed for the barger. Note 4: EBO is not supported for initiators behind an ATA. If *77 is dialed from behind an ATA with the expectation of EBO service, the initiator will now have anonymous calls rejected. *87 should be dialed to clear the condition.</extension></mode>			

Table 6. Voice DNA Features (Polycom)

	Voic	e DNA Feature Su	pport and Invocati	on
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>
Intercom Calling	Lets authorized callers, such as attendants, place a call to another user where the user's phone rings and the phone's microphone and speaker are automatically activated allowing the user to speak hands free with the caller. The intercom feature can also be programmed onto a speed dial button for easy access. The intercom feature is available within a specified user environment (tenant or group). Intercom can be enabled for an entire tenant. Check with your Voice DNA administrator to check whether this feature is enabled for all.	Lift the handset, press *96 followed by the extension then the Dial key. The called party's phone will ring the High Double Trill ring tone, the phone goes off-hook and the microphone is activated. *96 EXT SEND or *96 ONNET EXT SEND Note: For the Polycom 301, you will need to pick up the handset to answer.	Lift the handset, press *96 followed by the extension then the Dial key. The called party's phone will ring the High Double Trill ring tone, the phone goes off-hook and the microphone is activated. *96 EXT SEND or *96 ONNET EXT SEND	The calling phone must have the Intercom COS setting enabled. The called phone does not need the Intercom COS setting enabled. If the Called phone does not support Intercom, then the phone will ring rather than go offhook. Intercom only works if the target device is a Polycom or LG phone (e.g., Cisco can call Polycom/LG but when a Polycom/LG calls Cisco the Cisco phone will ring rather than go offhook). You cannot invoke Intercom to a forked device (e.g., MLA). If forking active (e.g., 2 registered devices for the destination TN), intercom will not work. If you intercom to the BLA manager (manager/secretary scenario), the manager registration will go offhook.
Last Call Return	Lets the user return the last incoming call where caller id was available by dialing a	Dial *69 Dial to return the last incoming call.	Dial *69 Dial to return the last incoming call.	

Table 6. Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation							
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>				
Last Number Redial	Lets the user automatically redial the last dialed number by pressing a single button on the business telephone, or accessible via Feature Access Code.	Press Redial button or dial *00, then Send	Enter *00, then Dial					
Message Waiting	For IP phones, a phone lamp notifies the user if a message has been left in the mailbox and a message button acts as a speed dial key to the mailbox. Analog phone users receive interrupted dial tone when a message is waiting. The customer must order voicemail for this feature to be enabled.	Flashing Message Waiting Indicator LED on the front of the phone. Press Menu. Select the Messages button and the select Connect to access the VM box.	Press the Menu button, select 1. Features, select 2. Messages , and then select Connect.					

 Table 6.
 Voice DNA Features (Polycom)

	Voic	e DNA Feature Su	pport and Invocation	on
Feature	Feature-Description	Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>
Reject or Ignore softkey	SIP devices have a reject soft key that appears when an incoming call is offered to the phone. If the user presses reject to dismiss the incoming call, the caller will be sent to Voicemail if the Voice DNA user is subscribed to Voicemail. If the Voice DNA user is not subscribed to voicemail, the caller will hear ringing followed by a busy signal.	Press the Reject soft key to dismiss the incoming call, the caller is sent to Voicemail if you are subscribed to Voicemail. If you are not subscribed to voicemail, the caller hears ringing followed by a busy signal.	Press the Ignore soft key to dismiss the incoming call, the caller is sent to Voicemail if you are subscribed to Voicemail. If you are not subscribed to voicemail, the caller hears ringing followed by a busy signal.	
Speed Dialing	Corporate - 28 speed dial codes (programmed by your Voice DNA administrator) and are available at the company level. Personal - Lets the user program up to 20 entries (from their Personal Web Site) accessible either by their phone keys or through a feature code.	For codes provisioned via the Voice DNA Personal Web Site, Press New Call soft key and dial the relevant speed dial code (*01 - *49) followed by Send soft key	Press the desired Speed Dial button, or lift the handset and press the desired Speed Dial button.	Do not enter 911 as a Speed Dial Code.
Transfer Call to VM	This feature lets user transfer a call directly to their Voicemail	While phone is ringing hit the Reject button and call will take next treatment if Locate Me assigned or will forward to VM.	While phone is ringing hit the Ignore soft key and call will take next treatment if Locate Me assigned or will forward to VM.	

Table 6. Voice DNA Features (Polycom)

	Voice DNA Feature Support and Invocation							
Feature Feature-Description		Polycom 301/600/601/650/560	Polycom 320/321/330/331	<u>Comments</u>				
Transfer caller directly to VM	Transfer a caller directly to another employee's voicemail box	Transfer via IP601/IP600 - Select Transfer - Press Blind - Enter *90xxxx (where xxxx is Employee 2's extension). - Press Send Transfer via IP301 - Select More - Press Transfer. - Press Blind. - Enter *90xxxx (where xxxx is Employee 2's extension). Transfer via IP560/650 - Press Transfer or select the Trnsfer soft key. - Press Blind - Enter *90xxxx (where xxxx is Employee 2's extension).	Transfer via 320/321/330/331 Press Transfer. Press Blind. Enter *90xxxx (where xxxx is Employee 2's extension).	The COS setting Call Forward Fixed to Voicemail must be enabled. For an ATA supported device, do a flashhook on the phone connected to the ATA, dial *90 <ext> # and then hang up. The call should complete to the called extension's VM.</ext>				

Voice DNA Features (Cisco, LG, and Softphone)

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation							
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments			
Account Codes Mandatory	Requires users to enter a code each time a call is made.	Feature not available with Cisco phones.	To place a call, dial off- net code, phone number, press the [Call] softkey, enter Account Code, and press the [OK] softkey.	Feature not available with Eyebeam phone.	For outside calls only. NOTE- YOU DO NOT NEED TO DIAL AN ACCOUNT CODE FOR 911 External Transfer must be enabled by your Voice DNA Administrator For more information about Account Codes, refer to the section Using Account Codes.			
2. Account Codes - Optional	Allows users to optionally enter a code when making a call.	Feature not available with Cisco phones.	To place a call, press *50, followed by Account Code, press the [Call] softkey, offnet code, phone number, and optional [OK] softkey.	Feature not available with Eyebeam phone.	For outside calls only. NOTE- YOU DO NOT NEED TO DIAL AN ACCOUNT CODE FOR 911 External Transfer must be enabled by your Voice DNA Administrator For more information about Account Codes, refer to the section Using Account Codes.			
3. Call Forward	Lets user forward all incoming calls to another number. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls and hit Dial . To turn off Call Forwarding, press *73 and hit Dial .	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls, and press [Call] softkey. To turn off Call Forwarding, press *73 and press [Call] softkey.	When the phone is idle, press *72 followed by the telephone number to which you want to forward all the calls and hit Send . To turn off Call Forwarding, press *73 and hit Send .	For Cisco/Polycom phones and the softphone, the confirmation is a fast busy tone.			

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

		Voice DNA F	Feature Support an	d Invocation	
<u>Feature</u>	<u>Feature-</u> <u>Description</u>	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments
4. Call Forward Busy	Lets user forward all incoming calls when busy to another number. Reason display: FWD <calling name=""></calling>	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls and hit Dial . To turn off Call Forwarding, press *63 and hit Dial .	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls, and then press [Call] softkey. To turn off Call Forwarding, press *63 and press [Call] softkey.	When the phone is idle, press *62 followed by the telephone number to which you want to forward all the calls and hit Send . To turn off Call Forwarding, press *63 and hit Send .	For Cisco and Polycom phones, the confirmation is a fast busy tone.
5. Call Forward No Answer	Lets the user redirect all calls to another telephone number when the user does not answer. Reason display: FWD <calling name="">.</calling>	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls and hit Dial. To turn off Call Forwarding, press *93 and hit Dial.	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls, and press [Call] softkey. To turn off Call Forwarding, press *93 and press [Call] softkey.	When the phone is idle, press *92 followed by the telephone number to which you want to forward all the calls and hit Send . To turn off Call Forwarding, press *93 and hit Send .	For Cisco and Polycom phones, the confirmation is a fast busy tone.
6. Call Hold	Lets the user place an active call on hold using their business phone. Analog Phone Sets: Lets the user place a call on hold by pressing flashhook.	For 7960G & 7940G: During a call, press the HOLD softkey to put a call on hold; Press the Resume softkey to take the call off hold or click on the line appearance.	LG 6812: During a call, press the [Hold] Softkey. To retrieve a held call: Press the blinking Line button, or press the [Rsum] Softkey. LG 6830: During a call, press the [Hold] button. To retrieve a held call: Press the blinking Line button, [Hold] button or press the [Rsum] Softkey.		

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

		Voice DNA F	eature Support and	d Invocation	
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>
7. Call Park	Lets users place an active call in a 'hold' state, where it can then be retrieved (picked-up) by another user. For analog phones, the feature is accessible via feature codes. The parked call is retrieved using Call Pick-up.	Answer incoming call, press the More softkey. Press the BIndXfer softkey. enter <extension be="" parked="" to="">, then press Dial. This phone will be disconnected from the call. The other caller will hear Music on Hold. With CISCO phones only, when a user parks a call to another extension, and then tries to park another call to the same extension, the second call is dropped.</extension>	Press the right More softkey. Press the Park softkey. Enter extension to which call will be parked. Press the Park softkey (again). This phone will be disconnected from the call. The other caller will hear Music on Hold. or Press the [Park] Softkey during a conversation. Enter an extension number and press the [OK] Softkey.	*98	Call Park (*98) should always be followed by a destination. In the absence of a destination, users of Polycom phones will be able to reconnect with the calling party by pressing Resume soft key. Users of Cisco/LG phones experience a dropped call (the calling party will also see a disconnect). Phone who originated the call park activity has 120 seconds to pick up the call. Subsequent to 120 seconds lapsing, a ringback to the parking phone will be attempted, after which the external party will then be redirected to the parking subscriber's voicemail. The timeout ringback handling for BLA/MLA will be: — If the BLA appearance Parks the call, the appearance rings but the main phone does not. If the main phones Parks, the main as well as the appearance phones ring. — For the MLA, regardless of who Parks the call, both the main phone as well as the appearance ring. Not supported for 7960G, or with ATA 186. The user should use the star codes (e.g., *98) or memorize the shortcuts for the soft key/menu items [e.g., press Menu hard key; press 1. Features, press 6 (for Park) or press 7 (pickup) or press 8 (pickup directed) or 9 (group)].

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation							
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>			
8. Call Pick-up	Lets users retrieve a call that has been parked against an extension. This feature is accessible via the LCD on IP or IP- enabled digital phones or feature codes for analog phones.	Pick up a parked call by going off hook, pressing *99 followed by the extension where the call was parked and hit Dial.	Using Softkey: To pick-up a Parked Call: Press the → to locate the [Pick] Softkey. Press the [Pick] Softkey, and enter an extension number, then press the [OK] Softkey. This phone will be connected with the other caller, who will no longer hear Music on Hold. Using Star Codes: Pickup handset. Enter *99 <extension call="" parked="" the="" was="" where="">, then press the Call softkey. This phone will be connected with the other caller, who will no longer hear Music on Hold.</extension>	Enter *99 <extension for="" softphone="" the="">, then press the green connect icon. This phone will be connected with the other caller, who will no longer hear Music on Hold.</extension>	Not supported for 7960G, or with ATA 186.			
9. Call Pickup - Directed	Lets the user pick up an incoming call ringing at an extension within the group by entering the feature code and the extension.	*53 <ext> <dial> - The call can be picked up by anyone in the call group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</dial></ext>	To pick up a call ringing at another extension: Press > to locate the [DPic] Softkey. Press the [DPic] Softkey, and enter the extension number, then press the [OK] Softkey.	*53 <ext> <send> - The call can be picked up by anyone in the call group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</send></ext>				

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation							
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments			
10. Call Pickup - Group	Lets the user pickup the most recent incoming call currently ringing at any phone in the user's call group.	*54 <dial> - The most recent call not answered in the group is picked up by the user in the group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</dial>	Press the → to locate the [GPic] Softkey. Press the [GPic] Softkey.	*54 <send> - The most recent call not answered in the group is picked up by the user in the group. If the call group is set to the default of none, anyone in the tenant with the call group of none can pick up the call.</send>				
11. Call Transfer - Blind	Lets a user transfer an active call to another extension through a series of keystrokes on their business phone or using flashhook and transfer on analog sets. Reason display: Trnsfr <calling name=""></calling>	Answer an incoming call and press the more soft key followed by BlndXfr soft key followed by extension, hit dial. Call is released from phone. Blind transfer to an invalid extension results in the caller being dropped.	To transfer an active call: Press [BXfr] Softkey. Dial the number you wish to receive the transfer. Hang up to complete the transfer. Blind transfer to an invalid extension results in the caller being dropped.	Establish call on line 1 and press the xfer button. Enter the transfer to TN. Either press enter or the xfer button again.	Transferring a caller to 911 is not recommended. Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly. For a Cisco 186 ATA: Answer an incoming call and perform a flashhook. Dial the TN to which you want to transfer. When the 3rd party phone starts to ring, hang up. Blind Transfer and Mandatory Account Codes are not compatible. If you attempt to Blind Transfer a call from a device that subscribes to Mandatory Account Codes and the transfer attempt is to a location that is not intratenant, the call drops.			

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

		Voice DNA F	eature Support an	d Invocation	
<u>Feature</u>	<u>Feature-</u> <u>Description</u>	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>
12. Call Transfer - Consultative	Lets a user converse with a 3rd party and then transfer the call to that party through a series of keystrokes on their business phone. From analog sets, the user performs a flashhook, dial, converse and hang-up to perform transfer. Reason display: Trnsfr <calling name=""></calling>	Answer an incoming call, press the more soft key followed by Trnsfer soft key followed by extension and hit dial. Wait for the transfer recipient to answer and then press Trnsfer again	To transfer an active call: Press the [Trns] Softkey. Dial the number you wish to receive the transfer. You may await answer and then hangup, or just hang-up (or press the [Trns] Softkey again) to complete the transfer. To cancel the Transfer: Press the [Rsum] softkey or the [EndC] softkey. If you press the [Endc] softkey the phone will ring and you may enter a (new) phone number to transfer the call OR you may press the [Rsum] softkey to speak with the caller.	Establish call on line 1. Put line 1 on hold. Click on line 2 and establish the second call. Consult with second the TN. Put line 2 on hold. Go back to line 1. Press the xfer button and the press line 2. or Establish call on line 1. Put line 1 on hold. Click on line 2 and establish the second call. Consult with the second TN. Press the xfer button and then press line 1.	Transferring a caller to 911 is not recommended. Instead of using Call Transfer, the Voice DNA user should advise their caller to hang up and call 911 directly.
13. Call Waiting – Analog Phones only	Notifies a user on an active call that there is a second incoming call. The user can switch between the two incoming calls using the Hold/Flash feature button on their business phone.	Not Available	Not Available	Not Available	Analog Phones only

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation								
<u>Feature</u>	<u>Feature-</u> <u>Description</u>	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>				
14. Call Waiting Turn off on Analog Phones	Dialing *70 turns off Call Waiting on analog phones for the current call only. When activated, a second incoming call is automatically transferred to Voicemail.	Not Available	Not Available	Not Available	Analog Phones only				

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

		Voice DNA F	eature Support and	Invocation	
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>
15. Conferencing (via SIP phones up to 3 call legs)	Lets the user add up to two other parties to a call for support of Centralized Conferencing via the phone.	Place a call to the first party, press Confrn and dial the second party's number. When the second party answers, press the Join or Confrn soft key. Cannot highlight participant's name to drop a party. On 7960, pressing end call drops the originating party and leave participants on the line.	To initiate a conference between the current party, you, and another party: Press the [Conf] Softkey [6830 could instead press the CONF button], and dial the number you want to join in the conference. When the party answers, press the [Join] Softkey to establish the conference. NOTE: If the new party does not answer, press the [Endc] Softkey and then the [Rsum] Softkey to return to the original party. If either of the called parties hangs up, the call will remain connected.		If external transfer is enabled for a user's COS settings, and a 3-way conference call is initiated from the phone, if the initiator drops out of the conference call, the other 2 parties can continue talking. If the desire is to disconnect the phone initiated conference call when the initiator drops, the external transfer setting must be disabled for that user (your Administrator can determine your external transfer status). Note that disabling external transfer will prevent the user from transferring calls outside of the Voice DNA group (a.k.a tenant). If a AT&T Voice DNA subscriber conferences two calls and subsequently puts these callers on hold, the parties on hold will not be able to talk with each other while on hold For Example: Bob at 7325555555 (PSTN) calls you at 73255553189 (AT&T Voice DNA) and you conference in Greg at 7325553192 (AT&T Voice DNA). Since you put the callers on Hold, Bob and Greg cannot talk with each other while on hold.

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation								
<u>Feature</u>	<u>Feature-</u> <u>Description</u>	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>				
16. Do Not Disturb – IP Phones	Lets the user specify 'do not ring this phone' from the personal communications portal. From the phone, the call treatments will remain in effect. Based on the specified call treatments calls will be forwarded to voicemail, if purchased; busy, if no voicemail.	Press the Menu button, then Call Preferences, and then hit select softkey. Highlight the Do Not Disturb entry and hit Yes softkey	To activate DND: Press the [DND] softkey [6830 could instead press the DND button] when activated Do not disturb is displayed on the LCD [6830 the button LED will illuminate red]. To deactivate DND: Press the [DND] softkey again [6830 could instead press the DND button].						
17. Do Not Disturb – Analog Phones	Lets the user send all calls to Voicemail	Not Available	Not Available	Not Available					

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation								
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>				
18. Intercom Calling	Lets authorized callers, such as attendants, place a call to another user where the user's phone rings and the phone's microphone and speaker are automatically activated allowing the user to speak hands free with the caller. The intercom feature can also be programmed onto a speed dial button for easy access. The intercom feature is available within a specified user environment (tenant or group). Intercom can be enabled for an entire tenant. Check with your Voice DNA administrator to check whether this feature is enabled for all.	Lift the handset, press *96 followed by the extension and the Send key. The called party's phone will ring. For Cisco phones, the user needs to pick up the handset to answer.	To access the intercom: Press the assigned Intercom button. Dial an extension number and press the [ok] Softkey. LG phones will double ring it's Type 1 ring followed by the phone speaker being automatically activated.		The calling phone must have the Intercom COS setting enabled. The called phone does not need the Intercom COS setting enabled. If the Called phone does not support Intercom, then the phone will ring rather than go offhook. Intercom only works if the target device is a Polycom or LG phone (e.g., Cisco can cal Polycom/LG but when a Polycom/LG calls Cisco the Cisco phone will ring rather than go offhook). You cannot invoke Intercom to a forked device (e.g., MLA). If forking is active (e.g., 2 registered devices for the destination TN), intercom will not work. If you intercom to the BLA manager (manager/secretary scenario), the manager registration will go offhook.				
19. Last Call Return	Lets the user return the last incoming call where caller id was available by dialing a feature code.	Dial *69 dial to return the last incoming call.	Dial *69 Send to return the last incoming call.	Dial *69 to return the last incoming call.					

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation								
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments				
20. Last Number Redial	Lets the user automatically redial the last dialed number by pressing a single button on the business telephone, or accessible via Feature Access Code.	Press Redial soft key or dial *00, then dial	Press [Redi] Softkey. Pressing the [Redi] Softkey activates the speakerphone and dials the last dialed number. Lift the handset prior to pressing the [Redi] Softkey to connect the call to the handset rather than the Speakerphone.	Press Redial soft key or dial *00					
21. Message Waiting	For IP phones, a phone lamp notifies the user if a message has been left in a mailbox and a message button acts as a speed dial key to the mailbox. Analog phone users receive interrupted dial tone when a message is waiting. The customer must order voicemail in order for this feature to be enabled.	Red Message Waiting Indicator light on handset. Press the Message soft key or the Envelope key to retrieve messages	Press [MSG] softkey [6830 could instead press the MSG button], when the Message Waiting Indicator LED indicates you have a message and follow the voice instructions.						

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation								
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>				
22. Bridged / Multiple Line Appearances	Lets the user have more than one phone line appear on their business phone set.	See Table 10	See Table 10						
23. Speed Dialing	Corporate - 28 speed dial codes (programmed by your Voice DNA administrator) and are available at the company level. Personal - Lets the user program up to 20 entries (from their Personal Web Site) accessible either by their phone keys or through a feature code.	Speed Dial codes can be enabled in the Voice DNA Personal Web Site or directly on the phone. If provisioned directly on phone via Menu/Directories/S peed dial, they can be utilized separately than what has been provisioned in the Personal Web Site. When entering on C7960 they will be presented on the phone as phone lines (up to 5 entries). Hit button entry on C7960 and select the relevant Speed dial entry in log and press Dial. To utilize Voice DNA speed dial entries enter *01 - *49 and hit the Dial softkey.	Press the desired Speed Dial button, or lift the handset and press the desired Speed Dial button.		Do not enter 911 as a Speed Dial Code.				

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation								
<u>Feature</u>	<u>Feature-</u> <u>Description</u>	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments				
24. Transfer Call to VM	This feature lets user transfer a call directly to their mailbox	For C7960 you must utilize the Send all calls to VM setting on the Voice DNA Personal Web Site if COS setting CFFV option is set or you can enter the VM number into the CFW'd field.	While phone is ringing press the [Refu] softkey and the call will forward to VM.						
25. Transfer caller directly to VM	Transfer a caller directly to another employee's mail box	 Select More Press Blndxfer Enter *90xxxx (where xxxx is Employee 2's extension). Press Dial. 	Press [BXfr] softkey, enter *90xxxx (where xxxx is Employee 2's extension) and press [Call] softkey.		The COS setting Call Forward Fixed to Voicemail must be enabled. For an ATA supported device, do a flashhook on the phone connected to the ATA, dial *90 <ext> # and then hang up. The call should complete to the called extension's VM.</ext>				
26. Reject softkey	SIP devices have a reject soft key that appears when an incoming call is offered to the phone. If the user presses reject to dismiss the incoming call, the caller will be sent to the mailbox if the Voice DNA user is subscribed to Voicemail. If the Voice DNA user is not subscribed to voicemail, the caller will hear ringing followed by a busy signal.	Press reject to dismiss the incoming call, the caller is sent to the mailbox if you are subscribed to Voicemail. If you are not subscribed to voicemail, the caller hears ringing followed by a busy signal.	While phone is ringing press the [Refu] softkey and call will forward to VM.	Press reject to dismiss the incoming call, the caller is sent to the mailbox if you are subscribed to Voicemail. If you are not subscribed to voicemail, the caller hears ringing followed by a busy signal					

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

		Voice DNA Fea	Voice DNA Feature Support and Invocation								
<u>Feature</u>	<u>Feature-</u> <u>Description</u>	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments						
27. Caller ID Block / Unblock	There are two modes of Caller ID Block/unblock. If the admin enables this feature for a user. Mode 1: Caller ID for that user is blocked on all calls outside the business (can be unblocked on a call by call basis using a star code). Mode 2: Caller ID				*67 (caller id blocking on a call-by-call basis) - turns on caller id blocking on a call (requires that the Administrator has enabled this feature for the user). *82 (caller id unblocking on a call-by-call basis) - turns calle id blocking off on a call for a user who has the automatic caller ID blocking feature enabled. If you have CBLK enabled (always block caller id, extratenant calls) and you make ar intra-tenant call that rolls over						
	is typically sent. The user may block the caller ID outside the				to VM. The user experience (for the party receiving the VM) is:						
	business on a call by call basis by dialing a star code.				The call log shows the tn/name as a missed cal The VM log shows Anonymous for tn/name for the corresponding VM						

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

	Voice DNA Feature Support and Invocation							
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	Comments			
28. Auto Call Back Busy		To activate: After busy signal is heard, end call (e.g., hang up or press end call button). Press *66, then press the Dial softkey. Voice prompt will confirm activation. When the other line becomes free, you will receive ring tone. Answer the phone and you will be connected to called party. To de-activate: Press *86, then press Dial soft key.	To activate: After busy signal is heard, end call (e.g., hang up or press end call button). Press *66, then press the Call softkey. Voice prompt will confirm activation. When the other line becomes free, you will receive ring tone. Answer the phone and you will be connected to called party. To de-activate: Press *86, then press Call soft key. Voice prompt will confirm activation.	To activate: After busy signal is heard, press *66, then click the green connect icon. You do not have to hang up from the previous call. Voice prompt will confirm activation and the Hold light is activated. When other line becomes free, you will receive a ring tone and get a pop-up window. Click the green connect icon on softphone image or the answer button on the pop-up window. To de-activate: Press *86, then click the green connect icon. Voice prompt will confirm activation.	*66 to activate *86 to cancel			

Table 7. Voice DNA Features (Cisco, LG, and Softphone)

		Voice DNA	Feature Support	and Invocation	
<u>Feature</u>	Feature- Description	Cisco 7960G/7940G	LG 6812 LG 6830	CounterPath Eyebeam	<u>Comments</u>
29. Executive Busy Override (EBO)	This feature allows the user insert them self	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension></extension></mode>	*77 <mode digit=""> <extension> The EBO modes are:</extension></mode>
(LBO)	into an existing two-party call.				DigitMode
	This feature is often called barge				1 3-Way 4 Override
	in.3-way — The feature initiator enters an existing				The initiator must have the correct class of service settings in order to use these features.
	two-way call, turning it into a 3- way conference.				Note 1: *77 EBO will work to a BLA extension.
	Override — The feature initiator can disconnect an ongoing call on the target's phone				Note 2: Polycom and LG phones allow the barger to pull another line into a conference. Cisco phones do not allow this function.
	and have themselves connected to the				Note 3: Call Hold and Call Transfer are not allowed for the barger.
	target. If the target is not on an active call, the caller can by-pass				Note 4: EBO is not supported for initiators behind an ATA. If *77 is dialed from behind an
	all call forwarding and do not disturb settings on the				ATA with the expectation of EBO service, the initiator will now have anonymous calls rejected. *87 should be dialed
	target's phone.				to clear the condition.

Phone Matrix

The following table identifies relevant Voice DNA Line Features that are supported by the various IP phones.

Table Legend Key:

 $\sqrt{\ }$ = Supported or committed to support

X = Not Supported or planned

Ind = The feature is phone-independent

Table 8. Phor	ne Feature Matrix	(
Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Auto Call Back Busy	Auto Call Back Busy (also known as Call Completion on Busy Subscriber or Repeat Dialing) is used when a caller reaches a busy number. When invoked the system will continue to check the busy number and when the line is free, the system rings the caller back and completes the call.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Invoked by dialing *66 after terminating the busy call. To cancel, dial *86 This capability must be enabled on the user's COS settings.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Billing (Account) Codes - Mandatory	Requires users to enter a code each time an outside call is made. This code is then output in a call detail report (accessible by the admin) allowing the call information to be used for other reason, e.g., billing codes can be used to identify all calls to a specific client so the client can be billed accordingly.	√	→	X	√	V	√	x	SIP-B feature
Billing (Account) Codes - Optional	Users can optionally enter a code when making a call. This code is then output in a call detail report accessible by the Administrator.	√	V	x	√	√	V	x	SIP-B feature. Invoked via *50.
Bridged Line Appearance (BLA)	Allows multiple extensions to appear on the same handset. As calls arrive on those extensions, the caller information is displayed on the attendant's LCD. BLA includes presence information.	1	1	х	1	X See note.	1	X	SIP-B feature A Polycom 4000/6000 can be the monitored device but it cannot be the monitoring device because it only supports a single line appearance.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Bridged Line Appearance (BLA) – Cross Location	An extension can be configured as a line appearance (BLA or MLA) on a phone whose owner is provisioned at ANY VDNA location within the same tenant.	X	X	X	1	X	X	X	For calls originating from the BLA line on a given device, the BLA telephone number assigned to that line will be used as the outbound caller id. For this reason, BLA lines bridged across locations (Polycom phones only) should not be used for making calls to 911. Always use the Primary phone extension (for exampe, line key 1) when calling 911.
Call Distribution Queue	The Administrator can set up a queue for answering calls. Agents log in to the queue and mark themselves as available/unavaila ble to take calls.	7	√	x	1	V	1	x	Use the phone softkeys for this feature. SIP-B feature. Note that any device can be the final Destination" for an ACD queue (Polycom, Cisco, Analog phone). Polycom: user must indicate available by pressing softkey after every call. LG: automatically logs user back into the queue after 10 seconds.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Call Forward - Busy	The user can automatically redirect all calls to another telephone when the user's extension is busy.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Independent of phone if invoked via AT&T Voice DNA Personal Web Site. Can also be invoked via *62 from all phones. *63 to deactivate
Call Forward - No Answer	The user can redirect all calls to another telephone number when the user does not answer. The user can control the duration of the no answer timer based on the phone that the call is forwarded.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Independent of phone if invoked via AT&T Voice DNA Personal Web Site. Can also be invoked via *92 from all phones. *93 to deactivate
Call Forward - Unconditional	The user can forward calls to an alternate number, either a business extension or an external number.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Independent of phone if invoked via user portal. Can also be invoked via *72 from all phones.
Call Groups	Companies can create groups of users for specific features. Group dependent features include group call pickup, directed call pickup and intercom.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Call Hold	The user can place an active call on hold using their business phone.	√	✓				V	1	Not a function specifically of the IAD. If the analog device behind the IAD supports a hold button, nothing in the network should prevent the feature from working. For the ATA, can use the phone's switchhook flash (switchhook flash is NOT currently supported on the VG224). From an analog phone: If you use the Hold button (on the phone) to place a caller on hold, the caller will not hear Music on Hold (provided Music on Hold is active).
									If you use the flash hook (on the telephone) to place a caller on hold, the caller will hear Music on Hold (provided Music on Hold is active). Note: If both caller on an intra-tenant call put each other on hold, neither caller will hear Music on Hold.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Call Logs	Using the LCD display of the SIP phones, the user can access the missed, outgoing and incoming call logs.	√	√	√	√	√	V	√	Also available via AT&T Voice DNA Personal Web Site.
Call Park	A user can park (put a call on hold) at one telephone set and pick up (continue) the conversation from any other telephone set.	√	√	1	√	V	√	√	Invoked via *98. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on device.
Call Pickup	The user can retrieve calls that have been parked against an extension.	1	√	1	1	1	1	1	Invoked via *99. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on device.

•	V	1	√	1	√	√	Invoked via *53. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS
							template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on
√	√ ·	√ ·	√	V	√	√	Invoked via *54. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key
а	at						

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes		
Call Reason Display	For Intra-tenant calls, a SIP phone display will indicate information about the call. Specifically:	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Not supported on IAD/analog phones. This capability must be enabled on the user's COS settings.		
	Hunt Group incoming call display: HG: <calling number=""></calling>										
	Call Distribution Queue incoming call display: ACD: <calling number=""></calling>										
	Call Forward display: FWD: <calling number=""></calling>										
	Blind or Consultative transfer (called phone): From: <calling number=""></calling>										
	Blind or Consultative transfer(callers phone): To: <called number=""></called>										
	Cisco phones display both the From: and To: field.										
	Calls dialed from the voicemail platform are NOT classified as an intra-tenant calls.										

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Call Restriction	An administrator can allow/restrict dialing of the following calls: long distance calls, information, international, international operator, local, local operator, long distance operator, premium service, and toll free.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	
Call Transfer - Blind	Lets a user transfer an active call to another extension through a phone soft key	7	√	√	√	√	√	√	Supported by the IAD only if the analog phone has a transfer key. For the ATA, works via the switchhook flash (switchhook flash is NOT currently supported on the VG224). SIP enabled softkey or hard key depending on device.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Call Transfer - Consultative	A user can converse with a 3rd party and then transfer the call to that party through a phone soft key.	✓	\	√	7		~	√	Supported by the IAD only if the analog phone has a transfer key. For the ATA, works via the switchhook flash (switchhook flash is NOT currently supported on the VG224). Note: On Aastra phones, since 3 lines are necessary to complete a consultative transfer, if one of the 3 available BLA lines is being used and a call comes in on that same line, a consultative transfer cannot be completed.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Call Waiting	Notifies a user on an active call that there is a second incoming call. The user can switch between the two incoming calls.	√	✓	1	1		√	1	From the phone enabled/disabled using *70 for the current call only. Call Waiting is enabled on the user's COS settings by default. The Admin can disable Call Waiting on the users' COS setting if a user needs Call Waiting disabled for all calls. Switchhook flash is required for Call Waiting and is NOT currently supported on the VG224. For the Softphone, 2 beeps are heard on the headset, a window displays on your PC with answer, ignore, and the caller's caller ID. In the softphone image window, Ignore appears with caller's caller ID. To Disable Call Waiting on the Softphone, enter *70 immediately before the number you are dialing and press the green connect icon. (e.g., off-net: *708005551234 <connect>, onnet: *701234 <connect>)</connect></connect>
Caller ID	Displays the phone number of the calling party to the user.	$\sqrt{}$	√	√	√	√	1	1	Supported by the IAD only if the analog phone has caller ID capabilities.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Caller ID block/unblock	There are 2 modes of Caller ID Block/unblock. Mode 1: The admin enables this feature for a user. Caller ID for that user is blocked on all calls outside the business (can unblocked call on a call-by-call using star code). Mode 2: Caller ID is typically sent. The user may block Caller ID outside the business on a call-by-call basis by dialing a star code.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	*67 (caller id blocking on a call-by-call basis) - turns on caller id blocking on a call (requires that the Administrator has enabled this feature for the user). *82 (caller id unblocking on a call-by-call basis) - turns caller id blocking off on a ca for a user who has the automatic caller ID blocking feature enabled. *67 functionality must be enabled on the user's COS settings
Caller ID Presentation Contact	A user, or corporate contact data, can provide the name associated with a phone number in the call logs and caller ID display.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Applies to Intra Tenant calls only.
Caller Name	The network- provided name (from the contact database) is displayed to the user's phone.	V	V	V	1	√	√	1	Supported by the IAD only if the analog phone has caller ID capabilities.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Called Party Display on Calling phone	The name of the Called number via the AT&T Voice DNA Personal Web Site input (User Contact, Company Contact). This name will be displayed on the Calling Party phone.	√	√	х	√	1	1	х	
Click to Call	The user can click on an entry in a call log or directory, or click a dial button to call a selected phone number.	√	√	√	√	٧	V	Ind	This feature is invoked via the AT&T Voice DNA Personal Web Site.
Phone Log: Select & Call	The user can select an entry displayed on the phone (e.g., in Call Logs) to call a selected phone number.	√	√	V	√	٧	V	√	Supported by the IAD only if the analog phone supports this feature.
Conferencing – On-Demand Conference (Click to Conference)	Setup for this feature is via the AT&T Voice DNA Personal Web Site. A user can perform Centralized Conferencing for up to 10 callers.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	10 callers including the call initiator.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Conferencing (via SIP phones up to 3 call legs)	A user can perform Conferencing via the phone for up to 3 callers.	√	X - See Notes	V	√	√	√	1	3 callers including the call initiator. Only supported on the ATA if the analog device has a conferencing capability. Will be available in a future release. for the Aastra 57i/6757iCT Handset
Configurable Speed Dial Keys	Ability for the end user to assign speed dial destinations to unused hard function keys (normally used as line keys).	√	√	V	√	x	√	х	With Polycom multi- line phones, speed dials appear after the line registrations. Does not interfere with BLA or MLA. Any new BLA or MLA appearances will be assigned first, followed by the end user set speed dials. End user sets the speed dial keys via the phone menu.
Direct Inward Dialing (DID)	A caller can access another user's extension directly, without going through an attendant.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Distinctive Ringing	Lets users hear different types of rings depending on internal or external callers. Rings are endpoint specific and distinctive ring is enabled through class of service. Only two types of rings are supported (for the inside or Outside calls): 1. Other tenant or PSTN, and 2. Internal Ring: another user within a user group	→		√				X	For Cisco VG224. there is no distinctive ring, both onnet & offnet calls are the same rings. Polycom phones allow you to set your ring tone from the phone menu. However, in order for the user to select the ring tone to be used, the Distinctive Ringing checkbox (located on the Ringer Settings page of the Personal Web Site for the user) must be unchecked (disabled). For MLA: The distinctive ring must be disabled for the employee who owns the TN of the line appearance. For example, for a user's line appearance on my phone to silent ring the Distinctive Ringing checkbox (located on the Ringer Settings page of the Personal Web Site for the user) unchecked

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Direct Outward Dialing (DOD)	A caller can place a call, without going through an attendant, to a PSTN number, by dialing an external access code (such as 9) as defined by the specified dialing plan.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	
Do Not Disturb	A user can specify do not ring this phone from the AT&T Voice DNA Personal Web Site. From the phone, the call treatments will remain in effect. Based on the specified call treatments, calls will be sent to the mailbox, if purchased; or busy, if there is no mailbox.	√	✓	√	√	√	√	√	Polycom - hard key. LG 630 – hard key Cisco - soft key (menu driven). LG 6812 – soft key (menu driven) Supported by IAD if phone device has this capability.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Executive Busy Override (EBO)	Executive Busy Override (EBO), also known as Barge-in, allows the user to insert themselves into an existing two-party call. There are two ways (modes) a user can use EBO: 3-way: the user	Ind	Ind	Ind	Ind	Ind	Ind	Ind	*77 <mode digit=""><extension> Invoked via *77 and the EBO mode number. The EBO modes are: Digit Mode 1 3-Way 4 Override This capability mus be enabled on the</extension></mode>
	enters an existing two-way call, turning it into a 3-way conference. Override: the user can disconnect an ongoing call on the target extension								EBO exemption is a another COS setting which allows the Administrator to mark an extension as exempt from barge-in.
	and have themselves connected to the target extension. If the target extension is idle, the EBO user will by-pass all Call Forwarding and Do Not Disturb settings to connect to the target extension.								EBO is not supported for initiators behind an ATA. If *77 is dialed from behind an ATA with the expectation of EBO service, the initiator will now have anonymous calls rejected. *87 should be dialed to clear the condition.
	Extensions can be marked as exempt from EBO so that NO barge-in is allowed.								- San and Gordano

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Hunt Groups	The Administrator can enter a series of numbers to route calls when the previous number dialed does not answer.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	
Intercept Announcements	Intercept Announcements are system wide announcements provided to the user as they use service features. Some announcements are informational such as "Do not disturb activated" and some provide guidance for using a feature such as "We are sorry. Call pick up could not be completed because there is no call parked to the number provided."	Ind	Ind	Ind	Ind	Ind	Ind	Ind	

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Intercom Calling	Allows the phones to act as an intercom. The user enters a feature code and extension to invoke the service. The called extension automatically answers and a voice path is established between the two phones.	→		X		√		X	Invoked via *96. SIP-B feature. If invoked to a non-SIP-B device, the terminating phone rings (not forced of hook, which is the intent of intercom). The calling phone must have the Intercom COS setting enabled. The called phone does not need the Intercom COS setting enabled. If the Called phone does not support Intercom, then the phone will ring rather than go offhook. Intercom only work if the target device is a Polycom or LG phone (e.g., Cisco can call Polycom/LG but when a Polycom/LG calls Cisco the Cisco phone will ring rather than go offhook). You cannot invoke Intercom to a forke device (e.g., MLA). If forking is active (e.g., 2 registered devices for the destination TN), intercom will not work.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Intercom Calling									If you intercom to the BLA manager (manager/secretary scenario), the manager registration will go offhook.
Last Call Return	A user can return the last incoming call where caller ID is available by dialing a feature code.	√	√	√	√	1	V	1	Invoked via *69
Last Number Redial	A user can automatically redial the last dialed number	٨	V	√	1	1	√	√	Invoked via *00 or redial button on the phone
Message Waiting Indicator (MWI)	A phone lamp notifies the user that a message has been left in the mailbox. A message button acts as a speed dial key to voicemail. Analog phone users receive interrupted dial tone when a message is waiting. A user must order voicemail (via the premium package).	√	√	√	√	√	V	√	For IAD supported devices - lamp if the phone supports an MWI lamp, otherwise interrupted dialtone. For the Polycom 4000/6000, the MW will be heard as a interrupted dial tone and the envelope icon on the display (the phone lights do not flash). You can clear the MWI indication from the phone by pressing the clear button.
Message Waiting Indicator (MWI) for MLA or BLA appearances		V	1	1	√	х	Х	х	Not supported for analog devices. The Polycom 4000/6000 and Eyebeam are single line phones.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Multiple Line Appearances – Basic	A user can have more than one phone line appear on the phone set OF DIFFERENT EXTENSIONS/DID .	7	√	√	√	x	√	x	Not supported for analog devices. The Polycom 4000/6000 and Eyebeam are single line phones. Eyebeam automatically supports 6 lines of the same telephone number.
Multiple Line Appearances – Repetitions	A user can have more than one phone line appear on the phone set OF THE SAME EXTENSION/DID.	*	√	x	~	X	√	\	Not supported for analog devices. The Polycom 4000 /6000 and Eyebean are single line phones. Eyebeam automatically supports 6 lines of the same telephone number. 7960/7940 - Cisco does not support repetitions of an extension (all MLA lines must be unique).

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Multiple Line Appearances – Cross-Location	An extension can be configured as a line appearance (BLA or MLA) on a phone whose owner is provisioned at ANY VDNA location within the same tenant.	X	x	x		x	x	x	For calls originating from the MLA line on a given device, the MLA telephone number assigned to that line will be used as the outbound caller id. For this reason, MLA lines configured across locations (Polycom phones only) should not be used for
Music on Hold	Provides incoming callers with a music selection while on hold for any reason (transfer, conference, hold, park, etc.)	Ind	Ind	Ind	Ind	Ind	Ind	Ind	making calls to 911. Note that Music on Hold will be disabled at a tenant level by default. Note: If both callers on an intra-tenant call put each other on hold, neither caller will hear Music on Hold.
Music on Hold Suspension	Callers to subscribers can choose not to listen to music while on hold. Callers can suspend music on hold by pressing the # key while on hold. To resume the music a user must press the # key.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	If on a conference call, the # key is not recognized.

Table 8. Pho	ne Feature Matrix								
Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Locate Me	The Locate Me feature gives you a variety of options for routing incoming calls to your desk phone, to your Voicemail, or to other telephone numbers. It also gives you a Locate Me Rule feature that allows you to route calls based on who they are coming from, and to route them along a sequential path to locate you at specified telephone numbers.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Ring Down	Ring Down is the capability to program a phone set such that, when the speaker key is pressed or the handset is taken offhook, a call is immediately dialed out to a predestined number. An example utilization of this feature would be campus safety phones or door entry phones. AT&T VDNA supports Hotline service where the call is immediately dialed upon taking the phone offhook.	x	X	X		x		X	
Simultaneous Ring	A user can initiate simultaneous calls, up to 3 extensions and/or PSTN numbers when incoming calls to a specific extension are received.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	This capability is part of the locate me feature and must be enabled on the user's COS settings.
SIP Forking	Allows a user to have more than one hard SIP device registered against their extension.	1	√	V	1	1	V	Х	Not supported for analog devices. SIP Forking is supported for the ATA but you cannot have the same TN forked across both ports of a single ATA.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Speed Dialing - Corporate	29 speed dial codes (programmed by the administrator) are available at the company level.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Invoked via *21 through * 49
Speed Dialing - Personal	A user can program up to 20 entries (from the AT&T Voice DNA Personal Web Site).	Ind	Ind	Ind	Ind	Ind	Ind	Ind	Invoked via *01 through * 20
Station to Station Dialing	A user can call other extensions by pressing 3-7 digit extensions within the office.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	
Switch Phones, aka, Mid Call Move	Once a call is established via the user portal (Click to Call), the calling party can use the AT&T Voice DNA Personal Web Site and click to move a call from the originating phone to another number (either VoIP or PSTN).	Ind	Ind	Ind	Ind	Ind	Ind	Ind	A call can only be switched/moved one time.

Feature	Feature Description	Aastra 6757i and 6757CTi (57iCT)	Aastra 6757i/6757i CT Handset	Cisco 7940 7960	Polycom 301 320 321 330 331 600 601 650 560	Polycom 4000/6000	LG 6812 6830	Counter Path eyeBeam	Notes
Tone Suppression	Silence the Call Waiting Tone on a SPECIFIC device/phone for ALL line appearances. A visual indication (blinking line key/ update to LCD display with new call information) will still be presented but there is NO longer an audible indication. There is NO impact on the MLA or BLA appearances of the same TN on other devices/phones.	√	√	x	√	√	√	X	Invoked via the Personal Web Site. Will take 4-5 minutes to take effect. The phone must be rebooted. Independent of the BLA Visual Alert Only Setting. NOT available: Edgewater EdgeMarc FXS supported devices/phones.
Transfer Saved Phone Settings	The user can transfer the Phone Contact directory to a new device. Polycom Phones: The major phone provisioned/user controlled phone settings are stored as part of the Phone Contact Directory (e.g., speed dial, distinctive ring).	X	x	x			x	x	
Voicemail integration	Integration of network-based voice-mail system.	Ind	Ind	Ind	Ind	Ind	Ind	Ind	

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter	Notes
					(IAD)	
Auto Call Back Busy	Auto Call Back Busy (also known as Call Completion on Busy Subscriber or Repeat Dialing) is used when a caller reaches a busy number. When invoked the system will continue to check the busy number and when the	Ind	Ind	Ind	Ind	Invoked by dialing *66 after terminating the busy call. To cancel, dial *86
	line is free, the system rings the caller back and completes the call.					This capability must be enabled on the user's COS settings
Billing (Account) Codes - Mandatory	Requires users to enter a code each time an outside call is made. This code is then output in a call detail report (accessible by the admin) allowing the call information to be used for other reason, e.g., billing codes can be used to identify all calls to a specific client so the client can be billed accordingly.	х	x	X	X	SIP-B feature
Billing (Account) Codes - Optional	Users can optionally enter a code when making a call. This code is then output in a call detail report accessible by the Administrator.	Х	Х	Х	x	SIP-B feature. Invoked via *50.
Bridged Line Appearance (BLA)	Allows multiple extensions to appear on the same handset. As calls arrive on those extensions, the caller information is displayed on the attendant's LCD. BLA includes presence information.	х	X	X	X	SIP-B feature A Polycom 4000/6000 can be the monitored device but it cannot be the monitoring device because it only supports a single lin appearance.

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Distribution Queue	The Administrator can set up a queue for answering calls. Agents log in to the queue and mark themselves as available/unavailable to take calls.	x	X	x	X	Use the phone softkeys for this feature. SIP-B feature. Note that any device can be the final Destination" for an ACD queue (Polycom, Cisco, Analog phone). Polycom: user must indicate available by pressing softkey after every call. LG: automatically logs user back into the queue after 10
Call Forward - Busy	The user can automatically redirect all calls to another telephone when the user's extension is busy.	Ind	Ind	Ind	Ind	Independent of phone if invoked via AT&T Voice DNA Personal Web Site. Can also be invoked via *62 from all phones. *63 to deactivate
Call Forward - No Answer	The user can redirect all calls to another telephone number when the user does not answer. The user can control the duration of the no answer timer based on the phone that the call is forwarded.	Ind	Ind	Ind	Ind	Independent of phone if invoked via AT&T Voice DNA Personal Web Site. Can also be invoked via *92 from all phones.

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Forward - Unconditional	The user can forward calls to an alternate number, either a business extension or an external number.	Ind	Ind	Ind	Ind	Independent of phone if invoked via user portal. Can also be invoked via *72 from all phones.
Call Groups	Companies can create groups of users for specific features. Group dependent features include group call pickup, directed call pickup and intercom.	Ind	Ind	Ind	Ind	

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Hold	The user can place an active call on hold using their business phone.		See Notes Notes			Not a function specifically of the IAD. If the analog device behind the IAD supports a hold button, nothing in the network should prevent the feature from working. For the ATA, can use the phone's switchhook flash (switchhook flash is NOT currently supported on the VG224). From an analog phone: If you use the Hold button (located on the telephone) to place a caller on hold, the caller will not hear Music on Hold is active). If you use the flash hook (on the telephone) to place caller on hold, the caller will hear Music on Hold (provided Music on Hold is active). Note: If both callers on an intra-tenant call put each other on hold, neither caller will hear Music neither will hear Music neither caller will hear Music neither caller will hear Music neither will hear Music neither caller will hear Music neither will hear Music neither caller will hear Music neither will hear Music neither caller will hear Music neither will hear

Table 9. IAD Fea	ature Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Logs	Using the LCD display of the SIP phones, the user can access the missed, outgoing and incoming call logs.	X	X	X	×	Also available via AT&T Voice DNA Personal Web Site.
Call Park	A user can park (put a call on hold) at one telephone set and pick up (continue) the conversation from any other telephone set.	X	X	X	1	Invoked via *98. If phone supports the Call Park/Pickup softkeys and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on device.
Call Pickup	The user can retrieve calls that have been parked against an extension.	X	X	X	1	Invoked via *99. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on device.

Table 9. IAD Fea	ature Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Pickup - Directed	A user can pick up an incoming call ringing at any extension by entering the feature code and the extension. The pickup can be restricted to the user's call group.	X	X	X	√ ·	Invoked via *53. If phone supports the Call Park/Pickup softkeys and Call Park and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on device.
Call Pickup - Group	A business user can pick up an incoming call currently ringing at any phone in the user's call group using a feature code.	X	X	X	1	Invoked via *54. If phone supports the Call Park/Pickup softkeys and Call Pickup are enabled in the user's COS template, you can use the phone softkeys for this feature. SIP B enabled softkey or hard key depending on device.

Feature	Feature Description	Edgewater EM-	VG224	Cisco	Citel P	Notes
		200EW (IAD)	(IAD)	ATA (IAD)	Phone Adapter (IAD)	
Call Reason Display	For Intra-tenant calls, a SIP phone display will indicate information about the call. Specifically:	X	Ind	Ind	Ind	Not supported on IAD/analog phones.
	Hunt Group incoming call display: HG: <calling number=""></calling>					This capability must be enabled on the
	Call Distribution Queue incoming call display: ACD: <calling number=""></calling>					user's COS settings
	Call Forward display: FWD: <calling number=""></calling>					
	Blind or Consultative transfer (called phone): From: <calling number></calling 					
	Blind or Consultative transfer(callers phone): To: <called number=""></called>					
	Cisco phones display both the From: and To: field.					
	Calls dialed from the voicemail platform are NOT classified as an intra-tenant calls.					
Call Restriction	An administrator can allow/restrict dialing of the following calls: long distance calls, information, international, international operator, local, local operator, long distance operator, premium service, and toll free.	Ind	Ind	Ind	Ind	

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Transfer - Blind	Lets a user transfer an active call to another extension through a phone soft key	1	√	V	V	Supported by the IAD only if the analog phone has a transfer key. For the ATA, works via the switchhook flash (switchhook flash is NOT currently supported on the VG224). SIP enabled softkey or hard key depending on device.
Call Transfer - Consultative	A user can converse with a 3rd party and then transfer the call to that party through a phone soft key.	1	See Notes	1	V	Supported by the IAD only if the analog phone has a transfer key. For the ATA, works via the switchhook flash (switchhook flash is NOT currently supported on the VG224).

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Waiting	Notifies a user on an active call that there is a second incoming call. The user can switch between the two incoming calls.		X			From the phone enabled/disabled using *70 for the current call only. Call Waiting is enabled on the user's COS settings by default. The Admin can disable Call Waiting on the users' COS setting is a user needs Call Waiting disabled for all calls. Switchhook flash is required for Call Waiting and is NOT currently supported on the VG224. For the Softphone: 2 beeps will be heard through your headset, a pop-up window is displayed on your PC screen, with answer, ignore and the caller ID of the caller. In window of the softphone image, Ignore will appear along with caller ID of the caller. To Disable Call Waiting on the

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Call Waiting						Enter *70 immediately before the number your are dialing, then press the green connect icon. (e.g., off-net: *708005551234 <connect>, onnet: *701234 <connect>)</connect></connect>
Caller ID	Displays the phone number of the calling party to the user.	V	V	√	1	Supported by the IAD only if the analog phone has caller ID capabilities.
Caller ID block/unblock	There are 2 modes of Caller ID Block/unblock. Mode 1: The admin enables this feature for a user. Caller ID for that user is blocked on all calls outside the business (can unblocked call on a call-by-call using star code). Mode 2: Caller ID is typically sent. The user may block Caller ID outside the business on a call-by-call basis by dialing a star code.	Ind	Ind	Ind	Ind	*67 (caller id blocking on a call-by call basis) - turns on caller id blocking on a call (requires that the Administrator has enabled this feature for the user). *82 (caller id unblocking on a call-by-call basis) - turns caller id blocking off on a call for a user who has the automatic caller ID blocking feature enabled. *67 functionality must be enabled on the user's COS settings
Caller ID Presentation Contact	A user, or corporate contact data, can provide the name associated with a phone number in the call logs and caller ID display.	Ind	Ind	Ind	Ind	Applies to Intra Tenant calls only.

Table 9. IAD Fea	nture Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Caller Name	The network-provided name (from the contact database) is displayed to the user's phone.	1	1	√	V	Supported by the IAD only if the analog phone has caller ID capabilities.
Called Party Display on Calling phone	The name of the Called number via the AT&T Voice DNA Personal Web Site input (User Contact, Company Contact). This name will be displayed on the Calling Party phone.	X	X	X	X	
Click to Call	The user can click on an entry in a call log or directory, or click a dial button to call a selected phone number.	Ind	Ind	Ind	Ind	This feature is invoked via the AT&T Voice DNA Personal Web Site.
Phone Log: Select & Call	The user can select an entry displayed on the phone (e.g., in Call Logs) to call a selected phone number.	See Notes	See Notes	See Notes	See Notes	Supported by the IAD only if the analog phone supports this feature.
Conferencing – On- Demand Conference (Click to Conference)	Setup for this feature is via the AT&T Voice DNA Personal Web Site. A user can perform Centralized Conferencing for up to 10 callers.	Ind	Ind	Ind	Ind	10 callers including the call initiator.
Conferencing (via SIP phones up to 3 call legs)	A user can perform Conferencing via the phone for up to 3 callers.	٨	х	√	х	3 callers including the call initiator. Only supported on the ATA if the analog device has a conferencing capability.

Table 9. IAD Fea	ture Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Configurable Speed Dial Keys	Ability for the end user to assign speed dial destinations to unused hard function keys (normally used as line keys).	х	X	X	X	With Polycom multi- line phones, speed dials appear after the line registrations. Does not interfere with BLA or MLA. Any new BLA or MLA appearances will be assigned first, followed by the end user set speed dials. End user sets the speed dial keys via the phone menu.
Direct Inward Dialing (DID)	A caller can access another user's extension directly, without going through an attendant.	Ind	Ind	Ind	Ind	

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Distinctive Ringing	Lets users hear different types of rings depending on internal or external callers. Rings are endpoint specific and distinctive ring is enabled through class of service. Only two types of rings are supported (for the inside or Outside calls): 1. Other tenant or PSTN, and 2. Internal Ring: another user within a user group	1	X	1	X	For Cisco VG224. there is no distinctive ring, both onnet & offnet calls are the same rings. Polycom phones allow you to set your ring tone from the phone menu. However, in order for the user to select the ring tone to be used the Distinctive Ringing checkbox (located on the Ring Settings page of the Personal Web Site for the user) must be unchecked (disabled). For MLA: The distinctive ring must be disabled for the employee who ownsthe TN of the line appearance. For example, for a user's line appearance on my phone to silent ring, the Distinctive Ringing checkbox (located on the Ring Settings page of the Personal Web Site for the user) unchecked (disabled).

Table 9. IAD Fea	nture Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Direct Outward Dialing (DOD)	A caller can place a call, without going through an attendant, to a PSTN number, by dialing an external access code (such as 9) as defined by the specified dialing plan.	Ind	Ind	Ind	Ind	
Do Not Disturb	A user can specify do not ring this phone from the AT&T Voice DNA Personal Web Site. From the phone, the call treatments will remain in effect. Based on the specified call treatments, calls will be sent to the mailbox, if purchased; or busy, if there is no mailbox.	٨	1	√	√	Polycom - hard key. LG 630 – hard key Cisco - soft key (menu driven). LG 6812 – soft key (menu driven) Supported by IAD if phone device has this capability.

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Executive Busy Override (EBO)	Executive Busy Override (EBO), also known as Barge-in, allows the user to insert themselves into an existing two-party call. There are two ways (modes) a user can use EBO: 3-way: the user enters an existing two-way call, turning it into a 3-way conference. Override: the user can disconnect an ongoing call on the target extension and have themselves connected to the target extension. If the target extension is idle, the EBO user will by-pass all Call Forwarding and Do Not Disturb settings to connect to the target extension. Extensions can be marked as exempt from EBO so that NO barge-in is allowed.	Ind	Ind	Ind	Ind	*77 <mode digit=""><extension> Invoked via *77 and the EBO mode number. The EBO modes are: Digit Mode 1 3-Way 4 Override This capability must be enabled on the user's COS settings EBO exemption is a another COS setting which allows the Administrator to mark an extension as exempt from barge-in. EBO is not supported for initiators behind an ATA. If *77 is dialect from behind an ATA with the expectation of EBO service, the initiator will now have anonymous calls rejected. *87 should be dialed to clear the condition.</extension></mode>
Hunt Groups	The Administrator can enter a series of numbers to route calls when the previous number dialed does not	Ind	Ind	Ind	Ind	

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Intercept Announcements	Intercept Announcements are system wide announcements provided to the user as they use service features. Some announcements are informational such as "Do not disturb activated" and some provide guidance for using a feature such as "We are sorry. Call pick up could not be completed because there is no call parked to the number provided."	Ind	Ind	Ind	Ind	

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Intercom Calling	Allows the phones to act as an intercom. The user enters a feature code and extension to invoke the service. The called extension automatically answers and a voice path is established between the two phones.	X	X	X	X	Invoked via *96. SIP-B feature. If invoked to a non- SIP-B device, the terminating phone will ring (will not be forced off hook which is the intent or intercom). The calling phone must have the Intercom COS setting enabled. The called phone does not need the Intercom COS setting enabled. If the Called phone does not support Intercom, the phone will ring rather than go offhook. Intercom only works if the target device in a Polycom or LG phone (e.g., Cisco can call Polycom/LG but when a Polycom/LG calls Cisco the Cisco phone will ring rather than go offhook). You cannot invoke Intercom to a forked device (e.g., MLA). If forking is active (e.g., 2 registered devices for the destination TN), intercom will not work.

Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Intercom Calling						If you intercom to the BLA manager (manager/secretary scenario), the manager registration will go offhook.
Last Call Return	A user can return the last incoming call where caller ID is available by dialing a feature code.	1	√	√	1	Invoked via *69
Last Number Redial	A user can automatically redial the last dialed number	√	√	√	V	Invoked via *00 or redial button on the phone
Message Waiting Indicator (MWI)	A phone lamp notifies the user that a message has been left in the mailbox. A message button acts as a speed dial key to voicemail. Analog phone users receive interrupted dial tone when a message is waiting. A user must order voicemail (via the premium package).	√	√	√ ·	V	For IAD supported devices - lamp if the phone supports an MWI lamp, otherwise interrupted dialtone. For the Polycom 4000/6000, the MWI will be heard as a interrupted dial tone and the envelope icon on the display (the phone lights do not flash). You can clear the MWI indication from the phone by pressing the clear button.
Message Waiting Indicator (MWI) for MLA or BLA appearances		x	X	х	Х	Not supported for analog devices. The Polycom 4000/6000 and Eyebeam are single line phones.

Table 9. IAD Fe	ature Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Multiple Line Appearances – Basic	A user can have more than one phone line appear on the phone set OF DIFFERENT EXTENSIONS/DID.	х	х	X	X	Not supported for analog devices. The Polycom 4000/6000 and Eyebeam are single line phones. Eyebeam automatically supports 6 lines of the same telephone number.
Multiple Line Appearances – Repetitions	A user can have more than one phone line appear on the phone set OF THE SAME EXTENSION/DID.	X	X	X	X	Not supported for analog devices. The Polycom 4000/6000 and Eyebeam are single line phones. Eyebeam automatically supports 6 lines of the same telephone number. 7960/7940 - Cisco does not support repetitions of an extension (all MLA lines must be unique).
Music on Hold	Provides incoming callers with a music selection while on hold for any reason (transfer, conference, hold, park, etc.)	Ind	Ind	Ind	Ind	Note that Music on Hold will be disabled at a tenant level by default. Note: If both callers on an intra-tenant call put each other on hold, neither caller will hear Music on Hold.

Feature	Feature Description	Edgewater EM- 200EW (IAD)	(IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Music on Hold Suspension	Callers to subscribers can choose not to listen to music while on hold. Callers can suspend music on hold by pressing the # key while on hold. To resume the music a user must press the # key.	Ind	Ind	Ind	Ind	If on a conference call, the # key is not recognized.
Locate Me	The Locate Me feature gives you a variety of options for routing incoming calls to your desk phone, to your Voicemail, or to other telephone numbers. It also gives you a Locate Me Rule feature that allows you to route calls based on who they are coming from, and to route them along a sequential path to locate you at specified telephone numbers.	Ind	Ind	Ind	Ind	
Ring Down	Ring Down is the capability to program a phone set such that, when the speaker key is pressed or the handset is taken offhook, a call is immediately dialed out to a predestined number. An example utilization of this feature would be campus safety phones or door entry phones. AT&T VDNA supports Hotline service where the call is immediately dialed upon taking the phone offhook.		X		x	For EM200IAD Ring Down must be configured for each line.
Simultaneous Ring	A user can initiate simultaneous calls, up to 3 extensions and/or PSTN numbers when incoming calls to a specific extension are received.	Ind	Ind	Ind	Ind	This capability is part of the locate me feature and must be enabled on the user's COS settings.
SIP Forking	Allows a user to have more than one hard SIP device registered against their extension.	1	Х	√	х	Not supported for analog devices. SIP Forking is supported for the ATA but you cannot have the same TN forked across both ports of a single ATA.

Table 9. IAD Fea	ature Matrix					
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes
Speed Dialing - Corporate	29 speed dial codes (programmed by the administrator) are available at the company level.	Ind	Ind	Ind	Ind	Invoked via *21 through * 49
Speed Dialing - Personal	A user can program up to 20 entries (from the AT&T Voice DNA Personal Web Site).	Ind	Ind	Ind	Ind	Invoked via *01 through * 20
Station to Station Dialing	A user can call other extensions by pressing 3-7 digit extensions within the office.	Ind	Ind	Ind	Ind	
Switch Phones, aka, Mid Call Move	Once a call is established via the user portal (Click to Call), the calling party can use the AT&T Voice DNA Personal Web Site and click to move a call from the originating phone to another number (either VoIP or PSTN).	Ind	Ind	Ind	Ind	A call can only be switched/moved one time.
Voicemail integration	Integration of network-based voice-mail system.	Ind	Ind	Ind	Ind	
Tone Suppression	Silence the Call Waiting Tone on a SPECIFIC device/phone for ALL line appearances. A visual indication (blinking line key/ update to LCD display with new call information) will still be presented but there is NO longer an audible indication. There is NO impact on the MLA or BLA appearances of the same TN on other devices/phones.	x	X	X	X	Invoked via the Personal Web Site. Will take 4-5 minutes to take effect. The phone must be rebooted. Independent of the BLA Visual Alert Only Setting. NOT available: Edgewater EdgeMarc FXS supported devices/phones.

Table 9. IAD Feature Matrix								
Feature	Feature Description	Edgewater EM- 200EW (IAD)	VG224 (IAD)	Cisco ATA (IAD)	Citel P Phone Adapter (IAD)	Notes		
Transfer Saved Phone Settings	Transfer Saved Phone Settings all the Phone Contact directory to be moved to a new device.	Х	X	X	Х			
	Polycom Phones: The major phone provisioned/user controlled phone settings are stored as part of the Phone Contact Directory (e.g., speed dial, distinctive ring).							

Bridged Line Appearances

This section describes how secondary lines behave in a variety of call situations. Assume a scenario where an assistant and a receptionist have secondary line appearances for their manager's phone.

Table 10. BLA Secondary Line Behavior

Call Event	LED Status	Secondary Line Behavior
The manager is on an active call.	The LED button for the manager's phone is lit.	The secondary lines are available.
An incoming call rings the manager's phone.	The LED button flashes on all three phones.	The assistant or the receptionist can pick up the call by pressing the line button for the manager's extension on their phone while on-hook, or by picking up the handset and then pressing the line button. When one picks up the call, the secondary line appearance on the other two phones are blocked and no one else can use that line.
The manager is on the phone and gets a call on their second line.	The LED button for the active call is lit. The LED button for the manager's second line flashes on all phones. If the assistant or the receptionist only has one line appearance for the manager's phone, they will only see the activity on the manager's first line.	The secondary line for the active call is blocked on the assistant's and the receptionist's phones. The assistant or the receptionist can pick up the call by pressing the line button for the manager's second line on their phone while on-hook, or by picking up the handset and then press the secondary line button. When one picks up the call, the secondary line appearances on the other two phones are blocked.

Table 10. BLA Secondary Line Behavior

Call Event	LED Status	Secondary Line Behavior
The manager puts a caller on hold.	The LED for the manager's line flashes on all three phones.	The manager, the assistant, or the receptionist can pick up the held call by pressing the associated line button.
		For Aastra Phone: With a call on hold, in order to make an outgoing call, a line button must be pressed (even if a dial tone is heard).
The manager receives a call but has activated Do Not Disturb on his phone.	For a LG phone, the manager phone does not ring or show the incoming call. It is quiet. For a Polycom phone, the phone does not ring but the display shows the caller-id of the incoming call and the manager has the option to pick up the call - despite DND.	The assistant or the receptionist can pick up the call by pressing the line button for the manager's second line on their phone while on-hook, or by picking up the handset and then press the secondary line button. When one picks up the call, the secondary line appearances on the other two phones are blocked.
The manager's phone is ringing. They do not want to be disturbed and presses the Do Not Disturb button before the assistant or the receptionist picks up the call.	When the manager's phone rings, the line LEDs flash on all phones.	The assistant or the receptionist can pick up the call by pressing the line button for the manager's second line on their phone while on-hook, or by picking up the handset and then press the secondary line button. When one picks up the call, the secondary line appearances on the other two phones are blocked.
The manager is not in their office and has set their call treatment to forward all incoming calls to their cell phone.	The incoming call does not ring on any of the phones.	The call cannot be picked up by the receptionist.

Table 10. BLA Secondary Line Behavior

Call Event	LED Status	Secondary Line Behavior
The manager is not in their office and has enabled the Reach Me feature to	When the manager's phone rings, the LED buttons flash on all phones.	If nobody picks up the call, it is forwarded according to the Reach Me setting.
forward their calls.		If the assistant or the receptionist picks up the call before the fourth ring, they are connected to the caller. The assistant or the receptionist must transfer the call back to the manager's extension to activate the Reach Me feature.
The manager parks a call at their extension.	While the manager is on the phone, the associated line LED is lit on all phones. Once the manager has parked the call, the LED buttons are not lit any longer.	The assistant and the receptionist cannot pick up the manager's active call. As soon as the call is parked, either one can pick it up using the call pick up feature.
The manager is a member of a hunt group. A call comes into the hunt group and rings the manager's phone.	When the manager's phone rings, the line LEDs flash on all phones.	The assistant or the receptionist can pick up the call by pressing the line button for the manager's extension on their phone while on-hook, or by picking up the handset and then press the secondary line button. If nobody picks up the call, it goes to the mailbox.

Table 10. BLA Secondary Line Behavior

Call Event	LED Status	Secondary Line Behavior
The manager is a representative in a Call Distribution group, logged into the queue, and available. A call comes into the call center group.	When the manager's phone rings, the line LEDs flash on all phones.	The assistant or the receptionist can pick up the call by pressing the line button for the manager's extension on their phone while on-hook, or by picking up the handset and then press the secondary line button.
		If nobody picks up the call, it is forwarded to the next representative in the Call Distribution group.
The manager is a representative in a Call Distribution group, logged into the queue but is unavailable. A call comes into the call center group.	Since the manager is unavailable for the call center group, the call will not ring on any of the phones.	The call will not ring the assistant phones.
911 restriction & BLA	The Manager's line appearance on the assistant's phone is consistent with the Manager's restriction.	Therefore, if the manager moves their device to an unknown location, the manager's appearance on the assistant's phone is also restricted (but the assistant's main line is fine).
A message is left for a manager.	There is MWI on the manager phone.	For LG Phone: No indication on the manager's line appearance on the assistant phone.
		For Polycom Phone: An indication appears on the assistant/s phone.

Package Features

The following is a listing of features. Not all of these features may be available to all users. These features will only be available if the feature is contained and activated in your Class of Service template.

Table 11. Package Features

Package Content	Description
Anonymous Call Rejection	Lets the user refuse all calls from callers who have their caller ID blocked. This allows the Administrator to configure a user's extension to reject all incoming calls without caller ID. This feature has no affect upon internal calls.
Authorization Code	Allows employee to unlock restricted phone by dialing *80*extension*authorization pin*TN. Employees can change their pin via the Security settings on the My Profile screen of the Personal Web Site.
Call Forward Busy	Lets the user automatically redirect calls to another number when the user's phone is busy. Refer to the Using Locate Me section.
Call Forward Fixed To Voicemail	Call Forward Fixed to Voicemail. Refer to the Using Locate Me section.
Call Forward No Answer	Lets the user automatically redirect calls to another number when the user does not answer. Refer to the Using Locate Me section.
Call Forward Unconditional/Call Forward Variable	Lets the user forward all calls to an alternate number, either a business extension or external number. Refer to the Using Locate Me section.
Call Park	Lets users place an active call in hold state for retrieval by another user. Refer to the Using Locate Me section.
Call Forward Busy	Lets the user automatically redirect calls to another number when the user's phone is busy. Refer to the Using Locate Me section.
Call Forward Fixed To Voicemail	Call Forward Fixed to Voicemail. Refer to the Using Locate Me section.

Table 11. Package Features

Package Content	Description
Call Pickup	Lets users retrieve a call that has been parked against an extension
Click To Conference	Allows employees to initiate 10-way conference call via end-user portal.
Com Portal	Provides access to the User Personal Web Site.
Contact Search Lookup	Allows users to search feature for phone contacts.
Distinctive Ringing	A different type of ring sounds depending upon internal or external caller. These sounds are set at the server level, so no user interaction is available.
External Transfer	Allows users to transfer incoming call to a number not within the tenant.
Intercom Within Tenant	Lets user place call to another user where the phone beeps and speakerphone is automatically activated.
Last Call Return	Lets user return the last incoming call via a feature code.
Locate Me	Allows users to control and manage their incoming calls. Call forwarding and Locate Me are prevented from sending calls to international phone numbers.
Mandatory Account Codes	Requires user to enter a code each time a call is made.
Optional Account Code	Allows user to enter a code each time a call is made.
Outlook Integration	Outlook Integration. Refer to the Outlook Integration Software section.

 Table 11.
 Package Features

Package Content	Description
Remote Click To Call	Remote Click to Call is similar to the Click to Call feature except you're not constrained to using your work TN as the start number (e.g., you can specify your cell or any other number). The called party still sees your AT&T Voice DNA work telephone number as the caller id even though you really started the call from an alternate telephone number/device. Remote Click to Call requires a COS setting to be enabled for the user by the Administrator. When invoking Remote Click to Call (or Click to Conference), always check the originating TN to ensure that it reflects the contact number for the device at which you will initiate the call.
Simultaneous Ring	Allows users to indicate whether or not all devices ring or just the primary.
Switch Phone & Remote Click To Call	Allows users to switch phones during a call.
Voicemail	Service that provides a login to access the voice mail feature. New message indicator will appear on tool bar. The user will be able to view voicemail and fax messages, Click to Call from message on Personal Web Site and direct messages to their e-mail via a .wav file. Refer to the Messages section.

Cisco Phones

Cisco Phone Limitations

Not all features are fully supported by the following Cisco Phones: 7940G, and 7960G.

These features are:

- Call Distribution
- Bridged Line Appearances
- Intercom Calling
- Account Codes

Display Date and Time on a Cisco Phone

Voice DNA provides the ability for the customer to display date and time on the phones (the default state for time display is turned **ON**). The Time and date display will need to be manually updated for Voice DNA remote workers. Time display is not supported for the following phone types (for Voice DNA or RW):

Analog phones behind an ATA 186.

Instructions on how Remote Workers can update their supported Cisco phone to display the date and time are included in the following section.

Setting the Display Date and Time for a Remote Worker on the Cisco 7940/7960 Phones

For a Remote Workers, the time server and offset will need to be manually configured on the phone. The Daylight Savings Time (DST) parameter would also be set to **Yes** normally, unless the phone is physically located in a state or territory that does not use DST and therefore, does **NOT** want DST adjustments to be automatically made.

To set the phone display data and time, perform the following:

- 1. Press the phones **Settings** (Menu) hard button.
- When Settings Menu is displayed, scroll to Unlock Config choice and press the Select soft button.
- 3. Choose the **Alpha** (default) or the **Number** soft button to enter password characters or digits (default is **cisco**).
- 4. Choose the **Accept** soft button once the password had been entered correctly.
- 5. When the **Settings Menu** is displayed, scroll to **Localization** choice and press the **Select** soft button.
- When the Localization Setup Menu is displayed, scroll to SNTP Settings and press the Select soft button.
- When the SNTP Configuration Menu is displayed, scroll to SNTP Server choice and press the Edit soft button.

- 8. Use the ← soft button to erase any existing value and then enter the IP address for the SNTP Server to be used making use of the . (period) soft button to separate the octets. Cisco phones require an IP address. To find the IP address for SNTP server do an nslookup on one of the following FQDNs:
 - time.nist.gov
 - time-nw.nist.gov
 - time-a.nist.gov
 - time-b.nist.gov

Or you can go to the following NIST page which lists IP addresses: http://tf.nist.gov/service/time-servers.html

■NOTE:

Customers who have the Cisco phones must enter the IP address of their SNTP server into the phones manually and then reboot. If the time and date display are incorrect after the reboot, reenter the IP address using a different NIST IP address and perform the following steps again.

- 9. Select the **Accept** soft button once the correct address has been entered.
- 10. Scroll to **DST Auto Adjust** and set to **Yes** or **No** (if in non-DST location) as appropriate.
- 11. Select the **Back** soft button to return to the **Localization Setup Menu**.
- 12. Scroll to **Time zone** and press the **Change** soft button repeatedly until the correct **Time zone** is displayed. The choices displayed move around the globe in an easterly direction, which means they are increasing. Therefore, if you need to shift to an hour earlier time zone for example, you would need to press the **Change** soft button 31 times.
- 13. Select the **Save** soft button. This causes the new settings to be used.
- 14. Select the **Back** soft button to exit each menu level.
- 15. Reboot the phone.

Cisco 7960

■NOTE:

For more information about the Cisco 7960 phone, go the Cisco web site at: http://cisco.com/, then search for the 7960 User Guide.

Indicator/Button	Description
Handset light strip	Indicates an incoming call and new message.
Phone screen	Shows phone features such as the time, date, your phone number, caller ID, line/call status, and softkey tabs.
Model type	Indicates your Cisco IP Phone model.
Programmable buttons	Configurable buttons that provide access to:
	Different phone lines, or extensions (line buttons)
	Frequently dialed phone numbers (speed dial buttons)
	Web-based phone services (service buttons)
	Specialized phone features, such as privacy (privacy button)
Footstand button	Allows you to adjust the angle of the phone base.
Directories button	Opens/closes the Directories menu. Use it to access call logs and corporate directories (if available).
Help button	Activates the Help menu.
or	
?	
Settings button	Provides access to phone settings such as contrast and ringer sound, network configuration, and status information.
Speaker button	Toggles the speaker on or off.
Mute button	Toggles the mute feature on or off.

Headset button	Toggles the headset on or off.
Volume button	Increases or decreases volume for the handset, headset, or speakerphone (depending upon which is currently active). Also controls the ringer volume (if on-hook), and the screen contrast.
Services button	Provides access to phone services (if available).
Messages button	Provides access to a message system (if available).
Navigation button	Allows you to scroll through menus and highlight items. Use in conjunction with softkeys to activate highlighted items.
Keypad	Allows you to dial phone numbers, enter letters, and choose menu items.
Softkey buttons	Each activates a softkey. Softkeys point to options displayed along the bottom of the phone screen. Softkey options can change depending on the status of your phone (for example, if the phone is active or idle).

Special Notes for Cisco 7960G

1. To unlock the phone press **#

Viewing or Dialing Missed Calls - 7960G

On your Phone:

If your phone display indicates you have missed one or more calls, you can use the **Missed Calls** option on the directories menu to view the call history.

- 1. Press the (Directories) button.
- 2. Press the **Select** softkey to select **Missed Calls** from the Directory menu. The call history for missed calls is displayed on the LCD screen.
- 3. Press the **Dial** softkey to speed dial the highlighted number, as displayed, from the missed call list.
- **4.** Press the **Exit** softkey twice to exit the Missed Calls directory and the directories menus.

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

Cisco 7940G

■NOTE:

For more information about the Cisco 7940G phone, go the Cisco web site at: http://cisco.com/, then search for the 7940G User Guide.

Feature	Function
Cisco Unified SIP IP phone model type	Shows the Cisco Unified SIP IP phone model number
LCD screen	Displays information such as line/call status, phone number, and soft key tabs.
Line or speed-dial button	Opens a new line, speed-dials a phone number or ends a call. The 7940G has two lines.
Footstand button	Allows you to adjust the angle of the phone base.
Directories button	Provides access to phone directories.
Question button	Not in use.
Settings button	Provides access to phone settings such as display contrast and ring type.
Speaker button	Toggles the speaker on or off.
Mute button	Toggles the mute feature on or off.
Headset button	Toggles the headset on or off.
Volume button	Increases or decreases volume for the handset, headset, or speakerphone (depending upon which is currently active). Also controls the ringer volume (if on-hook), and the screen contrast.
Services button	Provides access to phone services (if available).

Feature	Function
Messages button	Provides access to a message system (if available).
Navigation button	Allows you to scroll through menus and highlight items. Use in conjunction with softkeys to activate highlighted items.
Keypad	Functions like a traditional keypad.
Softkey buttons	Engage the functions displayed on the corresponding LCD tabs.
Handset with indicator light	Functions like a traditional handset and provides message waiting indicator light and message waiting indicator tone.

Viewing or Dialing Missed Calls - 7940G

On your Phone:

If your phone display indicates you have missed one or more calls, you can use the **Missed Calls** option on the directories menu to view the call history.

- 1. Press the (Directories) button.
- 2. Press the **Select** softkey to select **Missed Calls** from the Directory menu. The call history for missed calls is displayed on the LCD screen.
- 3. Press the **Dial** softkey to speed dial the highlighted number, as displayed, from the missed call list.
- Press the Exit softkey twice to exit the Missed Calls directory and the directories menus.

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

Special Notes for Cisco 7960G

For the 7940: If you use *69 to redial a user and then park the call, the call display shows **To:** *69 while the call is being parked. When the call is retrieved, the call display will show **To** *99<ext parked to>.

Polycom Phones

The first time a Polycom phone powers up, it could take a significant amount of time to download the necessary firmware, configuration files (10-15 minutes).

Time to Reboot Polycom Phone

When rebooting a Polycom phone it may takes as long as five minutes to reboot.

Dialing Instructions/Tips for Polycom Phones

On Polycom phones, if you start dialing the number without pressing **new call**, you can take your time dialing (more than 4 seconds). Then, when you are ready, you hit the **dial** key. The digits will not be sent until you press **dial**.

Display Date and Time on a Polycom Phone

Voice DNA provides the ability for the customer to display the date and time on the phones (the default state for time display is turned **ON**). The Time and date display will need to be manually updated for Voice DNA remote workers.

Instructions on how Remote Workers can update their supported Polycom phone to display the date and time are included in the following section.

Setting the Display Date and Time for a Remote Worker on Polycom Phones

To display the date/time on a Polycom phone, perform the following:

- 1. Select **Menu** hard key.
- 2. Select 3. Settings.
- 3. Select 2. Advanced.
- 4. Enter Admin Password 456 and then select Enter softkey.
- 5. Select 1. Admin Settings.
- 6. Select 1. Network Configuration.
- 7. Scroll down to **SNTP Address** and select **Edit** softkey.
- Enter a public SNTP server address from the following list and select the Ok softkey.
 - 0.attvdna.pool.ntp.org
 - 1.attvdna.pool.ntp.org
 - 2.attvdna.pool.ntp.org
 - 3.attvdna.pool.ntp.org
- 9. Scroll down to **GMT Offset** and select **Edit** softkey.
- Using the left and right scroll keys select the correct GMT offset for your time zone (GMT = Greenwich Mean Time; Northeast USA -5) and select the Ok softkey.
- 11. Select **Exit** softkey two times.

12. Select **3. Restart Phone** and wait for your phone to reboot. After the phone reboots, the correct date and time will be displayed.

Disabling the Polycom Display for Date/Time

Perform to the following procedure to disable the date and time in the display on the phone.

- 1. Press the phones **Menu** hard button to display the **Main Menu**.
- 2. Scroll to **Settings** and press the **Select** soft button.
- 3. Scroll to **Basic** and press the **Select** soft button.
- 4. Scroll to **Preferences** and press the **Select** soft button.
- 5. Scroll to **Time and Date** and press the **Select** soft button.
- Scroll to **Disable** and press the **Select** soft button, which will return you to **Preferences** menu.
- 7. Press the Exit soft button to return to Basic Settings menu.
- 8. Press the **Exit** soft button to return to **Settings** menu.
- 9. Press the Exit soft button to return to Main Menu.
- Press the Exit soft button to return to phone display. The phone display is now disabled.

Setting Ring Tones for Polycom Phones

Polycom phones allow you to set your ring tone from the phone menu. However, in order for the user to select the ring tone to be used, the Distinctive Ringing checkbox (located on the Ring Settings page of the Personal Web Site) must be unchecked (disabled). The phone will need to be rebooted (power cycle) for the changes to take effect.

Speed Dial Setup Locally on Polycom Phones

A speed dial number can be setup locally on the Polycom Phones. These will only visible to the telephone user.

To setup the speed dial, enter:

Menu > 1. Features >1. Contact Directory

If you add or edit an entry here, near the bottom, after entering name and phone number there are options for auto-divert (selective call forward) and auto-reject (similar to DND which sends the call direct to VM and does not ring the phone).

Polycom 301

■NOTE:

For more information about the Polycom 301 phone, go the Polycom web site at: http://www.polycom.com/, then search for the 301 User Guide.

Feature Key	Description	
®	Mutes audio transmission locally during calls.	
③	Allows users to place and receive calls through an optionally connected headset.	
	Line or Speed Dial keys.	
	Scrolls through menu options.	
	Scrolls through menu options.	
Menu	Access local and server features.	
Do Not Disturb	Cancels ringing and directs incoming calls to your mailbox (if supported). After pressing the DND button you get a choice: apply to all lines or by line (select by line label). If no mailbox is available it rejects to the user's final call treatment). The DND is set by registration (for example, if you have repetitions of your own TN, the DND would apply to all appearances of that single extension).	
Redial	Dials last connected party from the phone.	
Hold	Places current active call in Hold state.	
NewCall Fwd More	Soft keys to select from various context- sensitive options.	
	Volume keys to adjust audio and ringer volume.	
Dial Pad	General dialing and alphanumeric entry (including special characters)	

Special Notes for Polycom 301

- 1. The calling number may not display properly on the Polycom SIP 301 phone as not all of the extension may be displayed. For example, if the calling number is 3627, it may appear as ...627.
- 2. The speaker (no microphone) on the Polycom 301 SIP phone is a one-way monitor (listen only).
- 3. If a user tries to use the Intercom feature to call a user with a Polycom 301 SIP phone, the phone will ring as though it is an incoming call (auto-answer is **not** available for Intercom on the Polycom 301 SIP phone).
- 4. The Polycom SIP 301 phone can intercom out as the call initiator hands-free, but the call initiator has to pick up the handset on the intercom connect in order to speak to the called user.
- 5. For all Polycom phones the handset plug on the Polycom phones has a 2 step plug in. When plugging the cord into the handset you should be able to hear two clicks (2 "Click" process) for the handset to work properly.

Viewing or Dialing Call Lists – Polycom 301

On your Phone:

A local list of calls missed, received, and placed is maintained by the phone.

Press Select Features followed by Call Lists. Select Missed, Received, or Placed Calls as desired. Call information will be displayed.

Choose:

- 1. Edit to amend the dial string if necessary.
- Dial to return the call.
- 3. **Exit** to return to the previous menu.

Press the **Exit** soft key again to return to the idle display.

To quickly view the **Missed Calls** list from the idle display press .

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

Polycom 320/321/330/331

■NOTE:

For more information about the Polycom 320/321/330/331 phone, go the Polycom web site at: http://www.polycom.com/, then search for the 320/321/330/331 User Guide.

Feature Key	Description
MWI	Indicates that you have a new message.
Soft Keys	The screen displays labels for these keys, to identify their context-sensitive functions
Line Indicators	Individual multi-color LEDs display the dynamic call state and remote user status (presence). The mapping is:
	Solid green — An active call is in progress.
	Fast flashing green—There is an Incoming (ringing) call.
	Flashing green—The call is held by the other party.
	Flashing red—The call is on hold.
	Solid red—The line is busy remotely (shared lines).
Line 1 and Line 2	Use these keys to activate a line that is assigned to your phone.
Hold	Holds an active call or resumes a held call.
0	Headset — Allows you to place and receive calls through an optionally connected headset.
(4)	Speakerphone — Allows for hands-free communication during calls.
Hands-free Microphone	Picks up audio during hands-free calls. Place your phone on a hard, flat surface for best results.
	Use these to adjust the volume of
	the handset, headset, speaker, and ringer.

Feature Key	Description
	Microphone Mute —Mutes audio transmission locally during calls.
Dial	Dial phone number entered or view placed call list.
Menu	Access local (your phone) and call server features.
	Use the arrow keys to scroll (ⓓ left, ℗, right, ๊ up, ఄ down) through the displayed information.
	Use (the Select key) to select a field of displayed data or enter edit mode for some settings.
Dial Pad	These 12 keys provide the 10 digits, the alphabetic characters, and special characters available in context-sensitive applications.

Special Notes for Polycom 320/321/330/331

- 1. The Polycom 320/321/330/331 displays the Lines in descending order. For example, Line 1 is displayed at the bottom and Line 2 is at the top.
- 2. Re-setting a Polycom phone to the factory default erases ALL of the phone settings including AT&T service specific information. Any information that was entered into the phone manually via the phone menu would have to be reentered (e.g., protocol, boot server, time/day display enablement, contacts, etc.). Only when a phone has been moved from one Voice DNA account to another Voice DNA account is it worthwhile to use the 468* sequence to reset a Polycom phone to the factory defaults. All this accomplishes is to wipe out all of the previous account information from continuing to be displayed on the phone. When the phone is able to successfully download its new configuration files for the first time containing its new account information, the old account information is replaced.

■NOTE:

If you return the phone to factory defaults, you may need to call Customer Care to restore unencrypted files since restoring to factory defaults wipes out existing encryption keys.

3. For all Polycom phones - the handset plug on the Polycom phones has a 2 step plug in. When plugging the cord into the handset you should be able to hear two clicks (2 "Click" process) for the handset to work properly.

Viewing or Dialing Call Lists – Polycom 320/321/330/331 On your Phone:

A local list of calls missed, received, and placed is maintained by the phone.

The initial view of the lists shows the list title and the first two calls in the list, where the first call is displayed in reverse video to indicate that it is currently selected. A symbol to indicate whether the call was answered or missed is shown for each entry in the incoming list.

Received Calls:

Press the Callers soft key to view the Received Calls list.

Placed Calls:

- 1. Press Dial hard key to view the Placed Calls List.
- 2. Use the up/down arrow keys to scroll through the list and select the number.
- 3. Press the Dial hard key to complete the call.

You can also dial $^{*}00$ to redial the last Placed call on the Polycom 320/321/330/331 phone.

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

Polycom 600/601/650/560

■NOTE:

For more information about the Polycom 600, 601, 650, or 560 phones, go the Polycom web site at: http://www.polycom.com/, then search for the 600, 601, 650, or 560 User Guide.

Feature Key	Description
•	Allows for hands-free communication during calls.
	Mutes audio transmission locally during calls.
②	Allows users to place and receive calls through an optionally connected headset.
≈2077 ≈2078 ≈2079 ;;;;Jane	Line or Speed Dial keys with LED indicators.
Directories	Access to local directories and call lists.
Services	Access to special services. (Polycom 600/601/650 only)
Applications	Access to special applications (Polycom 560 only – Contact system administrator)
Conference	Allows setup of a 3-way local conference.
Transfer	Transfer of current call to third party.
Redial	Dials last connected party from the phone.
Hold	Places current active call in Hold state.
Do Not Disturb	Cancels ringing and directs incoming calls to your mailbox (if supported). After pressing the DND button you get a choice: apply to all lines or by line (select by line label). If no mailbox is available it rejects to the user's final call treatment). The DND is set by registration (for example, if you have repetitions of your own TN, the DND would apply to all appearances of that single extension).
Messages	Place/receive text and voice messages.
Menu	Access local and server features.
⊘ ⊗	Select and Delete controls for options and text within local menus.

Feature Key	Description
000	Scrolling of lists and control of text/number entry on display.
Call Forward My S	Soft keys to select from various context-sensitive options.
\oplus	Volume keys to adjust audio and ringer volume.
Dial Pad	General dialing and alphanumeric entry (including special characters)

Polycom 601 Unique Registrations

The following scenarios describe the number of lines and keys supported for the Polycom 601 phone.

- Polycom 601 (phone only)
 - 6 line keys
 - supports 6 unique registrations (a different extension mapped to each line if you so choose).
- Polycom 601 + 1 expansion module
 - 6 line keys for the phone + 14 line keys for the expansion module = 20 total line keys
 - supports 12 unique registrations (12 different extensions can be mapped to 12 of the 20 available keys. The remaining 8 keys could either be duplicates of a mapped extension or used for speed dial.)
- Polycom 601 + 2 expansion modules
 - 6 line keys for the phone + 14 line keys for each expansion module = 34 total line keys
 - supports 12 unique registrations (12 different extensions mapped to 12 of the 34 available keys. The remaining 22 keys could either be duplicates of a mapped extension or used for speed dial.)
- Polycom 601 + 3 expansion modules
 - 6 line keys for phone + 14 line keys for each exp module = 48 total line keys
 - supports 12 unique registrations (12 different extensions mapped to 12 of the 48 available keys. The remaining 36 keys could either be duplicates of a mapped extension or used for speed dial.)

Speed dials can be specified by the user via the phone menu interface. Registered Lines will be provisioned first, followed by whatever speed dials are specified. Therefore, if MLA lines are added via the Administrator Tool and the phone is then rebooted, any specified speed dials will move down on the phone, if space is available.

Polycom 650 Unique Registrations

The following scenarios describe the number of lines and keys supported for the Polycom 650 phones.

- Polycom 650(phone only)
 - 6 line keys
 - supports 6 unique registrations (a different extension mapped to each line if you so choose).
- Polycom 650 + 1 expansion module
 - 6 line keys for the phone + 14 line keys for the expansion module = 20 total line keys
 - supports 20 unique registrations (20 different extensions can be mapped to the 20 available keys.)
- Polycom 650 + 2 expansion modules
 - 6 line keys for the phone + 14 line keys for each expansion module = 34 total line keys
 - supports 24 unique registrations (24 different extensions mapped to 24 of the 34 available keys. The remaining 10 keys could either be duplicates of a mapped extension or used for speed dial.)
- Polycom 650 + 3 expansion modules
 - 6 line keys for phone + 14 line keys for each exp module = 48 total line keys
 - supports 24 unique registrations (24 different extensions mapped to 24 of the 48 available keys. The remaining 24 keys could either be duplicates of a mapped extension or used for speed dial.)

Speed dials can be specified by the user via the phone menu interface. Registered Lines will be provisioned first, followed by whatever speed dials are specified. Therefore, if MLA lines are added via the Administrator Tool and the phone is then rebooted, any specified speed dials will move down on the phone, if space is available.

Special Notes for Polycom 601/650/560

 For all Polycom phones - the handset plug on the Polycom phones has a 2 step plug in. When plugging the cord into the handset you should be able to hear two clicks (2 "Click" process) for the handset to work properly.

View Call Lists - Polycom 601/650/560

On the Phone:

A local list of calls missed, received, and placed is maintained by the phone. To view the log on the phone:

- 1. Press the Directories button followed by Call Lists and Missed, Received, or Placed Calls as desired. Call information will be displayed.
- 2. Choose Dial to return the call.
- 3. Choose Directories to return to the idle display.

To quickly view respective call lists from the idle display:

- Press for Placed Calls.
- Press or Received Calls.
- Press of for Missed Calls.

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

Polycom 4000/6000

■NOTE:

For more information about the Polycom 4000/6000 phone, go the Polycom web site at http://www.polycom.com/ and locate the Polycom 4000 or 6000 User Guide.

Feature Key	Description
Menu	Displays a menu of settings and options.
Exit	Exits from current screen to previous screen.
	The screen will display labels for these keys to identify their context-sensitive functions.
Polycom Polycom 4000 6000	Originates and end calls, answers incoming calls
Polycom Polycom 4000 6000	Dials last dialed party.
Dial Pad	These 12 keys provide the 10 digits, the 26 alphabetical characters and special characters available in context-sensitive applications.
•	Mutes your phone by toggling the microphone on and off during a conservation.
1	Use these to adjust the ringer and audio volume.
Select	Choose a menu item
9	Navigate up or down through displayed lists.

View Call Lists – Polycom 4000/6000

A local list of calls missed, received, and placed is maintained by the phone.



Select Features, followed by Call Lists, and Missed, Received, or Placed Calls as desired. Call information will be displayed.

■NOTE:

To quickly view the Missed Calls list from the idle display, press Choose:

- Edit to amend the dial string if necessary.
- Dial to return the call.
- 3. Press (for the Polycom 4000) or press (for the Polycom 6000) to return to the idle display

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

LG Phones

Important Information for LG Phones

- The LG 6830 has 5 additional buttons that the LG 6812 doesn't have: CONF, INFO, Hold, MSG, and DND.
- The LG Phones do not have electronic labels, (e.g., you will not see the label field on the MLA pages). The user must manually label the line keys on the phone.
- Ability for the LG phones users to add to the phone book will be available in a future release. This ability is currently available to the Administrator. Up to 20 Personal Speed Dial entries can be entered via the AT&T Voice DNA Personal Web Site.
- The LG phones can have up to 5 appearances of any given line. This limits a line to five repetitions on a given LG phone.

Display Date and Time on a LG Phone

Voice DNA provides the ability for the customer to display the date and time on the phones (the default state for time display is turned **ON**). The Time and date display will need to be manually updated for Voice DNA remote workers.

Instructions on how Remote Workers can update their supported LG phone to display the date and time is included in the following section.

Procedure for a Remote Worker to Display Date and Time on LG Phones

- 1. Press the **Settings** hard button to enter the **Configuration Menu**.
- Select Phone Settings and press the OK soft button.
- 3. Select **Time Configuration** and press the **OK** soft button.
- 4. Select **SNTP Server Address** and press the **OK** soft button.
- Enter the SNTP Server Address or FQDN to be used and press the OK soft button.
 - 0.attvdna.pool.ntp.org
 - 1.attvdna.pool.ntp.org
 - 2.attvdna.pool.ntp.org
 - 3.attvdna.pool.ntp.org
- 6. Scroll to **Time Zone** and press the **OK** soft button.
- Use the Prev and Next soft buttons to display the desired Time Zone value and then press the OK soft button.
- **8.** Pressing the **Settings** hard button exits all of the menus. You do need to reboot the phone manually in order for the changes to take effect.

Entering Optional Account Codes with LG Phones

To make a call using Optional Account Codes using an LG phone, perform the following:

- 1. Press *50, followed by the Account Code (e.g., 1234).
- 2. Wait for call to be made or press the [Call] softkey.
- 3. The LCD displays **[Enter Number]**. Enter the telephone number that you are trying to call and press the **[OK]** softkey.

LG 6812

■NOTE:

For more information about the LG 6812 phone, contact your AT&T Voice Administrator as they will have access to the latest guide.

Feature Key	Description
	Left direction button. When ↑ appears in the bottom-left corner of the LCD, the previous menu may be selected.
	Right direction button. When → appears in the bottomright corner of the LCD, the next group of Softkeys may be selected.
	Softkeys. Softkeys are interactive, changing function based on the LIP Phone status. When selected and verified, by pressing the [OK] Softkey, the LIP Phone performs the selected function.
Settings	Settings button. 'Settings' accesses and exits the menu for display and changes to the LIP Phone configuration.
▼ Volume ▲	Volume control button. Use to adjust Ring, Headset, Handset, and Speaker volume.
Speaker	Speaker button. Toggle the LG 6812 speakerphone On and Off.
Mute	Mute button. Toggle audio from the microphone to the connected party On and Off.
9•	Headset button. When using a headset, this button toggles the headset state. When the headset is active, this button is red.
	Flexible buttons (11) with Red/Green LEDs. Assign as Line or Feature using LCD Menu or Web Manager. Default assignments for buttons 1 and 2 are primary and secondary Line appearance for the extension.

LG 6812 Unique Registrations

The following scenarios describe the number of lines and keys supported for the LG 6812 phone.

■ LG 6812 — 11 line keys. The LG phones can have up to 5 appearances of any given line. This limits a line to five repetitions on a given LG phone.

View Call Lists - LG 6812

On the Phone:

A local list of calls missed, received, and placed is maintained by the phone. To view the log on the phone:

- 1. Press the [Clog] soft key.
- 2. Using the [Prev] and [Next] soft keys, scroll through the three call log options (Missed Calls, Received Calls, Placed Calls).
- 3. Press the [OK] soft key to select the desired call log, OR
- **4.** Enter the number to the left of the desired call log option on the phone's keypad.
- 5. Use the [Prev] and [Next] soft keys to scroll through the various log entries.

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

LG 6830

■NOTE:

For more information about the LG 6830 phone, contact your AT&T Voice Administrator as they will have access to the latest guide.

Feature Key	Description
	Left direction button. When ↑ appears in the bottomleft corner of the LCD, the previous menu may be selected.
	Right direction button. When → appears in the bottomright corner of the LCD, the next group of Softkeys may be selected.
•	Softkeys. Softkeys are interactive, changing function based on the LIP Phone status. When selected and verified, by pressing the [OK] Softkey, the LIP Phone performs the selected function.
Settings	Settings button. Settings accesses and exits the menu for display and changes to the LIP Phone configuration.
	Flexible buttons (24) with Red/Green LEDs. Assign as Line or feature in Phone configuration. Default assignments for buttons 1 and 2 are primary and secondary Line appearance for the extension.
9	Headset button. When using a headset, this button toggles the headset state. When the headset is active, this button is red.
CONF	CONF button. Use to establish a conference.
INFO	INFO button. Use to display configuration settings of the LIP-6830.
Hold	Hold button. Use to place a call on Hold. Also, use to access a held call.

Feature Key	Description
MSG	MSG button. When the MWI LED indicates you have a message waiting. Use to access Voice Mail server.
DND	DND (Do-Not-Disturb) button. Use to activate DND so that extension will not ring.
Ø Mute	Mute button. Toggle audio from the microphone to the connected party On and Off.
Speaker	Speaker button. Toggle the LIP-6830 speakerphone On and Off.
▼ Volume ▲	Volume control button. Use to adjust Ring, Headset, Handset, and Speaker volume.

LG 6830 Unique Registrations

The following scenarios describe the number of lines and keys supported for the LG 6830 phone.

LG 6830 (phone only):

- 24 line keys. The LG phones can have up to 5 appearances of any given line. This limits a line to five repetitions on a given LG phone.
 - supports 24 unique registrations (a different extension mapped to each line if you so choose).

View Call Lists – LG 6830

On the Phone:

A local list of calls missed, received, and placed is maintained by the phone. To view the log on the phone:

- 1. Press the [Clog] soft key.
- 2. Using the [Prev] and [Next] soft keys, scroll through the three call log options (Missed Calls, Received Calls, Placed Calls).
- 3. Press the [OK] soft key to select the desired call log, OR
- **4.** Enter the number to the left of the desired call log option on the phone's keypad.
- 5. Use the [Prev] and [Next] soft keys to scroll through the various log entries.

On the Personal Web Site Home Page:

Missed calls will be included on the **Incoming Calls** tab on the home page. Missed calls will have a status of **Missed**. Click on the number to dial the missed call.

or

Missed calls are included on the **Calls Logs** menu item on the home page. Select the **Call Logs** menu item and select **Missed** from the drop down menu. Click on the number to dial the missed call.

Caller ID Scenarios

Scenario 1: A calls B, B blind transfers call to C

- a. A = PSTN Caller, B = Voice DNA User with 10 digit TN, C = Voice DNA User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on C's phone: B's TN and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.
- A = PSTN Caller, B= Voice DNA User with Extension only, C = Voice DNA User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on C's phone: B's extension and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.
- A = Voice DNA Caller, B = Voice DNA User with 10 digit TN, C = Voice DNA User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: B's TN and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.
- d. A = Voice DNA Caller, B= Voice DNA User with Extension only, C = Voice DNA User
 - Caller ID on B's phone: A's TN and Voice DNA Company Name
 - Caller ID on C's phone: B's extension and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.
- e. A = PSTN Caller, B = Voice DNA User with 10 digit TN, C = PSTN User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on C's phone: B's TN and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.
- f. A = PSTN Caller, B= Voice DNA User with Extension only, C = PSTN User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on C's phone: B's DCN and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.

- g. A = Voice DNA Caller, B = Voice DNA User with 10 digit TN, C = PSTN User
 - Caller ID on B's phone: A's TN and Voice DNA Company Name
 - Caller ID on C's phone: B's TN and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.
- h. A = Voice DNA Caller, B= Voice DNA User with Extension only, C = PSTN User
 - Caller ID on B's phone: A's TN and Voice DNA Company Name
 - Caller ID on C's phone: B's DCN and Voice DNA Company Name when Phone C is ringing, A's TN and name (varies depending on scenario) when Phone C answers.

Scenario 2: A calls B, B performs consultative transfer to C, then drops off the call

Caller ID on B's phone: A's TN and name (if available)

Caller ID on C's phone: A's TN and name (if available)

- a. A = PSTN Caller, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA or PSTN User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on C's phone: A's TN and name (if available)
- b. A = Voice DNA Caller with 10 digit TN, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: A's TN and name
- c. A = Voice DNA Caller with Extension only, B = Voice DNA User (Extension-only or with 10 digit TN), C = PSTN User
 - Caller ID on B's phone: A's Extension and name
 - Caller ID on C's phone: DCN of A's location and Voice DNA Company Name

Scenario 3: A calls B, B has call forwarding set to C

- a. A = PSTN Caller, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA or PSTN User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on C's phone: A's TN and name (if available)
- b. A = Voice DNA Caller with 10 digit TN, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: A's TN and name
- c. A = Voice DNA Caller with Extension-only, B = Voice DNA User (Extension-only or with 10 digit TN), C = PSTN User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: DCN of A's location and Voice DNA Company Name
- d. A = Voice DNA Caller with Extension-only, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: A's extension and name

Scenario 4: A calls B, B Transfers to C, C has call forwarding set to D

- a. A = PSTN Caller, B = Voice DNA User (Extension-only or with 10 digit TN), C
 = Voice DNA or PSTN User, D = Voice DNA or PSTN User
 - Caller ID on B's phone: A's TN and name (if available)
 - Caller ID on D's phone: A's TN and name (if available)
- b. A = Voice DNA Caller with 10 digit TN, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA or PSTN User, D = PSTN User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: A's TN and Voice DNA Company Name
- c. A = Voice DNA Caller with Extension-only, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA or PSTN User, D = PSTN User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: DCN of A's location and Voice DNA Company Name

- d. A = Voice DNA Caller with Extension-only, B = Voice DNA User (Extension-only or with 10 digit TN), C = Voice DNA User, D = Voice DNA User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: A's extension and name
- e. A = Voice DNA Caller with Extension-only, B = Voice DNA User (Extension-only or with 10 digit TN), C = PSTN User, D = Voice DNA User
 - Caller ID on B's phone: A's TN and name
 - Caller ID on C's phone: DCN of A's location and Company Name

Scenario 5: A calls B (a Direct Answer Auto Attendant CTN) and selects option that sends call to C

Caller ID on C's phone: A's TN and name

Scenario 6: A calls B (a Live Answer Auto Attendant CTN), B transfers call to C (a Direct Answer Auto Attendant) and selects option that sends call to D

- Caller ID on B's phone: A's TN and name (if available)
- Caller ID on D's phone: A's TN and name (if available)

Using Account Codes

If the account code feature has been configured by your phone manager, you can use account codes to track outside calls made from your phone for professional billing purposes. For example, account codes can be used to track the time spent on a phone consultation with a client, or to track all calls that are made in a specific business department. Account codes can be 2 to 9 digits long. The actual account codes are created by your phone manager.

Your phone manager can implement the account code feature in one of two ways:

- Mandatory account codes—you must always enter an account code before you can make an outside call, transfer an outside call to an external number, initiate an outside call using Click to Call, initiate a call using a speed dial code, or initiate a call using a feature button.
- Optional or voluntary account codes—you do not need to enter an account code but have the option to do so before you make an outside phone call, transfer a call to an external number, or initiate an outside call using Click to Call, a speed dial or feature button.

Account codes are not required for emergency calls (E911), and calls between extensions.

■NOTE:

Blind Transfer and Mandatory Account Codes are not compatible. If you attempt to Blind Transfer a call from a device that subscribes to Mandatory Account Codes and the transfer attempt is to a location that is not intratenant, the call drops.

Table 12. Using Account Codes describes how to use the account codes with different types of calls.

Table 12. Using Account Codes

Activity	Account Code	Action	
Make an outside call.	Mandatory	Dial the offnet code (e.g., 9) for an outside line, dial the external number, press <send>, enter the account code, and press <send>.</send></send>	
	Optional	For Non-LG phones: Dial *50, followed by the account code, <send>, the offnet dialing code (e.g., 9, outside line), the telephone calling number, and then press <send></send></send>	
		The <send></send> at the end (after the account code) is optional. This <send></send> , eliminates the 4 second delay before the system initiates the call.	
		For LG Phones: Press *50, followed by the Account Code (e.g., 1234). Wait for call to be made or press the [Call] softkey. The LCD displays [Enter Number]. Enter the telephone number that you are trying to call and press the [OK] softkey.	
Make an internal call.	Mandatory	Dial the extension you want to call.	
	Optional	Dial the extension you want to call.	
Transfer a call to an external number.	Mandatory	Press Conf/Trn , or the softkey below TRNS , and dial the external number. When you hear a secondary dial tone, enter a valid account code.	
		Optionally, you can press <send></send> after you have entered the account code. If you do not press <send></send> , there is a 4 second delay before the system initiates the call.	
	Optional	Press Conf/Trn, or the softkey below TRNS, then dial *50 and enter a valid account code. Dial the external number.	
Make an emergency	Mandatory	Dial 911. You do not need to enter an account code.	
call.	Optional	Dial 911 .	
Call directory assistance.	Mandatory	Dial 9 for an outside line and dial 411 . When you hear dial tone again, enter the account code.	
	Optional	Dial 411 .	
Redial a number you	Mandatory	You must completely re-enter the number.	
previously called and had already entered an account code.	Optional	Press the Redial button, or dial *00.	

Table 12. Using Account Codes

Activity	Account Code	Action	
Make a call using Click to Call.	Mandatory	Click on the phone number you want to call and choose an account code from the account code drop down list, as described in <i>AT&T Voice DNA User Guide</i> .	
	Optional	Click on the phone number you want to call and choose an account code from the account code drop down list if desired, as described in AT&T Voice DNA User Guide .	
Make a call using a feature button.	Mandatory	To use feature buttons in conjunction with mandatory account codes, you can do one of the following:	
		Assign the account code and the phone number to the feature button, as follows: <offnet code="" dialing=""><phone number=""><send><account code=""><send></send></account></send></phone></offnet>	
		For more information about assigning account codes to your feature buttons, see the <i>AT&T Voice DNA User Guide</i> .	
	Optional	To use feature buttons in conjunction with optional account codes, you must first assign the account code to the feature button, as follows:	
		*50 <offnet code="" dialing=""><send><account code=""><phone number=""></phone></account></send></offnet>	
		For more information about assigning account codes to your feature buttons, see the <i>AT&T Voice DNA User Guide</i> .	

Table 12. Using Account Codes

Activity	Account Code	Action	
Make a Conference Call using Click to Conference	Not Used	If Account Codes are not used (not enabled in your Class of Service template), you will not see the Account Code field located on the bottom-right-hand side of the On-Demand Conference Screen.	
	Mandatory	If Account Codes are Mandatory , an Account Code is required to be used for the conference. The Account Code that has been selected will be displayed and the choose an account code link will be located to the right of the Account Code field located on the bottom-right-hand side of the On-Demand Conference Screen. This Account Code will be associated with all of your conferences until you change the Account Code. You can change the Account Code by clicking on the choose an account code link. Only Account Codes that exist in the pull down menu can be used. These codes must have been entered into the pull down menu by the Administrator.	
	Optional	If Account Codes are Optional , you will see the Account Code that has been selected and the change link to the right of the Account Code on the bottom-right-hand side of the On-Demand Conference Screen. The Account Code displayed on the screen will be used for this conference. This field can be blank since the Account Code is optional. You can change the Account Code by clicking on the change link.	

■>NOTE 1:

Contact your office phone manager to find out whether your company uses account codes, and if so, whether they are mandatory or optional account codes.

■NOTE 2:

Account codes are not supported on the softphone.

Using Mandatory Account Codes

To make an outside call if the account code is mandatory, follow these steps:

- 1. Lift the handset, press **Spkr**, or press a line button.
- 2. Dial for an outside line (e.g., 9) and then enter the phone number you want to call.
- 3. Press <send>.
- 4. Enter the account code.

If the account code you enter is invalid, you will receive an error message and you will need to start over.

5. Press <send>.

■NOTE 1:

Emergency calls (E911) and internal calls override the mandatory account code feature. You do not need to enter an account code when you call 911, or when you make a call to another extension within your company.

■NOTE 2:

Blind Transfer and Mandatory Account Codes are not compatible. If you attempt to Blind Transfer a call from a device that subscribes to Mandatory Account Codes and the transfer attempt is to a location that is not intratenant, the call drops.

Using Optional Account Codes (Non-LG Phones)

To make an outside call if account codes are optional, follow these steps:

- 1. Lift the handset, press **Spkr**, or press a line button.
- 2. Dial *50.
- 3. Enter the account code.
- 4. Enter <send>.
- 5. Enter the Offnet code for an outside line (e.g., 9).
- **6.** Enter the phone number you want to call.
- 7. Optionally, you can press <send> after you have entered the account code. If you do not press <send>, there is a 4 second delay before the system initiates the call.

Entering Optional Account Codes with LG Phones

To make a call using Optional Account Codes using an LG phone, perform the following:

- 1. **Press *50**, followed by the **Account Code** (e.g., 1234).
- 2. Wait for call to be made or press the [Call] softkey.

3. The LCD displays **[Enter Number]**. Enter the telephone number that you are trying to call and press the **[OK]** softkey.

Feature (Star) Codes

Table 13. Feature (Star) Codes

Code	Description	Dial string
*00	Redials the last number called.	*00 <send></send>
*01 – *20	Reserved for personal speed dial numbers.	*xx <send> (xx 01-20)</send>
*21 – *49	Reserved for company speed dial numbers.	*xx <send> (xx 21-49)</send>
*50	Turns on the account code feature for this call only.	See Using Account Codes .
*53	Activates the Directed Call Pickup feature, which lets you retrieve a call that is ringing at another extension by dialing *53 followed by the extension.	*53 <ext> <send></send></ext>
*54	Activates the Group Call Pickup feature, which lets you pick up the first of any calls currently ringing at any extension in your call group.	*54 <ext> <send></send></ext>
*62	Activates the call forwarding busy feature. This feature lets you forward all incoming calls to a specified number when your desk phone is busy. Press 62 <number>#, where <number> is the internal or external number to which you want the calls forwarded</number></number>	*62 <number> <send></send></number>
*63	Turns off the call forwarding busy feature.	*63 <send></send>
*66	Auto Call Back Busy allows you to automatically call back a previously called busy line when the line becomes idle. Whenever the called number becomes idle, the call is connected to the original caller.	*66 after terminating the busy call *86 to cancel
*67	Blocks your number from being transmitted on any outgoing call by pressing *67 before dialing the number.	*67 <number> <send></send></number>
*69	Redials the last incoming number when the phone is busy.	*69 <send></send>

Table 13. Feature (Star) Codes

Code	Description	Dial string	
*70	Enables or disables Call Waiting for the current call only.	*70 <send></send>	
*72	Activates the call forwarding feature. This feature lets you forward all incoming calls to another number. Press *72 <number>, where <number> is the internal or external number to which you want all calls are forwarded</number></number>	*72 <number> <send></send></number>	
*73	Turns off the call forwarding feature.	*73 <send></send>	
*77	Executive Busy Override (EBO) allows the	*77 <mode digit=""><extension></extension></mode>	
	user to insert them self into an existing two-party call. This feature is often called	The EBO modes are:	
	barge in. The initiator must have the	Digit Mode	
	correct class of service settings in order to use these features.	1 3-Way	
	3-way — The feature initiator enters an	4 Override	
	existing two-way call, turning it into a 3-way conference.	This capability must be enabled on the user's COS settings.	
	Override — The feature initiator can disconnect an ongoing call on the target's phone and have themselves connected to the target. If the target is not on an active call, the caller can by-pass all call forwarding and do not disturb settings on the target's phone.	EBO exemption is a another COS setting which allows the Administrator to mark an extension as exempt from bargein.	
		EBO is not supported for initiators behind an ATA. If *77 is dialed from behind an ATA with the expectation of EBO service, the initiator will now have anonymous calls rejected. *87 should be dialed to clear the condition.	
*80	Lets you make an external call on a phone that allows internal calls only. You must dial *80 followed by your extension number, your authorization pin and the phone number you want to call, as follows: *80* <extension>*<pin>*<number></number></pin></extension>	*80* <extension>*<pin>* <number></number></pin></extension>	
*82	For users with caller ID blocked. Press *82 before dialing to selectively unblock the caller ID on any outgoing call.	selectively unblock the	

Table 13. Feature (Star) Codes

Code	Description	Dial string	
*86	Used to cancel Auto Call Back Busy	*86	
*90	Transfer to third party's VM directly.	*90 <ext><send></send></ext>	
*92	Activates the call forwarding no answer feature. This feature lets you forward all incoming calls to another number when you do not answer your desk phone. Press *92 <number>#, where <number> is the internal or external number to which you want all calls are forwarded.</number></number>	*92 <number> <send></send></number>	
*93	Turns off the call forwarding no answer feature.	*93 <send></send>	
*95	Remove MWI	Not supported	
*96	Lets you use your telephone as an intercom	*96 <extn> <send></send></extn>	
*98	Parks a call on a specified number.	*98 <ext> <send></send></ext>	
*99	Picks up a call parked on your own number.	*99 <send></send>	

■> NOTE 1:

Feature (Star) codes cannot be dialed via the AT&T Voice DNA Personal Web Site (e.g., quick call or on click to conference).

■>NOTE 2:

If a quick call or click to conference is used, the service will establish the first part of the call and then tries the far end. If the far end is unavailable (busy or ring no answer), or if the number entered is invalid, the only indication will be that the call does not connect. You will not hear a busy signal or announcement.

Intercept Announcements

■NOTE:

These announcements are system wide and are not configurable per tenant.

Table 14. Intercept Announcements

Intercept Description	Intercept Announcement	
Call setup failed due to network condition or resources.	We are sorry. Your call cannot be completed at this time. Please try your call again later.	
Call restriction enabled for the dialed number.	Your call cannot be completed because your service has been restricted. Please contact your AT&T Voice DNA administrator for assistance.	
Call setup failed because dialed phone number is incomplete or not recognized.	We are sorry. Your call cannot be completed as dialed. Please check the number and try your call again.	
Call setup failed due to network condition or resources.	We are sorry. The party you called is either not answering or temporarily unavailable. Please try your call again later.	
Call setup failed due to network condition or resources.	We are sorry. Your call did not go through. Please try your call again later.	
Caller ID block or User Controlled Line Identification Restriction feature code (*) code failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to block your caller ID. Please contact your AT&T Voice DNA administrator for assistance.	
Caller ID Unblock * code failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to override caller ID blocking. Please contact your AT&T Voice DNA administrator for assistance.	
Intercom feature code (*) failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to use the intercom capability. Please contact your AT&T Voice DNA administrator for assistance.	
Call Pickup failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to use call pickup. Please contact your AT&T Voice DNA administrator for assistance.	
Directed or group call pickup feature code (*) failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to use directed or group call pickup. Please contact your AT&T Voice DNA administrator for assistance.	
Call pickup, directed or group call pickup failed because the park number provided does not exist.	We are sorry. Call pick up could not be completed because there is no call parked to the number provided.	
Last Call Return * code failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to use last call return. Please contact your AT&T Voice DNA administrator for assistance.	

Table 14. Intercept Announcements

Intercept Description	Intercept Announcement	
Last Call Return * code failed. Caller's ID is not available.	We are sorry. We were unable to invoke call return because the caller ID is unavailable.	
Do not disturb activation * code processed successfully. Feature is active.	Do not disturb activated.	
Do not disturb deactivation * code processed successfully. Feature is not active.	Do not disturb de-activated.	
Call failed because mandatory account * code is required when making outgoing calls.	We are sorry. Your call cannot be completed without an account code following the number you are dialing. Please try your call again or contact your AT&T Voice DNA administrator for assistance.	
Account code entered with the mandatory account * code is not in the tenant's account code list.	We are sorry. The account code you provided was not recognized. Please check the account code and try your call again or contact your AT&T Voice DNA administrator for assistance.	
Optional account * code failed. Feature is disabled in the user's class of service.	We are sorry. You are currently not allowed to use an account code. Please contact your AT&T Voice DNA administrator for assistance.	
Account code entered with the optional account * code is not in the tenant's account code list.	We are sorry. The account code you provided was not recognized. Please check the account code and try your call again or contact your AT&T Voice DNA administrator for assistance.	
Outbound Call Barring * code failed. Feature not enabled in the user's class of service.	We are sorry. Outbound Call Barring is not supported.	
Outbound Call Barring activation * code processed successfully. OCB is active.	Outbound call barring activated. To de-activate, please hang up and dial the outbound call barring de-activation code.	
Outbound Call Barring deactivation * code processed successfully. OCB is not active.	Outbound call barring de-activated. To activate, please hang up and dial the outbound call barring activation code.	
Outbound Call Barring activation * code failed.	We are sorry. We were unable to activate outbound call barring. Please check the access code and the personal identification number you dialed and try again.	
Outbound Call Barring deactivation * code failed.	We are sorry. We were unable to de-activate outbound call barring. Please check the feature code and try again.	
Call forward on busy activation * code processed successfully. Call forward if busy is active.	Call forwarding on busy activated. To de-activate, please hang up and dial the call forwarding on busy de-activation code.	

Table 14. Intercept Announcements

Intercept Description	Intercept Announcement
Call forward on busy deactivation * code processed successfully. Call forward if busy is not active.	Call forwarding on busy de-activated. To activate, please hang-up and dial the call forwarding on busy activation code.
Call forward unconditional activation * code processed successfully. Call forward unconditional is active.	Unconditional call forwarding activated. To deactivate, please hang-up and dial the unconditional call forwarding activation code.
Call forward unconditional deactivation * code processed successfully. Call forward unconditional is not active.	Unconditional call forwarding de-activated. To activate, please hang-up and dial the unconditional call forwarding de-activation code.
Call forward no answer activation * code processed successfully. Call forward no answer is active.	Call forwarding on no answer activated. To deactivate, please hang-up and dial the call forwarding on no answer de-activation code.
Call forward no answer deactivation * code processed successfully. Call forward unconditional is not active.	Call forwarding on no answer de-activated. To deactivate, please hang-up and dial the call forwarding on no answer activation code.
Call forwarding * code failed. Feature is not enabled in the user's class of service.	We are sorry. You are currently not allowed to use call forwarding. Please contact your AT&T Voice DNA administrator for assistance.
Caller with caller ID blocking enabled places a call to a Synergy user.	We are sorry. The party you have called does not accept anonymous calls. Please unblock your caller ID and try your call again.
Call forwarding activation * code failed. May be due to incorrect feature code sequence or the feature may not be active in the user's class of service.	We are sorry. We were unable to activate call forwarding. Please try again. If the problem persists, please contact your AT&T Voice DNA administrator for assistance.
Call forwarding de-activation * code failed. May be due to incorrect feature code sequence.	We are sorry. We were unable to de-activate call forwarding. Please try again. If the problem persists, please contact your AT&T Voice DNA administrator for assistance.
Call park failed. Feature is not enabled in the user's class of service.	We are sorry. You are currently not allowed to park calls. Please contact your AT&T Voice DNA administrator for assistance.
Call park feature failed.	We are sorry your call park request could not be completed. Please try again. If the problem persists, please contact your AT&T Voice DNA administrator for assistance.
Call park feature successful.	Call has been parked.

Table 14. Intercept Announcements

Intercept Description	Intercept Announcement	
Caller ID Blocking activation * code failed. May be due to incorrect feature code sequence or the feature may not be active in the user's class of service.	We are sorry. We were unable to complete your request to block caller ID. Please try again. If the problem persists, please contact your AT&T Voice DNA administrator for assistance.	
User Controlled Line Identification Restriction (UCLIR) * activation code processed successfully. Feature is active.	Caller ID blocked for all calls.	
UCLIR is active and the Caller ID Block per Call * code processed successfully.	Caller ID blocked for this call.	
User Controlled Line Identification Restriction (UCLIR) * deactivation code failed. May be due to incorrect feature code sequence or the feature may not be active in the user's class of service.	We are sorry. We were unable to complete your request to unblock caller ID. Please try again. If the problem persists, please contact your AT&T Voice DNA administrator for assistance.	
Speed dial assignment feature code (*) failed.	We are sorry. We were unable to set up the speed dial entry. Please check your entry and try again.	
Speed dial assignment feature code (*) processed successfully.	Speed dial entry accepted.	
Speed dial assignment feature code (*) failed.	We are sorry. We were unable to reset the speed dial entry. Please check your entry and try again.	
Speed dial assignment feature code (*) processed successfully.	Speed dial entry has been reset.	
Authorization code feature code (*) failed. Feature is not enabled in the user's class of service.	We are sorry. You are currently not allowed to use authorization codes. Please contact your AT&T Voice DNA administrator for assistance.	
Authorization code feature code (*) failed because of an invalid PIN number.	We are sorry. The authorization code you entered was not recognized. Please check the number and try again.	

Long Duration Call Problem

For calls that have a long duration (4 hours), it's possible that the call can be disconnected. This only applies to calls that originate from certain features, such as transfers back to Voice DNA for Call Tree (Auto Attendant) and Mail Messages Reply to Sender. Though any call of a long term duration that involved these features, will be affected (e.g., Click to Conference).

Mac/PC Differences

The following information was determined using:

- Safari browser running on Mac computer only.
- Configurations tested:
 - Mac OS X version 10.4.10, 10.5.2 & 10.5.3
 - Safari browser version 2.0.4 & 3.1.1
 - Mac laptop and desktop
- AT&T Voice DNA Personal Web Site only (accessed via BusinessDirect portal).
 - Not supported Outlook Integration

Table 15. Mac/PC Differences

Function	Mac (Safari)	PC (Internet Explorer)
Browser window refresh	Cmd – R or the Refresh icon on the browser screen.	F5 or the Refresh icon on the browser screen.
Buttons (Save, Delete, Previous, Next, etc)	Not grayed out on Safari when they are not applicable.	Grayed out on IE when they are not applicable.
Listening to messages	Messages played through iTunes by default but can set to using other media player (e.g., RealPlayer, QuickTime).	Windows Media Player
Logging Off	Safari must be closed (Quit Safari). Closing screen by using the red dot does not close the session. User would be taken directly back in upon opening browser if Safari is not quit (native to Safari).	Use the Close link to close the Personal Web Site.
Tab key	Safari default is fillable fields. Can set to Press Tab to highlight each item on a web page under Preferences -> Advanced.	Moves to next field.

Per Call Caller ID Blocking Behavior

Table 16. Per Call Caller ID Blocking (*67) Behavior

Dial String	Per Call Blocking all calls Caller ID block (per call)+ Caller ID block (intratenant)	Per Call Blocking Extratenant Calls only Caller ID block (per call) only	No Per call Blocking Enabled
Dial extra-tenant number	Caller ID is passed	Caller ID is passed	Caller ID is passed
Dial intra-tenant number	Caller ID is passed	Caller ID is passed	Caller ID is passed
Dial *67 + extra-tenant number (not 911)	Caller ID is blocked	Caller ID is blocked	Call Fails (if CIDBLK is disabled, *67 string will fail)
Dial *67+ intra-tenant number (not VM access number or 911)	Caller ID is blocked	Caller ID is passed (we do not have the ability to fail the call in this case).	Call Fails (if CIDBLK is disabled, *67 string will fail)
Dial *67 + 911 (both public and tenant routed 911)	Caller ID is passed	Caller ID is passed	Call Fails (if CIDBLK is disabled, *67 string will fail)
Dial *67 + VM access number	Caller ID is passed	Caller ID is passed	Call Fails (if CIDBLK is disabled, *67 string will fail)

Table 17. All Call Caller ID Blocking with Override (*82) Behavior

Dial String	Always Block Caller ID – all calls Caller ID Block (all calls with override) + Caller ID Block (intratenant)	Always Block Caller ID – Extra-tenant Calls only Caller ID Block (all calls with override) only	No Always Block Caller ID
Dial extra-tenant number (not 911)	Caller ID is blocked	Caller ID is blocked	Caller ID is passed
Dial intra-tenant number (not 911 or VM access number)	Caller ID is blocked	Caller ID is passed	Caller ID is passed
Dial *82 + extra-tenant number	Caller ID is passed	Caller ID is passed	Caller ID is passed (the *82 dialing is superfluous).
Dial *82+ intra-tenant number	Caller ID is passed	Caller ID is passed (the *82 dialing is superfluous).	Caller ID is passed (the *82 dialing is superfluous).
Dial 911 (both public and tenant routed 911)	Caller ID is passed	Caller ID is passed	Caller ID is passed
Dial VM access number	Caller ID is passed	Caller ID is passed	Caller ID is passed

Fixed Destination Phones

A fixed-destination (ring-down), phone calls a designated fixed phone number when a caller picks up the phone's handset. For example, if your company has a phone at a secured entrance to the company premises, you can configure the phone to ring the security desk whenever a visitor picks up the handset.

To configure a ring-down phone, the Administrator must create an extension and assign ring-down phones to the extension. The Administrator can change an existing extension to a ring-down extension.

The Administrator can assign multiple phones to the extension, but the phones must all be configured as ring-down phones. You can't have both a ring-down phone and a regular phone on a single extension.

Messages Telephone User Interface (TUI)

AT&T Voice DNA Mailbox allows you to retrieve your messages, making them accessible to you from your AT&T Voice DNA Personal Web Site, any Touch-Tone® telephone, or your AT&T wireless phone. Additionally, since notification of messages waiting is sent to your phone, you are always in touch with your messages.

The following are some of the most important features of AT&T Voice DNA Mailbox:

- Hear your voicemail messages on your PC, over the telephone, or a wireless phone.
- View your fax messages on your PC
- Receive notification of messages waiting, via your telephone.
- Reduce your chances of missed messages.

Getting Started With Your AT&T Voice DNA Mailbox

Important Items Which You Need

The following lists the items that you will need before you start setting up your mailbox:

- Your Access Number The telephone number (provided by your Administrator) which allows you to access your messages. This telephone number is the number that when called will accept messages.
- Your **Temporary PIN** Provided by your Administrator. You should change this PIN, the first time you log into your mailbox.
- Your Mailbox Number Telephone number associated with your mailbox. This is your personal Voice DNA number (Office Telephone Number) that customers will use to reach you.

Setting Up Your Mailbox From Your Office Phone Number

The first time you call in, if from your office phone number:

- Dial your access number. The access number should be provided by your AT&T Voice DNA Administrator. Also, if you have access to the AT&T Voice DNA Personal Web Site, the access number is displayed on the My Setting → Mailbox Settings page.
- 2. Enter your **Temporary PIN** and press #.
- 3. When prompted, enter a new PIN and press ∰. Your new PIN must be from 6 to 10 digits in length. If you make a mistake, press ∰ to start again. For security reasons, you should change your Temporary PIN immediately, even if you are not yet using your mailbox. When choosing your PIN, you should NOT:

- Use your telephone or mailbox number (or any part of the telephone or mailbox number) as part of the PIN.
- Repeat digits (e.g., 444444).
- Use sequential digits (e.g., 345678).
- Use easily identifiable numbers (e.g., ZIP code, street address, etc.).

Treat your mailbox PIN as you would your ATM PIN. For added security, change your PIN periodically.

- **4**. Re-enter your new PIN followed by ## to confirm.
- 5. The system requests that you set up an authentication code. This authentication code is used to reset your PIN should you forget your PIN in the future. You may choose option ① to set up the authentication code or option ② to choose to set up the authentication code at a later time. This is a good time to enable this feature during your initial set up as it allows you to reset your PIN should you forget it in the future.

To set up the **Authorization Code**, press ①.

- Press 1 to use your mother's date of birth.
- Press 2 to use your father's date of birth.
- Press 3 to use your spouse's date of birth.
- Press 4 to use your child's date of birth.
- 6. Next, depending on the option you chose in **Step 5**, enter the chosen date as 8 digits. For example, July 10, 1950 should be entered as 07101950.
- 7. Re-enter the date. If successful, you will be told that your **Authorization Code** has been saved.
- 8. Next, say your name after the tone and press #. To use this recorded name, press 1.

To re-record your name, press ②. If you need more time, press ⑧ to pause for up to twenty seconds. Press any key to continue.

- 9. You will now be prompted to record your Personal Greeting or to select a Prerecorded greeting for your office phone number. This greeting is what callers will hear when you do not answer your phone. You have the following options:
 - To record a personal greeting, press \square and begin speaking after the tone. When you finish speaking press #.
 - To use the recorded greeting, press ①.
 - To re-record the greeting, press 2.
 - To use another type of pre-recorded greeting press 3.
 - If you need more time, press [®] to pause for up to twenty seconds. Press any key to continue.

- **10.** After recording your greeting, you'll hear some **helpful hints** on how to use and to set options for your mailbox. To skip the **helpful hints** press # to go to **Main Menu**.
- 11. You are now taken to the **Main Menu**. This is the **home** for your mailbox when navigating around your mailbox.
- 12. Hang up to leave your mailbox. Your mailbox is now ready to use.

Setting Up Your Mailbox From Another Telephone

If you are calling from a telephone number other than the one that is connected to your AT&T Voice DNA Mailbox to check for messages, the first time you call in:

- Dial your access number. The access number should be provided by your AT&T Voice DNA Administrator. Also, if you have access to the AT&T Voice DNA Personal Web Site, the access number is displayed on the My Setting → Mailbox Settings page.
- 2. Enter your phone number (mailbox number) and then press #.
- 3. Enter your **Temporary PIN** and press #.
- 4. When prompted, enter a new **PIN** and press ∄. Your new PIN must be from 6 to 10 digits in length. If you make a mistake, press ∄ to start again. For security reasons, you should change your Temporary PIN immediately, even if you are not yet using your mailbox. When choosing your PIN, you should **NOT**:
 - Use your telephone or mailbox number (or any part of the telephone or mailbox number) as part of the PIN.
 - Repeat digits (e.g., 444444).
 - Use sequential digits (e.g., 345678).
 - Use easily identifiable numbers (e.g., ZIP code, street address, etc.).

Treat your mailbox PIN as you would your ATM PIN. For added security, change your PIN periodically.

- **5.** Re-enter your new PIN followed by # to confirm. You will be prompted to record your name.
- 6. The system requests that you set up an authentication code. This authentication code is used to reset your PIN should you forget your PIN in the future. You may choose option ① to set up the authentication code or option ② to choose to set up the authentication code at a later time. This is a good time to enable this feature during your initial set up as it allows you to reset your PIN should you forget it in the future.

To set up the **Authorization Code**, press ①.

- Press 1 to use your mother's date of birth.
- Press 2 to use your father's date of birth.

- Press 3 to use your spouse's date of birth.
- Press 4 to use your child's date of birth.
- 7. Next, depending on the option you chose in **Step 5**, enter the chosen date as 8 digits. For example, July 10, 1950 should be entered as 07101950.
- 8. Re-enter the date. If successful, you will be told that your **Authorization Code** has been saved.
- 9. Next, say your name after the tone and press $\mbox{$\#$}$. To use this recorded name, press $\mbox{$\widehat{\square}$}$.

To re-record your name, press 2. If you need more time, press 8 to pause for up to twenty seconds. Press any key to continue.

- **10.** You will now be prompted to record your Personal Greeting or to select a Prerecorded greeting for your office phone number. This greeting is what callers will hear when you do not answer your phone. You have the following options:
 - To record a personal greeting, press \square and begin speaking after the tone. When you finish speaking press #.
 - To use the recorded greeting, press ①.
 - To re-record the greeting, press 2.
 - To use another type of pre-recorded greeting press 3.
 - If you need more time, press
 to pause for up to twenty seconds. Press any key to continue.
- 11. After recording your greeting, you'll hear some **helpful hints** on how to use and to set options for your mailbox. To skip the **helpful hints** press # to go to **Main Menu**.
- **12.** You are now taken to the **Main Menu**. This is the **home** for your mailbox when navigating around your mailbox.
- 13. Hang up to leave your mailbox. Your mailbox is now ready to use.

Adding AT&T Wireless Phones to Your Mailbox

As an AT&T Voice DNA mailbox user, you can connect up to two AT&T wireless phones to your VDNA mailbox.

■NOTE 1:

This feature is available only if you have the voice mail feature through AT&T Voice DNA. The wireless phone must be in the same local service area as the mailbox. Only wireless telephones from AT&T can be added to your AT&T Voice DNA mailbox. This service is not available in all areas. Your AT&T Voice DNA Administrator can contact AT&T Voice DNA Customer Care or your AT&T Account Representative to check on availability in your area.

■NOTE 2:

Please review and delete any existing wireless messages and greetings on your wireless phone before adding it to your mailbox. Existing messages and greetings are **deleted and not retrievable** upon adding the phone to your mailbox. See the **Greetings** section for more information on greetings.

- 1. Access your AT&T Voice DNA mailbox.
- 2. If Autoplay is on, press # to access the Main Menu.
- 3. At the Main Menu, press 4. for Mailbox Settings.
- 4. Press 2 for Administrative Options.
- Press 6 for Additional Settings.
- **6.** Press \Box . to add a wireless phone to your mailbox.
- 7. When prompted, enter your wireless number and press #. Then, for verification purposes, enter the last four digits of your Social Security Number or Tax ID.

Your AT&T Voice DNA access number is now the mailbox access number for your wireless phone. Your access number will also be the call forwarding number for all busy/no answer calls to your wireless phone.

■NOTE 3:

It is recommended that you place a test call to your mailbox to ensure that set up was successful. If set up was not successful, repeat steps 0 through 7 or call Customer Care for assistance.

■NOTE 4:

The last four digits of the Social Security Number or Tax ID entered in Step #7 must match those associated with the account of the wireless phone being added.

8. Once the account has been associated with your AT&T Voice DNA account, you will need to dial into the voice mail system using the wireless phone to set up your PIN, record your name, and select a greeting. You can select the same greeting associated with your AT&T Voice DNA account or record another separate one to use. Once setup has been completed, the wireless phone message count will be in sync with the new message count on your AT&T Voice DNA phone. If you have chosen to use the Message Waiting Indicator (MWI), all new messages left for either phone will result in a MWI light on the VDNA phone and a Message Indicator on the associated wireless phone.

Adding a Second AT&T Wireless Phone to Your Mailbox

Follow Steps 1 through 8 in the section "Adding AT&T Wireless Phones to Your Mailbox."

Removing an AT&T Wireless Phone From Your Mailbox

- 1. Access your AT&T Voice DNA mailbox.
- 2. If Autoplay is on, press # to access the Main Menu.
- 3. At the Main Menu, press 4. for Mailbox Settings.
- 4. Press 2 for Administrative Options.
- 5. Press 6 for Additional Settings.
- 6. Press 1 for Wireless Number.
- 7. Select and continue to follow the prompts to remove the wireless phone from your mailbox.
- 8. You will get a confirmation upon successful removal of the wireless number. The Mailbox Access Number will have changed to the access number for the AT&T Wireless phone. Power off and then power back on the wireless phone.
- **9.** Once the wireless phone has been powered back on, log in to your mailbox to setup the PIN, record your name and set up a greeting.
- 10. When the mailbox setup is complete, place a call to the wireless phone and leave a message. This will sync the phone to its voicemail system. Once you leave a message, the wireless phone should signal that the message has been received.

Accessing Your Mailbox

Once your AT&T Voice DNA Mailbox has been configured, you can access the mailbox using a Touch-Tone® or wireless phone. The various features of your mailbox and how to use them are explained in the following sections.

How to Access Your Mailbox

- Dial your access number for access to your mailbox from the telephone that is connected to your AT&T Voice DNA Mailbox. The access number should be provided by your AT&T Voice DNA Administrator. Also, if you have access to the AT&T Voice DNA Personal Web Site, the access number is displayed on the My Setting

 Mailbox Settings page.
- 2. Enter the PIN that you created during the setup of your mailbox and press #. If you haven't set up your PIN, enter your Temporary PIN and press # (see Setting Up Your Mailbox From Your Office Phone Number_for instructions on changing your temporary PIN). Your Temporary PIN is provided to you by your AT&T Voice DNA Administrator.

■NOTE:

If you fail to enter your PIN correctly in seven successive attempts, the phone access to your messages is disabled. If this happens, you need to contact your AT&T Voice DNA Administrator, so they can perform a Mailbox PIN reset. The TUI disconnects the call if you exceed four invalid PIN attempts in one TUI session, although your phone access to your messages may not be disabled at this point.

- The first time you access your mailbox, you will hear some helpful hints on how to use your mailbox. The next time you access your mailbox, the helpful hints will not be played.
- 4. You are now alerted how many new messages you have received, including the number of urgent messages and then taken to the Main Menu. This is the home for your mailbox whenever you are in your mailbox.

■NOTE:

If Autoplay is **ON**, the **Main Menu** is bypassed and your applicable messages will begin playing automatically. To go to the **Main Menu**, press 👻

- 5. If you have messages, you will hear, **To get your messages, press** ①. After pressing ①, you are prompted to press ① again to listen to your voice messages.
- 6. To leave your mailbox, hang up or press ቜ.

Get Messages	
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How to Get Your Voice Messages

The following steps describe how to get your voice messages.

- 1. Access your AT&T Voice DNA Mailbox.
- 2. Before you're taken to the **Main Menu**, you'll be alerted how many new messages you have received, including the number of urgent messages.

≡>NOTE:

If Autoplay is on, the **Main Menu** is bypassed and your messages will begin playing automatically. The Autoplay feature is set from the Main Menu by pressing 4, 2, and following the prompts. To go to the **Main Menu** at any time, press to back up until you get to the Main Menu. If you are in the Message Summary, will disconnect you.

- 3. If you have messages, you will hear, **To get your messages, press** ①. After pressing ①, you are prompted to press ① again to listen to your voice messages.
- 4. When you press ①, messages are played one at a time and include the date and time that the message was sent as well as the recorded name of other AT&T mailbox subscribers within your tenant (if available) and the phone number of the calling party (if available). The messages are played in the following order:
 - messages marked as urgent by callers.
 - new messages
 - saved messages.

- 5. While you are listening to your messages, you can:
 - Repeat or listen again press 4.
 - Save press ^⑨.
 - Erase press 7.
 - Reply (if another AT&T Voice DNA Mailbox user sent the message) press
 Pressing ® gives you additional options:
 - To place a call directly to this person press ①, to cancel the call, press ※
 - Reply to sender press 2.
 - Reply to all recipients press 3.
 - Forward a copy to another AT&T Voice DNA Mailbox user − press 6.
 - Play Header press 5.
 - Back up a few seconds press ①.
 - Slow Down press 4.
 - Pause and resume press 2.
 - Forward a few seconds press 3.
 - Speed Up press 6
 - Helpful Hints press ①.

If Autoplay is ${\bf ON}$ and you do not take any action, the message is saved and you will move on to the next message.

6. To leave your mailbox, hang up or press

★.

Quick Keypad Guide

Figure 134 and Table 18 show a quick guide when listening to your messages.

Guide to Listening to Your Messages

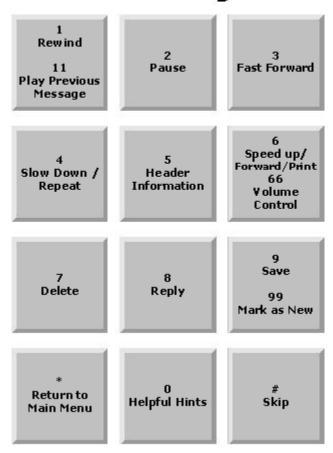


Figure 134. Telephone Keypad Shortcuts

Table 18. Telephone Keypad Shortcuts

Key	Action	
0	Helpful Hints	
1	Rewind Message	
2	Pause Message	
3	Fast Forward Message	
4	Repeat Message	
5	Header Information	
6	Forward a Few Seconds	
7	Delete Message	
8	Reply	
9	Save Message	
11	Play Previous Message	
66	Volume Control	
99	Mark as New Message	
*	Return to Main Menu	
#	Skip Message	

Send Voice Message

If you would like to leave a voice message for another AT&T Voice DNA Mailbox user within your tenant (or to a distribution list without calling them, you can compose a voice mail message and then send it to the user or the distribution list.

Compose New Voicemail Message

- 1. Access your AT&T Voice DNA Mailbox.
- 2. From the Main Menu, press ②. (If Autoplay is ON, press Ⅎ to access the Main Menu.)
- 3. Enter the 10-digit telephone number of the person or distribution list number to whom you want to send the voicemail message. You can send the voicemail to more than one person or to multiple distribution lists. If you get an error when entering the phone number, remember you must include the area code (see Distribution Lists for more info on distribution lists).
- **4.** You should hear the person's recorded name or telephone number if they are a valid user.
- 5. Press # to confirm you have finished addressing the message.
- 6. Record your message after the tone.
- 7. Press # to stop recording.
- 8. Press the #\ key to send the message or Press \(\frac{1}{2}\) to hear all of the **Delivery Options**.
 - Press ① to review.
 - Press 2 to mark message urgent.
 - Press 3 to mark message private.
 - Press 4 to re-record.
 - Press 5 to request a delivery report.
 - Press 6 to request a read report.
 - Press to request future delivery.
 - Press # to send message.
- **9.** If you are finished with your delivery options, pressing # sends the voice message.

Future Delivery of Voice Messages

You can schedule a voice message for a delivery up to 365 days in advance.

- 1. Complete steps 1-7 in the **Compose New Voicemail Message** section.
- 2. Press T to request future delivery. You will be prompted to enter the time and date of the message delivery.
- 3. Enter the number of the desired month for delivery (e.g. enter 3 for March).

■NOTE:

Enter 0 to denote the current month or day.

- **4.** Enter the number of the desired day for delivery. You will be prompted to confirm your entry.
- 5. Press 1 to confirm your entry or 2 to re-enter the day.
- 6. Enter the desired hour of day for delivery (e.g. enter 8 for eight o'clock).
- 7. Enter the minute of the hour for delivery (e.g. enter 30 for thirty).
- 8. Press 1 for AM or 2 for PM.
- 9. You will be prompted to confirm the time you entered. Press \square to confirm your entry or \square to re-enter the time.
- **10.** Press # to continue.
- 11. Press # to send your message.

Reviewing Your Future Delivery Queue

If there are future delivery messages scheduled, you will receive a future message delivery queue option in the Main Menu.

To Review the Message in Your Future Delivery Queue

- 1. Press 9.
- 2. The system responds with a message similar to the following:

"You have one message scheduled for delivery. First future scheduled for delivery, today at 8 PM. It will be sent to (phone number). To edit this message, press 1. To delete this message, press 2."

- To move to the next message, press #.
- To reschedule a message press ①.
- To re-record the message, press 2.
- To change message flags such as Urgency and Privacy, press 3.
- To save current changes, press 4.

Greetings

You can have several greeting options tied to the phone number associated with your mailbox. The greeting types are:

- Personal Greeting for everyday use. Your personal greeting (e.g., Hello, you have reached the office of Malcolm and Associates, we are on another call or are away from the phone, please leave a message.) is the main greeting that callers will hear when they call your office and you do not answer the telephone.
- Extended Absence Greeting gives callers a specific message and allows you the option of whether to allow voicemail messages to be left in your mailbox (Default is OFF no voicemails allowed). On special occasions you may want to use an Extended-Absence Greeting to give callers specific information. For example: I will be out of the office today, returning tomorrow. Please call back tomorrow.
- Pre-recorded Greeting can be used if you don't want to record your own personal greeting. You can use a pre-recorded greeting that:
 - contains your name along with a pre-recorded greeting. The name used is the name you recorded during the mailbox setup.
 - contains only your phone number along with a pre-recorded greeting.
 - does not contain your name or phone number along with a pre-recorded greeting.

For example, you could use the AT&T Voice DNA Mailbox Pre-recorded greeting, which says, <**Your recorded name**> can't take your call now. At the tone, please record your message. When you have finished recording, simply hang up or press the pound key for more options. Once the caller has recorded a message, they are provided prompts to either review the message, mark the message urgent or private.

You can change your greeting from any Touch-Tone® phone. You can record new greetings, switch among your Personal Greeting, the Extended-Absence Greeting and one of three Pre-recorded Greetings for the phone number associated with your AT&T Voice DNA Mailbox. You can also record a new mailbox name announcement. Your name announcement is what other subscribers will hear when they send you a message from their mailbox or when you send them a voice message and when the Pre-recorded Greeting is used.

Selecting a Greeting

- 1. Call into your mailbox using the phone whose greeting you want to change.
- 2. Press 4 at the Main Menu for Mailbox Settings. Then press 3 for Greetings.
 - Press ① to change the personal or pre-recorded greeting.
 - Press 2 to set the extended absence greeting to ON.
 - Press 3 to change the name.
 - If you have multiple phone lines, press 6 to record greetings for additional lines.
 - If you have multiple phone lines, press to use your primary phone line greeting on additional lines.

Mailbox Settings

When calling to access your AT&T Voice DNA Mailbox, you will have access to the following settings: **Distribution Lists**, **Mailbox Settings**, **Security Options**, **Notifications** and **Additional Settings**.

Distribution Lists

A distribution list is a group of telephone numbers of AT&T Voice DNA Mailbox subscribers to which you can send (or forward) messages.

Your distribution list can be used to reach several people with one message. For example, you can create List 1, which contains the phone numbers for people on your sales team. Then you can record one message (e.g., **Hi, everyone — today's sales meeting has been rescheduled to this Friday at 9 a.m.**) and send it to everyone on the list with a single call.

Create a Distribution List

- 1. From the Main Menu, press 4 for Mailbox Settings.
- 2. Press 2 for Administrative Options.
- 3. Press 2 for Group Distribution Lists.
- 4. Press 1 to add a new list.
- 5. Enter a **one-** or **two-digit** number for this list. This number will be what you use when addressing voice messages over the phone. You can have 15 distribution lists (25 maximum of AT&T Voice DNA subscribers per list). Press # when finished entering the list number.
- **6.** Record a **name** for the list, such as **Sales Team**. Press # when finished recording the list name.
- 7. Enter the 10-digit telephone numbers (or 1-2 digit Distribution List) of other AT&T Voice DNA subscribers you want to be on the list. Press # when finished entering the list item. You can have a maximum of 25 AT&T Voice DNA subscribers per distribution list (you can have up to 15 distribution lists).
- 8. When you have added all of the phone numbers for the list, press # to end the list.

Voice Mail Settings

Setting Your Administrative Options

- 1. Log into your mailbox using the phone number for which you want to change the Mailbox Settings.
- 2. Press 4 for Mailbox Settings.
- 3. Press 2 for Administrative Options.
- 4. Press 4 for Message Settings.
- 5. Press ① to configure your **Voicemail** settings. The system will state your current Voicemail settings and the option to change the setting. The following options are available:
 - Press 1 to turn ON/OFF hearing the Caller Name or Number while listening to Voicemails.
 - Press 2 to turn ON/OFF hearing the Date and Time of a Voicemail.
 - Press 3 to turn ON/OFF hearing the Current Body of a Voicemail.
 - Press ⁴ to turn **ON/OFF** hearing the Voicemail as soon as you log into you mailbox. When this **Autoplay** feature is set to **ON**, the **Main Menu** is skipped and your messages begin playing automatically. When **Autoplay** is active, you have the ability to manipulate the messages while they are playing. You can press ¹ to repeat, ² to save or ³ to delete while the message is playing. In addition, after each message you'll be given a few seconds of silence to decide what you want to do with the message. If you don't do anything, the AT&T Voice DNA mailbox will automatically save the message and move on to the next message. **This feature is useful when using a speakerphone or mobile phone to check your messages and you want to get through all your messages with minimal key presses.** After all of your new messages are played, you are directed to press ★ to enter the **Main Menu**.
 - Press 5 to turn ON/OFF hearing only the Current Urgent Messages played automatically.

Security Options

You have the following options under the Security Options:

- Change your telephone PIN.
- Set Fast Login to ON/OFF.
- Set Pin Skip to ON/OFF.
- Set Authentication Code for PIN resets.

Change Your Telephone PIN

- 1. Access your AT&T Voice DNA Mailbox.
- 2. From the Main Menu, press 4 for Mailbox Settings.
- 3. Press 2 for Administrative Options.
- 4. Press 1 for Security and Hands Free Options.
- 5. Press 1 to change your PIN.
- 6. Enter your new PIN. Your PIN should be 6 to 10 digits in length. If you make a mistake, press ★ to start again. When choosing your PIN, you should NOT:
 - Use your telephone or mailbox number (or any part of the telephone or mailbox number) as part of the PIN.
 - Repeat digits (e.g., 444444).
 - Use sequential digits (e.g., 345678).
 - Use easily identifiable numbers (e.g., ZIP code, street address, etc).

Treat your mailbox **PIN** as you would your ATM PIN. For added security, change your **PIN** periodically.

Fast Login, PIN Skip*

When you call into your mailbox from the telephone number associated with your mailbox, you will only need to enter your **PIN**. If you are calling from another phone, you will need to enter your phone number before you enter your **PIN**. These steps can be bypassed by turning **ON** the **Fast Login** (bypasses entering your phone number) and **PIN Skip*** (bypasses entering your **PIN**) features.

If both of these features are turned **ON**, when entering your mailbox, you will go directly to the **Main Menu** or directly to your messages if **Autoplay** is turned **ON**.

≡>NOTE:

Fast Login is initially set to ON at creation, but can be set to OFF. PIN Skip* can only be turned ON when Fast Login is turned ON.

A CAUTION:

* Use of the **PIN SKIP** feature reduces the security of your service by making your messages more vulnerable to unauthorized access by third parties. This includes unauthorized persons calling from the location associated with your mailbox or gaining access by using equipment to make it appear such calls are originating from the location associated with your mailbox. It is recommended that you always require access to your mailbox by using a secure **PIN**.

Set Fast Login and PIN Skip* to ON/OFF

- 1. Access your mailbox using the phone number for which you want to speed up the login process.
- 2. Press 4 at the Main Menu for Mailbox Settings.
- 3. Press 2 for Administrative Options.
- 4. Press 1 for Security and Hands Free Options.
- 5. Press 2 for **Fast Login** setup.
- 6. Press 1 to set Fast Login to ON/OFF (toggles).
- 7. Press 3 for PIN Skip*.
- 8. Press 1 to set PIN Skip* to ON/OFF (toggles).

■NOTE:

Fast Login is initially set to **ON** at creation, but can be changed to **OFF**. **PIN Skip*** can only be set to **ON** when **Fast Login** is set to **ON**.

A CAUTION:

* Use of the **PIN SKIP** feature reduces the security of your service by making your messages more vulnerable to unauthorized access by third parties. This includes unauthorized persons calling from the location associated with your mailbox or gaining access by using equipment to make it appear such calls are originating from the location associated with your mailbox. It is recommended that you always require access to your mailbox by using a secure **PIN**.

Authentication Code

- 1. Access your AT&T Voice DNA Mailbox.
- 2. From the Main Menu, press 4 for Mailbox Settings.
- 3. Press 2 for Administrative Options.
- 4. Press 1 for Security and Hands Free Options.
- 5. Press 4 for Authentication Code.
- **6.** Press ① to add or change. To use your:
 - Mother's date of birth press ①.
 - Father's date of birth press 2.
 - Spouse's date of birth press 3.
 - Child's date of birth press 4.
 - Play options again press 5.

After selecting an option, you are prompted to enter the chosen date as 8 digits. For example, **July 10, 1950** should be entered as **07101950**. You are prompted to re-enter the chosen date again to ensure a mistake was not made. If both 8 digit selections were entered the same, your authentication code will be saved.

7. Press 2 to skip (this option disappears if you skip once) or delete. If the Authentication Code is deleted, you will need to contact a customer care representative to reset your PIN. You can skip the initialization once. After skipping once, the next time you will be forced to make a selection (e.g., setup or delete).

Notifications Settings

The AT&T Voice DNA Mailbox has several ways of keeping you informed of new messages that have been left in your mailbox. This means you are able to respond to individuals leaving messages in a timely manner.

Once these are set up, you then have the option to turn the notification features **ON** or **OFF** over the telephone.

(phone) Notification

The AT&T Voice DNA mailbox notifies you with a Message Waiting Indicator (MWI) on the telephone number associated with your service. This indicator can either be a light on specially equipped phones or an **interrupted** dial tone for analog phones behind an adapter.

Setting up Your Notification Options

Your notification options can be set to **ON** or **OFF**.

Set Notification to ON/OFF

- 1. Access your AT&T Voice DNA Mailbox
- 2. Press 4 from the Main Menu.
- 3. Press I for the **Notifications** menu.
- **4.** Press ① for **Notification** on your land line. This will change your (MWI) notification setting (only for your land line). Once you've pressed ①, you can toggle the notification **ON** or **OFF** by pressing ① again.
- 5. Press ② for **Pager Notification**. This will change your pager notification setting. Once you've pressed ②, you can toggle the notification **ON** or **OFF** by pressing ② again.

Additional Settings

Attendant Call Coverage

Attendant Call Coverage is a feature that allows your callers to press ① while your greeting is playing, and to be transferred to a different phone number within your tenant. This allows you to set up a ten digit telephone number to which a caller is **transferred** if they decide to press ② during your greeting.

Setup Attendant Call Coverage Number

To set up your ten digit **transfer** number, when calling in to your mailbox from a telephone:

- Press 4 to enter Mailbox Settings.
- Press 2 for Administrative Options.
- Press 6 for Additional Settings.
- Press 2 to enter the **Attendant Number** and follow the instructions.

For example, you can have the caller transferred to an administrative assistant as part of a company greeting or to a telephone number where you may be at for a period of time (as explained in your own personal greeting). If you have the attendant number set up, you should include the option for the caller to press ① as part of your greeting. An example of a personal greeting might be, Hi this is Larry. I am either away from my desk or on the phone. If you need to reach someone immediately, press ① to be connected to my assistant. If the caller presses ① they will hear, Please hold while I transfer your call. If there is no answer, the caller is returned to your greeting and will be able to leave a voice message in your mailbox.

Delete Oueue

Your deleted messages are placed in the queue. You can recover deleted messages in this queue for approximately 48 hours after they were deleted. To recover a deleted message, press **6** (Delete Queue) from the **Main Menu** and follow the prompts.

How Messages Are Played When You Call Into Your Mailbox

See the **Security Options** and **Voice Mail Settings** sections.

Reminders Messages

The **Reminders** feature allows you to record and schedule a message to be delivered to your telephone number. Reminders can only be set from the phone number associated with your mailbox.

To work with your reminder messages from the **Main Menu** press 5.

Messages can be scheduled for a one-time delivery, weekday (Monday-Friday) delivery, and everyday (Sunday through Saturday) delivery. Since this feature actually rings your telephone line, some user may use it as a way to remind themselves of an upcoming event. If you do not answer the phone (the system tries to deliver the reminder 3 times), the reminder will be deposited in your mailbox.

If the reminder is unanswered, on your Personal Web Site, the TN field only displays "-", (no numbers) and the Name Field displays "Unavailable". The Call Log for Received records associated with this feature indicate the user's own TN and Name.

Call Back Now

The Call Back Now feature allows you to call the sender of a message you have received in your mailbox. Press $\$ 1 to reply to the message, and if the caller's telephone number is available, press $\$ 1 to call the sender, the system then dials that number. You will be placed on hold until a connection is made. After you complete that call, you will be returned to your mailbox to continue reviewing messages. Some callers, for privacy reasons, do not want their telephone numbers known to others and/or the system, so those callers' numbers are blocked and Call Back Now option will not be an option for them.

Navigation Map

Figure 135 shows a quick diagram of options available from the **Main Menu**.

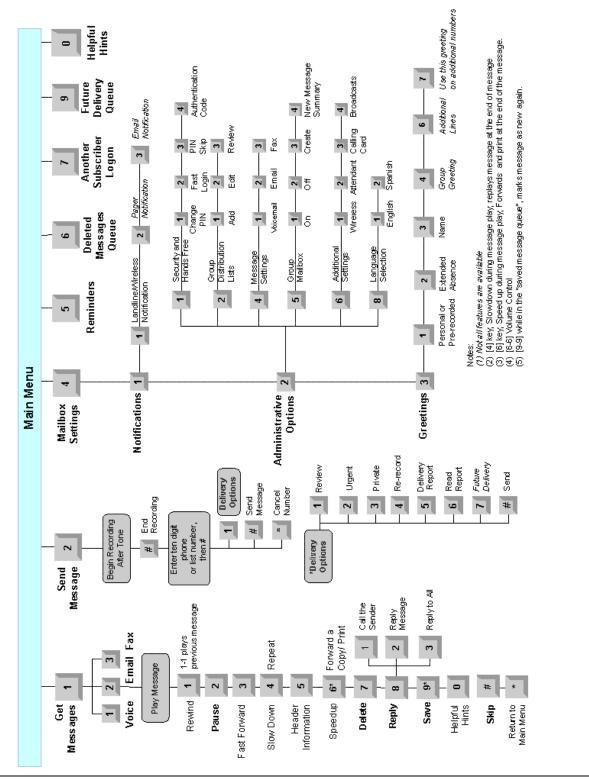


Figure 135. Navigation Map

Frequently Asked Questions (Mailbox)

Q: Can I access my mailbox from any phone, or do I have to use my own phone?

A: You can access your AT&T Voice DNA Mailbox from any Touch-Tone® phone, no matter where you are, at any time of day. You can use your wireless phone, your office phone or any pay phone to access the system. Depending on where you are calling from, local, local toll or long distance usage charges may apply.

Q: One of my callers could not leave me a message because my mailbox was full. How can I prevent that?

A: After listening to a message, we suggest deleting/erasing it in order to maximize capacity of storage. When your mailbox is full, callers are not able to leave you a message until you delete messages from your mailbox. Your mailbox capacity is set at 100MB. Each one-minute message left in your mailbox equates to approximately 1 MB of storage capacity.

Q: Do I have to listen to all the options before I make my selection from a menu?

A: No. You can press a key at any time. You don't have to wait for the system to list each menu option. The **Navigation Map** will help you become familiar with your menu options.

Q: Some of my callers want to skip listening to my greeting and record their messages immediately. How can they do this?

A: Tell your callers to press any key when they hear your greeting. The greeting will stop playing, and they will hear the record tone.

Q. If I delete a voice message over the telephone, will I be able to access it again?

A: If you erase a message from the telephone, the message is available for up to 48 hours, at which time it is permanently deleted from the Delete Queue.

Q: How long can I make my greeting?

A: Up to three minutes.

Q: Why is the Message Waiting Indicator light on my phone lit even though I do not have any new messages?

A: Your phone's Message Waiting Indicator light (or **interrupted** dial tone) may remain active if you have accessed and listened to new messages from the telephone and have not yet hung up your phone. As long as you have taken action to either delete or save the messages, and no new messages have arrived, the light should turn off. The light will turn off within seconds of you logging out of your mailbox or hanging up the phone as long as no new messages have arrived in the interim that would re-activate the light.

AT&T Voice DNA Mailbox Definition of Terms (Glossary)

Access Number – The number you were given to access your messages over the telephone.

Distribution List – A list of AT&T Voice DNA Mailbox, Call-in-One and/or Mailbox subscribers to which you can send or forward messages within your Enterprise. You can use a distribution list to send the same message to a group of people within your Enterprise with one phone call from your AT&T Voice DNA Mailbox.

Fast Login – The feature that enables your system to recognize if you are calling from the phone associated with your AT&T Voice DNA Mailbox. When this feature is turned on, you do not have to press # when logging in to your mailbox. When on, the system recognizes your phone number (mailbox number). This feature can also be used in combination with the PIN Skip* feature so that neither a phone number nor PIN is required for you to log in to your mailbox over the phone.

Interrupted Dial Tone – A special-sounding dial tone that plays intermittently. It is one of the forms of MWI (Message Waiting Indicator). This **interrupted** dial tone alerts you to new messages when you pick up your phone. This is used for analog phones behind an adapter.

Mailbox Number – This is the phone number associated with your AT&T Voice DNA Mailbox.

MWI - Message Waiting Indicator

PIN Skip¹ – The feature that works with Fast Login and allows you to access your mailbox without entering your PIN if you are calling from the phone number associated with your AT&T Voice DNA Mailbox.

Subscriber – This term refers to an AT&T Voice DNA Mailbox user.

Use of this feature reduces the security of your service by making your messages more vulnerable to unauthorized access by third parties. This includes unauthorized persons calling from the location associated with your mailbox or gaining access by using equipment to make it appear such calls are originating from the location associated with your mailbox. It is recommended that you always require access to your mailbox by using a secure PIN.

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Supported AT&T Voice DNA Service Features on the AT&T Messaging Platform

Table 19. Supported AT&T Voice DNA Service Features on the AT&T Messaging Platform

Feature	TUI Prompt	Parameter
GREETINGS		
Personal Greeting	4,3,1	180 seconds
System Greeting	4,3,1	
Extended Absence Greeting	4,3,2	
Turn Off Messages (with Extended Absence Greeting)	4,3,2,	Default is turned off
Turn On Messages (with Extended Absence Greeting)	4,3,2,1	
Name Announcement	4,3,3	
MESSAGE SIZE and MAILBOX CAPACITY		
Message Length		3 minutes
Storage Capacity		100 MB
Mailbox Full Warning – 90% and 100%		
MAILBOX ACCESS		
Multiple access methods (from user's primary phone or any other phone)		
Default PIN/Password		Contact your AT&T Voice DNA Administrator.
		PIN length: 6-10 digits
Multiple Login tries		Max. 7 successive attempts; phone access is disabled.
		If disabled, contact your AT&T Voice DNA Administrator.
		Call disconnects on four invalid PIN attempts in one session.
Fast Login - allows subscribers to login without inputting their 10 digit telephone number before entering a PIN	Set up – Prompts 4,2,1,2	

Table 19. Supported AT&T Voice DNA Service Features on the AT&T Messaging Platform

Feature	TUI Prompt	Parameter
PIN Skip when accessing from subscriber's phone	Set up – Prompts 4,2,1,3	
Auto Play - begins playing new messages in the mailbox without any keys being pressed or going through any menus. Note that AutoPlay will not prompt the user to save/delete message after the message is played. It automatically saves the message and then plays the next new message		Only if PIN Skip is turned ON.
Access to Mailbox from Greeting (barge in)	Press * key when greeting is being played	
Exit Voicemail		
RETRIEVED MESSAGE HANDLING AND RETENTION		
Play Message	1	Via TUI and Personal Web Site
Back up while listening to message	1,1	Back up 3 seconds
Forward while listening to message	1,6	Forward 3 seconds
Message Pause via TUI	1,2	20 seconds
Save Message	1,9	Saved messages played in LIFO order
Repeat Message	1,4	
Play Previous Message	1,1,1	
Erase/Delete Message	1,7	Available for retrieval within 2 days
Retrieve Deleted messages	6	Available for retrieval within 2 days
Restore Deleted messages	6,2	
Reminder messages	5	Up to 5 one-time reminders and 5 recurring reminders per subscriber; Reminder Message
		length – 60 seconds
Reply - Two-Way Messaging between AT&T Messaging subscribers (Mailbox to Mailbox)	1,8,2	

Table 19. Supported AT&T Voice DNA Service Features on the AT&T Messaging Platform

Feature	TUI Prompt	Parameter
Live Reply – Reply to messages left via call answering	1,8,1	If Privacy indicator of sender set to N
Message Retention Period		Unlimited
SEND VOICE HANDLING & FEATURES		
Message Review prior to Send	2,1,1,#	
Name verification of called party - mailbox to mailbox		Between 2 UM subscribers only
Urgent Message marking	2,1,2,#	
Private Message marking	2,1,3,#	
Confirm Message delivery	2,1,5,#	
Confirm Read Message	2,1,6,#	
Private Delivery (callers can mark a message private)	2,1,3,#	
Caller Announcement (Called phone number/Caller ID info)	4,2,4	
Marketing Broadcast toggle (On/Off)	4,2,6,4	
Distribution Lists (created via TUI)	4, 2,2	Max. 15 lists per subscriber;
		Max. 25 members per list
MAILBOX MANAGEMENT		
PIN lockout	Call Voice DNA Administrator	Via Voice DNA Administrator Tool
Password change	4,2,1,1	Via TUI and Personal Web Site
Password reset by Administrator		Via Voice DNA Administrator Tool
Password reset by User (via Authentication)	4,2,1,4	Via TUI
E-mail alert upon receipt of message		Via Personal Web Site
Operator Revert/Zero out	4,2, 6,2	User-defined via TUI
Time and Date stamp toggle	4,2,4	Via TUI
Language toggle	4,2,8	English and Spanish

Transfer a Call Directly into Voicemail

Overview

The below listed instructions provide steps for transferring a caller directly to a Voice DNA user's mailbox by invoking feature code *90. This feature allows any employee to transfer a caller directly to another employee's mailbox to avoid the delays associated with the caller having to first wait for the employee's phone and any subsequent **Locate Me** numbers to ring before their voicemail picks up.

Instructions

The following instructions are for the scenario where:

- Caller A calls into company
- Caller A is answered by Employee 1 but Caller A wants to leave a voicemail message with Employee 2.

Transfer via Polycom IP601 Model Phone

Employee 1 Actions:

- 1. Select < Transfer > feature button.
- 2. Press the **Blind** soft key button.
- 3. Enter *901234 (where 1234 is Employee 2's extension).
- 4. Press the **Send** soft key button (return the handset to its cradle).

Employee 1 is now dropped from the call and Caller A hears Employee 2's Voicemail greeting.

Transfer via Polycom IP301 Model Phone

Employee 1 Actions:

- 1. Select the More soft key button.
- 2. Press the **Transfer** soft key button.
- 3. Press the **Blind** soft key button.
- 4. Enter *901234 (where 1234 is Employee 2's extension).
- 5. Press the **Send** soft key button (return the handset to cradle).

At this point Employee 1 is dropped from the call and Caller A hears Employee 2's VM greeting.

Transfer via Cisco 7960 Model Phone

Employee 1 Actions:

- 1. Select the **More** soft key button.
- 2. Press the **Bindxfer** soft key button.
- 3. Enter *901234 (where 1234 is Employee 2's extension).
- 4. Press the **Dial** soft key button (return the handset to cradle).

At this point Employee 1 is dropped from the call and Caller A hears Employee 2's VM greeting.

Transfer via LG 6812/6830 Model Phone

Employee 1 Actions:

- 1. Press the Blndxfer soft key button.
- 2. Enter *901234 (where 1234 is Employee 2's extension).
- 3. Press the **Call** soft key button (return the handset to cradle).

At this point Employee 1 is dropped from the call and Caller A hears Employee 2's VM greeting.

Transfer via ATA Supported Device

Employee 1 Actions:

- 1. Do a flashhook on the phone connected to the ATA.
- 2. Enter *901234# (where 1234 is Employee 2's extension).
- 3. Hang up.

At this point Employee 1 is dropped from the call and Caller A hears Employee 2's VM greeting.

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