Vocera Voice Server Telephony Configuration Guide

Version 5.4.0

Notice

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Contents

Getting Started

This chapter describes how to get started using the Vocera SIP Telephony Gateway.

About Vocera Telephony

Vocera offers **Vocera SIP Telephony Gateway** which is a software telephony solution that provides a Session Initiation Protocol (SIP) telephony gateway between the Vocera Voice Server and an IP PBX or a Voice over Internet Protocol (VoIP) gateway. Vocera SIP Telephony Gateway supports non-SIP enabled PBXs via Dialogic Media Gateway or other SIP/TDM gateway products.

The Vocera SIP Telephony Gateway provide the following key benefits:

- Call to and from PBX extensions, voicemail, and the public telephone network
- Outgoing digital paging with direct callback to the Vocera badge
- Direct inward dialing (DID)
- Speech-to-touch-tone dialing
- Supports installation of multiple telephony servers for N + 1 redundancy, scalability, and load balancing
- Support for deployment in a VMware virtualized environment
- Vocera Access Anywhere (phone access to the Vocera Genie) for all users

Session Initiation Protocol Support

Vocera SIP Telephony Gateway is based on Internet Engineering Task Force (IETF) standards for Session Initiation Protocol (SIP) 2.0 and Real Time Transport Protocol (RTP).

Vocera SIP Telephony Gateway communicates via a SIP trunk with a SIP-enabled PBX or a SIP Gateway and provides basic SIP telephony functionality, including placing and receiving calls, OPTIONS keep-alive messages, and obtaining ANI and DNIS information. The Vocera SIP Telephony Gateway is interoperable with SIP-enabled PBXs and SIP Gateways as long as they follow SIP 2.0 and RTP standards.

For audio transport, Vocera SIP Telephony Gateway uses Real-time Transport Protocol (RTP), an Internet protocol standard for delivering multimedia data over unicast or multicast network services. For more information refer to RFC 3550 at http://tools.ietf.org/html/rfc3550 and RFC 35515 at http://tools.ietf.org/html/rfc3551.

Vocera Voice Server uses Vocera proprietary signaling and transport protocols for all communication between the server and Vocera badges. Consequently, Vocera SIP Telephony Gateway converts from SIP and RTP protocols to Vocera protocols, and vice versa, to enable communication between the Vocera Voice Server and the IP PBX.

Vocera SIP Telephony Gateway SIP Trunk Specifications

Understand requirements for using a hosted or cloud-based telephone system with Vocera Voice Server.

Requirements for Connecting the Vocera SIP Telephony Gateway to a Cloud-based or Hosted Telephone System

The Vocera SIP Telephony Gateway (VSTG) can connect to a SIP trunk on a hosted or cloud-based PBX either directly or through a SIP gateway configured at the network edge. To successfully connect and interoperate the SIP trunk used by the VSTG, your telephone system must support the following requirements:

- SIP version 2.0 as specified in RFC 3261.
- RFC 2833 standard DTMF in order to support features such as paging and dialing Vocera extensions through the guest access number.
- Real Time Protocol (RTP) audio to the VSTG must use the G.711 μ-Law or A-Law codec. The packet spacing must be 20 milliseconds.

If you experience interoperability challenges with a SIP trunk supporting these standards, please contact Vocera Technical Support for further assistance

What is Not Supported

The VSTG does not support these features:

- SIP Trunk Authentication
- SIP Trunk Registration
- NAT Traversal

Using the Administration Console

The Vocera Administration Console is a browser-based application that allows you to configure the Vocera Voice Server and the Vocera SIP Telephony Gateway.

This section describes how to get started using the Administration Console.

Browser Requirements

To access Vocera Voice Web applications (Administration Console, User Console, Report Console, and Staff Assignment), your computer must have the following required software:

Table 1: Web application software requirements

Applications	Client-side component	Requirement
All applications	Browser	Internet Explorer versions 10 or later (Be sure that Compatibility Mode is turned off).

Important: Do not install another JRE on the Vocera Voice Server or Vocera Report Server machines. The required version of Java is installed with those servers.

Browser Recommendations and Requirements

Vocera recommends Internet Explorer and suggests configuration settings required for internet and browser security.

The list below includes the configuration recommendations for Internet Explorer security settings. Required items are flagged.

- **Do not access a Vocera Voice Web application from the server on which it is running** – By default, Windows Server 2008 and Windows Server 2012 ship with Internet Explorer Enhanced Security Configuration enabled, which may display frequent security prompts when you access a Web application from the server on which it is running. Rather than disable Internet Explorer Enhanced Security Configuration on the server, we recommend that you access Vocera Voice Server Web applications from your desktop or laptop computer.
- **Configure the Internet Explorer security level to Medium-low or lower** – Otherwise, Internet Explorer prevents the scripts used by the consoles from executing completely. You can configure security settings through **Tools > Internet Options > Security** in Internet Explorer. See your Internet Explorer documentation for complete information.
- **Required: Disable the pop-up blocker** – Vocera consoles display information in pop-up windows, so disable pop-up blocking in Internet Explorer (that is, configure the browser to allow pop-up windows). Choose **Tools > Internet Options > Privacy**, and then uncheck the **Turn On Pop-Up Blocker** box.

  If you are using a third-party tool to block pop-ups, refer to the tool's documentation.
- **Remove scroll bars from pop-up windows** – Pop-up windows may display scroll bars. To remove the scroll bars, choose **Tools > Internet Options > Security**, and select the **Local Intranet** zone. Click **Custom Level** to display the Security Settings dialog box. Enable **Allow script-initiated windows without size or position constraints**.
- **Required: If necessary, add the Vocera Voice Server and Vocera Report Server IP addresses to the list of Trusted Sites** – The security policy in certain situations may prevent you from setting the Internet Explorer security level for the local intranet below Medium. If Internet Explorer continues to display pop-up windows with scroll bars, follow these steps to configure a trusted site for the Vocera Voice Server:

  To add the Vocera Voice Server and Vocera Report Server to the list of trusted sites:

  1. In Internet Explorer, choose **Tools > Internet Options**. The Internet Options dialog box appears.
  2. Click the **Security** tab.
  3. Click **Trusted Sites**.
  4. In the **Security Level for this Zone** box, set the security level to Medium-low, and click **Apply**.
  5. Click the **Sites** button. The Trusted Sites dialog box appears.
  6. Type the IP address of the Vocera Voice Server, and click **Add**.
  7. Type the IP address of the Vocera Report Server, and click **Add**.
  8. Click **Close** to close the Trusted Sites dialog box.
  9. Click **OK** to close the Internet Options dialog box.

  A system administrator can manage the Internet Explorer Trusted Sites for an entire organization using Group Policy Objects (GPOs).
- **If your Vocera Voice Server or Vocera Report Server has enabled SSL, configure Internet Explorer to NOT save encrypted pages to disk** – If you enable SSL on theVocera Voice Server or Vocera Report Server, you may need to update the browser security settings for Internet Explorer to make sure the browser does delete cached-from-HTTPS resources when the browser is closed. Otherwise, certain pages of the Administration Console, such as the Permission Browser, will not work properly.

  To update Internet Explorer security settings for SSL access:

  1. In Internet Explorer, choose **Tools > Internet Options > Advanced**
  2. Make sure the **Do not save encrypted pages to disk** option is checked.
  3. Click **OK**.

Logging In

Learn about accessing the Administration Console.

The Administration Console lets you manage a Vocera system. It is a browser-based application, accessible from any computer on the network.

The console URL is either of the following, where vocera_ip_address is the numeric IP address of the Vocera Voice Server:

Table 2: Administration Console URLs

Type of Access	URL
Unencrypted	`http://` vocera_ip_address `/console/adminindex.jsp`
SSL	`https://` vocera_ip_address `/console/adminindex.jsp`

After you complete the initial setup and your organization starts using Vocera, access the Administration Console from a computer that is not running the Vocera Voice Server so you don't degrade system performance.

Logging In Using Active Directory Authentication

When Active Directory authentication is enabled, the Administration Console welcome page has an additional field, the **Active Directory** list, which specifies the Active Directory to use for your login.

To log into the Administration Console using Active Directory credentials:

1. Open an Internet Explorer browser window.
2. Enter the Administration Console URL to open the Administration Console welcome page.
3. Specify the following values:

Field	Description
User ID	Enter your Active Directory user ID (up to 250 characters). You must be a member of a Vocera group that has administrator privileges.
Password	Enter your Active Directory password (up to 127 characters).
Active Directory	Select the name of your Active Directory from the list. If there are multiple Active Directories listed and you're unsure which one to select, ask your system administrator.

4. Click **Log In**.

   The Administration Console opens.

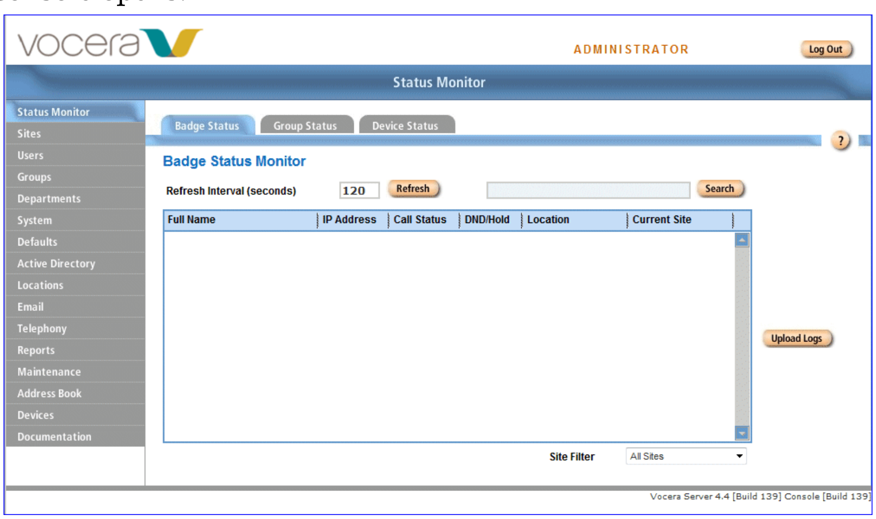


Figure 1: Administration Console opening screen

Logging In Using Vocera Authentication

If Active Directory authentication is not enabled, the **Active Directory** list does not appear on the Administration Console welcome page and you must log into the Administration Console using your Vocera credentials.

To log into the Administration Console using Vocera credentials:

1. Open an Internet Explorer browser window.
2. Enter the Administration Console URL to open the Administration Console welcome page.
3. Specify the following values:

Field	Description
User ID	Enter your Vocera user ID. You must be a member of a Vocera group that has administrator privileges.
Password	Enter your Vocera password.

4. Click **Log In**.

   The Administration Console opens.

Logging In Using the Default Administrator Account

Vocera provides a built-in administrator account with the user ID **Administrator**.

The default Administrator password is **admin**, but you can change it to something more secure.

Note: Regardless if Active Directory authentication is enabled, the default Administrator account does not use Active Directory credentials to log in.

Administration Console User Interface

Learn the basic User Interface controls on the Administration Console.

The following figure uses the Sites screen to show some of the user interface controls available in a typical Administration Console screen.

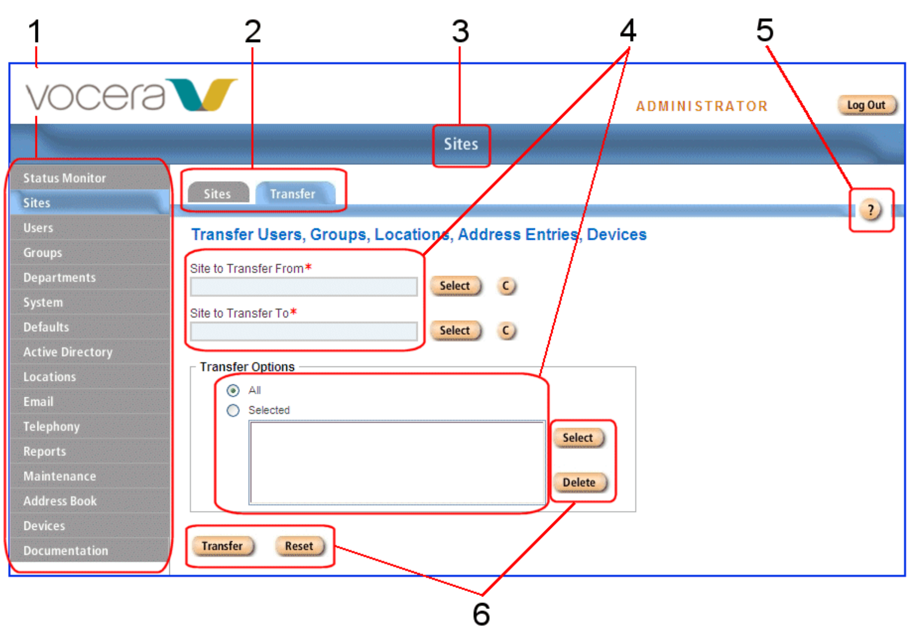


Figure 2: Administration Console user interface

1. **Navigation bar.** Click a button to display a screen.
2. **Tabs.** Click a tab to display a page in a screen.
3. **Screen title.** Displays the screen name.
4. **Fields.** Click a field to select or edit a value in a page.
5. **Help button.** Click the **?** button to display context-sensitive help.
6. **Buttons**. Click a button in a page to perform an action.

Some pages have buttons that open dialog boxes. For example, when you click the **Add New User** button on the Users page, it opens the Add New User dialog box:

Figure 3: Add New User dialog box

Dialog boxes have tabs that group complex information to make it easier to enter and understand. Most have both a **Save** and **Save & Continue** button.

- Click **Save** to save the data and close the dialog.
- Click **Save & Continue** to save the data and clear the fields. It leaves the dialog open to allow faster entry of new information.
- Click **Cancel**, or close the dialog without clicking **Save** or **Save & Continue**, to discard changes made.

Displaying Vocera Documentation

You can display Vocera documentation online through the Administration Console.

Vocera documentation includes manuals in PDF (Portal Document Format) files and context-sensitive help.

Displaying Documentation Online

The Adobe Reader, a free software program from Adobe Systems Incorporated, is required to display the Vocera documents installed as PDF files.

You can download the Adobe Reader or learn more about it on the Adobe Reader home page.

Use the following steps to view the Vocera documentation:

1. Click **Documentation** on the navigation bar.

The Documentation page displays links to PDF versions of Vocera documents.

2. Click a link to view a document.

   The manual or instruction sheet opens in a separate window.

Displaying Help

The Administration Console provides context-sensitive help.

To display help, click the **?** button on any console page.

Using the Vocera Control Panel

This section describes how to control Vocera SIP Telephony Gateway using the Vocera Control Panel.

Vocera System Tray Icon

Learn about the use and function of the Vocera system tray.

When the Vocera Voice Server, Vocera SIP Telephony Gateway, or Vocera Client Gateway server starts running, the Vocera system tray icon appears in the server notification area at the right of the taskbar.

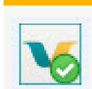
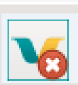
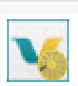
Figure 4: System tray

You can use the Vocera system tray icon to start the Vocera Control Panel for your user session. The Vocera Control Panel displays status messages and lets you control the server.

Note: Windows 2008 R2 systems may require additional configuration to add the Vocera Control Panel system tray options to the notification area.

The Vocera system tray icon changes to display the status of the server:

Table 3: Vocera system tray icons

Icon	Description
	The server is running. You can use the Vocera system tray icon to start the Vocera Control Panel.
	The server is not running. You can use the Vocera system tray icon to start the server.
	The server is processing a stop or start request.

To display the Vocera Control Panel:

- Right-click the Vocera system tray icon , and select the following command appropriate for your server:
  - Vocera Voice Server = **Vocera Control Panel**
  - Vocera SIP Telephony Gateway = **VSTG Control Panel**
  - Vocera Client Gateway = **VCG Control Panel**

  The Control Panel window appears on the desktop.

Note: On the Vocera Voice Server, a Command Prompt window called **Vocera Launcher Console** also appears. It displays status messages as Vocera processes are started and stopped.

To start up the server:

- Right-click the Vocera system tray icon, and select the following command appropriate for your server:
  - Vocera Voice Server = **Start Vocera**
  - Vocera SIP Telephony Gateway = **Start VSTG**
  - Vocera Client Gateway = **Start VCG**

  The Control Panel window appears on the desktop.

Vocera Control Panel Menus

Learn about the Vocera Control Panel commands and functionality.

Table 4: Control Panel menus

Menu	Command	Description	Servers
Run	**Start**	Starts the server.	VS, VSTG, VCG
	Stop	Temporarily suspends the server.	
	Shutdown	Shuts down the server.	
Display	**Normal**	Displays only the most significant system events. This is the default.	VS only
	Detailed	Displays all events.	
	Off	Displays no events.	
Cluster	**Start Standalone**	Temporarily removes a Vocera Voice Server from a cluster and restarts it as a standalone system.	VS only
	Failover	Fails over to the standby Vocera Voice Server, or restarts the server if it's currently in standby.	
Server	**IP Address(es)**	Specifies the Vocera Voice Server IP address(es) used by the server.	VSTG, VCG
Help	**Contents**	Displays online help.	VS, VSTG, VCG
	About	Displays version information.	

Determining the Status of the Server

The Vocera Control Panel provides a status indicator below the menu bar at the top of the screen.

The indicator displays one of the following states to tell you whether the server is available for use:

Table 5: Control Panel status

Status	Description
Active	The server is running and available for use. A standalone Vocera Voice Server is always active unless you have stopped it. A Vocera Voice Server that is part of a cluster is active when it is the primary machine, unless you have stopped it.
Standby	The server is running but is not available for use. A Vocera Voice Server that is part of a cluster is in the standby state when it is one of the secondary machines.

Stopping and Restarting the Server

Learn how to start and stop the sever, the conditions under which should, and the affect to the clients.

In certain situations, you may need to stop and restart the server. For example, if you want to update the properties in all your badges at the same time, you must stop the Vocera Voice Server and then restart it.

You may want to restart the server when only a few people are using the system. When the server is stopped, clients are unable to connect and communication is temporarily suspended:

- When the Vocera Voice Server is stopped, users cannot communicate with their badges.
- When the Vocera SIP Telephony Gateway is stopped, users cannot place or receive phone calls.
- When the Vocera Client Gateway is stopped, users cannot communicate with the Vocera Collaboration Suite app or with Vocera Smartphones.

The server stops and starts fairly quickly, so if few people are using the system, there will be very little interruption.

Note: You can also use the Server page of the **Maintenance** screen in the Administration Console to stop and start the server.

To stop and restart the server:

1. In the Vocera Control Panel, choose **Run > Stop** or click .
   The Control Panel displays messages indicating that the server has stopped.
2. Choose **Run > Start** or click .
   The Control Panel displays messages indicating that the server has started.

Shutting Down the Vocera Voice Server

Follow these steps to stop the Vocera Voice Server.

When you shut down the server, you stop the Vocera Voice Server and all its related services. In the case of the Vocera Voice Server, this includes MySQL, Tomcat, Apache Web Server, Nuance, and ASR Broker Service if you have the license with enhanced voice entitlement.

To shut down the server:

1. In the Vocera Control Panel, choose **Run > Shutdown**
   A confirmation dialog box appears.
2. Click **OK**.
   The dialog box closes, and the Control Panel also closes.
   If you shut down the Vocera Voice Server, the launcher Command Prompt window displays messages indicating that Vocera and its related services are stopping. When all Vocera services have stopped, the Command Prompt window closes.

Changing the Vocera Voice Server IP Address

Follow these steps to change the Vocera Voice Server IP address.

The Vocera SIP Telephony Gateway, and Vocera Client Gateway need to know the IP address(es) of the Vocera Voice Server. You enter this IP address(es) when you install the software. However, you can use the Vocera Control panel to change the address.

To change the Vocera Voice Server IP address used by the server:

1. In the Vocera Control Panel, choose **Server > IP Address(es)**
   The IP Address dialog box appears.

2. Use the **Server IP Address** field to provide the address of the Vocera Voice Server.

   Enter the numeric IP address using dot notation. For example:

   192.168.15.10

   For a Vocera Voice Server cluster, enter a comma-separated list of IP addresses. For example:

   192.168.15.10,192.168.15.11,192.168.15.12
3. Click **OK**.

   The dialog box closes, and the server begins using the new Vocera Voice Server IP address immediately.

Vocera Control Panel Title Bar

Learn about the function of the Vocera Control Panel title bar.

The following figure shows the Vocera Control Panel title bar and identifies what each field represents.

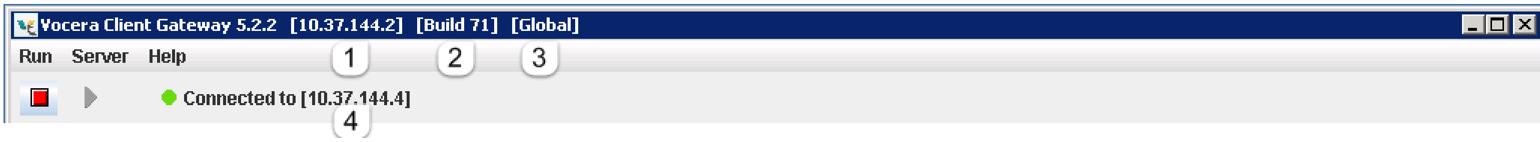


Figure 5: Control panel title bar

1. **Client Gateway IP Address**: Displays the IP Address for the Client Gateway.
2. **Build**: Displays the build number for the Vocera Voice Server installation.
3. **Vocera Site**: Displays the Vocera site for Vocera SIP Telephony Gateway (VSTG) and Vocera Client Gateway (VCG).
4. **Vocera Voice Server IP Address**: Displays the IP Address for the Vocera Voice Server.

Working with Phone Numbers

When a user issues a voice command to dial a telephone number, or when Vocera forwards a badge call to a telephone or to voicemail, the Vocera Voice Server sends a sequence of digits to the Telephony server. In addition to the phone number itself, the sequence can contain access codes needed to obtain an outside line, to authorize a long distance call, or to access company voicemail.

You can enter phone numbers, extensions, and access codes in various places in the Administration Console. For example, when you add a user to the Vocera system, you can specify the user's desk extension, cell phone number, pager number, and home phone number. Users can also enter or update this information in the User Console.

A field that requires a phone number, an extension, or an access code may contain any of the following characters:

- Digits. 1234567890
- Special dialing characters. A special dialing character is a non-numeric character that you can enter in an Administration Console field that requires an access code, phone number, or extension. For example, you can use an asterisk (*) to simulate pressing the star key on a touch-tone phone, or enter an X at the beginning of a number to tell Vocera to treat that number as an extension.
- Special dialing macros. A dialing macro represents a dialing sequence. In data entry fields where you cannot enter a specific number—because the number varies with the user who accesses the feature—you can enter a dialing macro. Vocera replaces that dialing macro with the actual number on demand.

Dialing macros are especially useful when editing Company Voicemail Access Codes and Address book entries. For example, the Company Voicemail Access Code field specifies the dialing sequence that Vocera uses to forward an incoming call to company voicemail. As part of the dialing sequence, you typically need to specify a desk phone extension to identify the voicemail box you want to access. You cannot enter a specific desk extension in this field, because the number will vary depending on which user is forwarding calls. Instead, you use the **%D** macro as part of the dialing sequence. Vocera replaces that macro with the actual desk extension of the user who is forwarding calls. See Special Dialing Macros on page 77 for a complete list of dialing macros.

- PIN template macros. Each PBX has different rules for adding a PIN to a dialing sequence. Some require the phone number followed by the PIN. Some require the PIN before the phone number. Some require an access code for an outside line, or a feature code to indicate that a number is a PIN. Some require a separator character between the PIN and the number. A telephony PIN template can use macros to specify and format the information in a PIN. See PIN Template Macros on page 78 for a complete list of PIN macros.

Vocera ignores any other character that you enter in these fields. For example, you can enter `(408) 790-4100`, to make a number more readable, instead of `4087904100`. Vocera ignores the extra spaces, dashes, and parentheses when the number is actually dialed.

Telephony Email Alerts

You can configure your system to send email alerts to notify you when there are problems with the Vocera Voice Server and the telephony server. The Vocera Voice Server sends a telephony email alert when one of the following events occurs on the telephony server:

- The Vocera Voice Server is unable to connect to a Vocera SIP Telephony Gateway, or Vocera Client Gateway, whether it is a single server or a member of an array.
- The Vocera Voice Server reconnected to a Vocera SIP Telephony Gateway.

For information about how to configure alert settings for your Vocera Voice Server, see the Vocera Voice Server Administration Console Guide.

Telephony Logs

Vocera SIP Telephony Gateway and Vocera Client Gateway all maintain log files used for trouble shooting problems in the **\vocera\logs** directory. These servers use different prefixes for their log files, as shown in the following table:

Table 6: Prefixes for VSTG, and VCG log files

Prefix	Description
vtg-dlog	Vocera SIP Telephony Gateway debug-level log file
vcg	Vocera Client Gateway log file
vcg-dlog	Vocera Client Gateway debug-level log file

For more information about Vocera log files, see the Vocera Voice Server Administration Console Guide. For details on how to configure Vocera SIP Telephony Gateway and Vocera Client Gateway logging, see Configuring Logging on page 57.

High Availability for Gateway Clients

This section discusses high availablity for the Vocera SIP Telephony Gateway and Vocera Client Gateway.

Telephony High Availability

You can install multiple Vocera SIP Telephony Gateway servers — also called a telephony server array — at each site. By installing an array of telephony servers at a site, you can take advantage of the following high availability features:

- **Redundancy** – If one of the telephony servers stops responding, the Vocera Voice Server automatically redirects outbound calls to another available telephony server for uninterrupted service.
- **Scalability** – You can purchase and install as many telephony servers as you need to increase telephony capacity.
- **Load balancing** – For outbound calls, the Vocera Voice Server automatically allocates calls to the least busy telephony server. The PBX equipment handles inbound load balancing.

Important: The Administration Console allows you to specify only one telephony configuration per site. If you deploy multiple telephony servers at one site, all of them must use the same configuration. Each telephony server installed at a site must use the same signaling protocol and have the same capacity.

Generally, all telephony servers at a site will use the same PBX. However, they could use different PBXs as long as all PBXs have the same configuration for the trunks to the telephony servers and the same capabilities for off-PBX dialing (for example, tie lines).

Telephony servers in an array do not communicate with each other. Instead, the telephony servers respond to requests from the Vocera Voice Server. All communication with telephony servers is handled by the Vocera Voice Server.

Telephony in a Multi-Site Environment

The Vocera SIP Telephony Gateway (VSTG) uses a telephony server to communicate with a PBX. You must install and configure a VSTG for each PBX that Vocera will use. When your Vocera deployment supports multiple sites, you can accommodate any of the following configurations:

- An environment where each site has its own PBX.
  Enable and configure a VSTG for each site.
- An environment where sites that do not have their own PBX share with sites that do have a PBX (the principal sites).
  Enable and configure a VSTG for the principal sites only.
- A mixed environment where some sites (the principal sites) have their own PBX, and other sites share a PBX with a principal site.
  Enable and configure a VSTG for each principal site.

If any site—including the Global site—does not have telephony enabled or does not share a PBX with a principal site, the site will not have telephony access.

When you install a VSTG, you specify the IP address of the Vocera Voice Server or the list of nodes in a cluster and the name of the site it is going to connect to. The installation program sets the `VOCERA_SITE` environment variable based on the site name that you specify.

If you do not specify a site, the `VOCERA_SITE` environment variable is not set, and the telephony server is associated with the Global site. As each telephony server boots, it associates itself with the Vocera Voice Server and site specified during the installation.

Note: After installation, you can associate the VSTG with a different site by changing the value of the `VOCERA_SITE` environment variable.

About Shared Telephony

Vocera allows sites to share a PBX with its associated VSTG. However, each physical location that is large enough to have its own PBX should also typically have its own VSTG unless one of the following situations is true:

- The total number of badges in use at that location is very low.
- The total number of badge-to-telephone and telephone-to-badge calls at that location is very low.

In general, the potential problems that can occur when sites share a VSTG are similar to the types of problems that occur when locations share a PBX. For example, if the number of calls exceeds the number of available lines, incoming callers will receive a busy signal, and outgoing callers will hear a message telling them to try again later. Carefully consider your call volume if you plan to share a VSTG.

The use of sites partitions the recognition space and improves speech recognition for large deployments. When multiple sites share a PBX, you must configure separate hunt numbers to realize those speech recognition benefits for incoming callers.

Because separate hunt numbers require coordination with the PBX administrator and possible user re-training, these speech recognition enhancements for incoming callers come with a certain price. There are no firm rules for determining whether the overhead incurred by multiple hunt numbers outweighs the speech recognition benefits they provide; however, following are some guidelines to consider:

- If you are upgrading an existing deployment and the sites sharing the PBX are relatively small, setting up independent hunt numbers provides relatively minor speech recognition benefits compared to the user retraining that would be necessary.
- If you are setting up a new deployment, the user training and PBX administrator coordination required by multiple hunt numbers is essentially the same as that required by a single hunt number, making it relatively easy to take advantage of their speech recognition benefits.

Shared Telephony Deployment Scenarios

The following scenario shares a telephony server between sites. In this example, the telephony server uses the PBX at site A. The telephony server is shared with site B, which may or may not have its own PBX. Because a single telephony server instead of an array of telephony servers is installed at site A, high availability features are not supported.

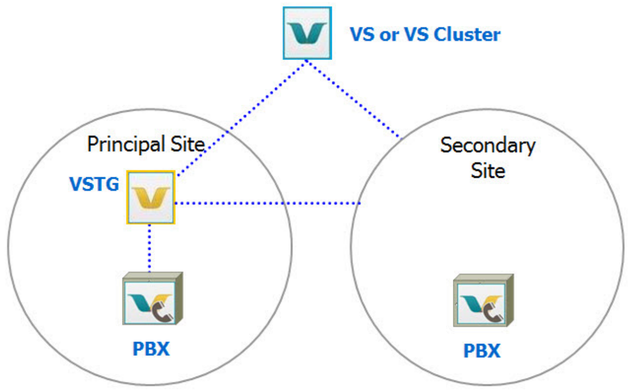


Figure 6: Multiple Site Scenario Using 1 PBX

Summary	
Sites:	Multiple
Telephony Sharing:	Yes
High Availability:	No

The following scenario is a variation of the previous one. An array of telephony servers has been added, which provides redundancy, scalability, and load balancing.

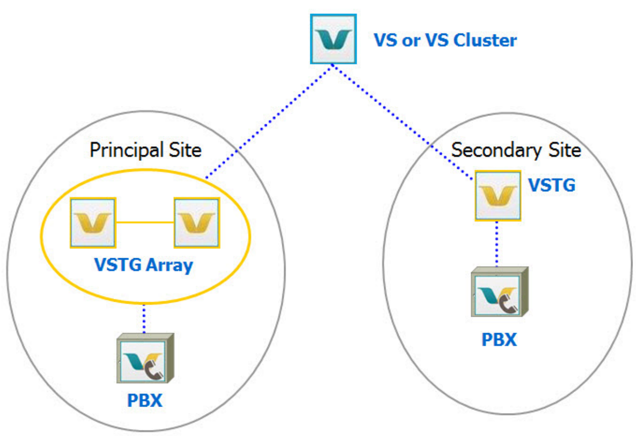


Figure 7: Multiple Site Scenario Using 1 PBX and a VSTG Array

Summary	
Sites:	Multiple
Telephony Sharing:	Yes
High Availability:	Yes

Shared Telephony and Incoming Calls

When an incoming telephone call arrives at the hunt group Genie, the caller can respond to the Genie by saying a name or by using the keypad to enter an extension (DTMF tones). When the telephony server is not shared, Vocera searches the combined grammars of the called site and the Global site to match caller responses.

When the telephony server is shared, Vocera determines the set of grammars to search in either of the following ways:

- If the caller entered DTMF tones, Vocera searches the combined databases of the Global site and every site that shares the telephony server.
- If the caller provided a spoken response, Vocera searches the combined grammars of the Global site and any sites associated with the line that the call arrived on.

Shared Telephony and Outgoing Calls

Vocera does not reserve any lines for outgoing calls. Outgoing calls from any site sharing telephony will use the next available line.

Shared Telephony and Desk Extensions

When a user responds to the hunt group Genie by entering the keypresses for a desk extension, Vocera searches the database of every site that shares a telephony server and also the Global site to find the appropriate user or group for the extension.

Consequently, when you share a telephony server, desk extensions must be unique across all the shared sites and also the Global site, regardless of whether the hunt numbers and range of lines used by those sites are unique.

Connecting to a Site to Use Its Telephony Server

When you use the "Connect to Site" command to connect to another Vocera site, the telephony server and PBX that are used when you make telephone calls depends on whether you are dialing an extension or an outside number:

- **Dialing an extension** – the telephony server and PBX of the site to which you are connected are used.
- **Dialing an outside number** – the telephony server and PBX of your home site are used. Your long distance permissions and PIN information apply to your home site.

For example, say your company has sites in Los Angeles and New York. If your home site is Los Angeles, you can say the command "Connect to New York" to connect to that site. Once you are connected, you can then say "Dial extension 3145". The Vocera Voice Server uses the telephony server for that site to dial the local extension.

If the site that you connect to does not have its own Vocera SIP Telephony Gateway and instead shares the telephony server at another site, then calls are made through the shared telephony server. When you set up shared telephony for a site, you can specify the prefix of the dial string used to place calls through a tie line to the site. This tie line prefix allows you to dial extensions at the remote site even though the site does not have its own Vocera SIP Telephony Gateway Server. For more information, see Configuring Shared Telephony on page 41.

Multi-Site Inbound Redundancy Using DNIS

In a multi-site environment, the Vocera Voice Server uses the Dialed Number Identification Service (DNIS) to determine which site's hunt group number was called and thus which site grammars to use for the call. With additional PBX configuration or utilizing Advanced Routing Features possibly offered by your Central Office, your inbound calls can be routed automatically to the PBX at a different site when one of your telephony servers goes down.

Important: This feature is available only if the target (ending) telephony server uses ISDN or SIP signaling protocol.

The following figure illustrates multi-site inbound redundancy using DNIS:

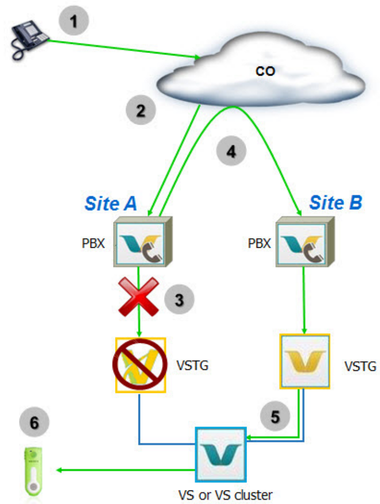


Figure 8: Multiple site inbound redundancy

1. Someone places a call to a badge user at Site A.
2. The Central Office routes the call to Site A based on the called number.
3. The VSTG at Site A is down, so the call is routed back to the Central Office.
4. The Central Office routes the call to another PBX at a different site using an alternate routing plan.
5. The call is received by the Vocera Voice Server, which knows which site grammars to use based on the dialed number.
6. The badge user at Site A receives the call.

If a call made to a site is rejected by the initial PBX for any reason (for example, the VSTG may not be responding), the call can be routed by the Central Office to another PBX at a different site using an alternate routing plan. When the call arrives at the telephony server at the other site, it sends the call to the Vocera Voice Server. The Vocera Voice Server knows which site grammars to use based on the dialed number.

Multi-Site Inbound Redundancy and Shared Telephony

When multiple sites share a PBX, normally you must specify separate hunt numbers and separate ranges of incoming lines for each site to realize speech recognition benefits for incoming callers. However, if the telephony server uses ISDN or SIP signaling protocol and you have configured the PBX properly, you can use the same range of lines for sites that share a telephony server. This allows full use of the lines across multiple sites.

Important: To take advantage of multiple site inbound redundancy features using shared telephony, your PBX must be configured properly and you must have a uniform dialing plan for the sites.

Suppose a deployment has three sites: West Philadelphia, South Philadelphia, and Center City (the principal site). The following table shows the grammars searched for speech recognition at the hunt group prompt when each site has a separate hunt number and all sites share a pool of lines for incoming calls. The system relies on the DNIS to determine which site grammars to use for the incoming call.

Table 7: Shared telephony with one shared pool of lines for all sites

Site	Hunt Numbers	Lines	Grammars Searched
West Philadelphia	215-549-1300 215-549-1301	0-22	• West Philadelphia • Global

Site	Hunt Numbers	Lines	Grammars Searched
South Philadelphia	215-549-2300 215-549-2301	0-22	• South Philadelphia • Global
Center City	215-549-3300 215-549-3301	0-22	• Center City • Global

The following figure illustrates inbound redundancy using DNIS when multiple sites share a VSTG array:

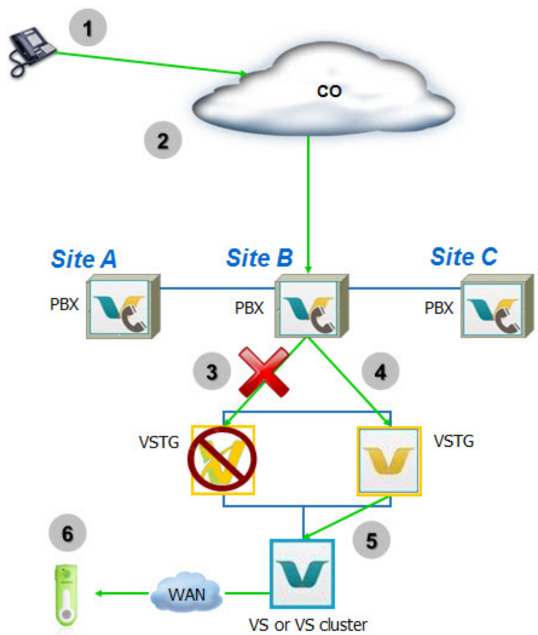


Figure 9: Multiple site inbound redundancy using shared telephony

1. Someone places a call to a badge user.
2. The Central Office routes the call.
3. The PBX skips the VSTG that is down (even though it is in the routing table).
4. The call is routed to the VSTG that is online.
5. The call is received by the Vocera Voice Server, which knows which site grammars to use based on the dialed number.
6. The badge user receives the call.

Vocera SIP Telephony Gateway and PBX Failover Support

For PBX failover support, you can configure Vocera SIP Telephony Gateway to use multiple call signaling addresses. On the **Telephony > Basic Info**page of the Administration Console, select the **Call Signaling Address** field and enter a comma-separated list of call signaling addresses for two or more IP PBXs or VoIP gateways. At startup, Vocera SIP Telephony Gateway tries each PBX or gateway in the order specified and uses the first one that responds. If that PBX or gateway goes down, Vocera SIP Telephony Gateway switches to another one.

The VSTG uses the response to a SIP OPTIONS message to determine if the PBX or gateway is currently available. See Detecting the Connection to the IP PBX on page 68.

In some situations, using TCP as the signaling transport protocol reduces the length of time required for the VSTG to recognize that the current PBX is down and move to the next PBX in the list. see Using UDP, TCP, or TLS Transport to the IP PBX on page 70.

You can override the call signaling address for a particular Vocera SIP Telephony Gateway and have it connect to a different PBX than the one used by other Vocera SIP Telephony Gateway servers in the array. For more information, see Overriding the Call Signaling Address to Connect to a Different IP-PBX on page 70.

Vocera Client Gateway High Availability

You can install multiple Vocera Client Gateway (VCG) servers at each site. When you install an array of client gateways at each site, you take advantage of the following high availability features:

- **Redundancy** – If a client gateways server stops responding, the phone client in that site automatically connects to the next available one for uninterrupted service.
- **Scalability** – You can purchase and install as many client gateway servers as you need to support your phone clients.
- **Load balancing** – The phone clients keep a list of multiple VCGs internally with a random selection algorithm which cycles between them.

Important: VCG in an array do not communicate between each other.

Vocera Client Gateway in a Multi-Site Environment

The Vocera Client Gateway (VCG) allows phone clients to communicate with the Vocera Voice Server. You must install and configure a VCG that the Vocera Smart Phone Client will use. When your Vocera Deployment supports multiple sites, you can accommodate any of the following configurations:

- Enable and configure VCGs for each site.
- Enable and configure VCGs for the Global or default site.

All client devices and the VCG server should be configured to connect to the same site location. If a VCG is not configured for a particular site, by default, the client devices connects to the VCG Global site. When you install a VCG, the user interface provides a field where you specify the IP address of the Vocera Voice Server or the list of nodes in a cluster and the name of the site with which you want the VCG to connect. During the VCG installation, the program sets the **VOCERA_SITE** environment variable according to the the site name that you specify in the user interface.

If you do not specify a site, the **VOCERA_SITE** environment variable is set to Global by default, and the client Gateway server is associated with the Global site. As each client gateway server boots, it associates itself with the Vocera Voice Server and site specified during the installation.

Note:

After installation, you can associate the VCG with a different site by manually changing the value of the **VOCERA_SITE** environment variable.

The figure below shows an example of a VCG array which provides redundancy, scalability, and load balancing.

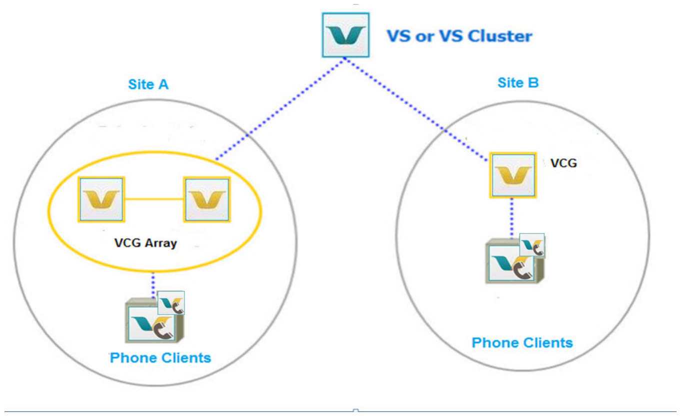


Figure 10: Multiple-site Vocera Client Gateway

Configuring Telephony

You configure Vocera SIP Telephony Gateway settings through the Administration Console's Telephony screen. You also use the Administration Console to configure other data and settings related to your telephony integration. For example, user profiles can include phone numbers and pager numbers, group profiles can specify phone numbers and telephony permissions, and address book entries can include phone numbers and pager number.

Important: Check **Enable Telephony Integration** in the Basic Information page to allow Vocera and your phone system to communicate with each other. Checking this field also enables several other fields in the Basic Information, Access Codes, Toll Info, and PIN pages. When **Enable Telephony Integration** is unchecked, those fields are disabled. You can disable the telephony integration by unchecking this option and clicking **Save Changes**. Vocera saves your telephony settings, but disables communication to the PBX.

Configuring Basic Information

Use fields in the Basic Information page to specify the basic information Vocera needs to communicate with your PBX.

To specify the basic telephony settings:

1. Click **Telephony** in the navigation bar.
2. Click the Basic Info tab.
3. Specify basic telephony settings.
4. Do either of the following:
    - Click **Save Changes** to save these values.
      The telephony server restarts automatically, using your new settings.
    - Click **Access Codes** to continue setting telephony properties.

Table 8: Telephony basic information fields

Field	Description
Select Site	Use the **Select Site** field to choose a site to configure. If you do not have multiple sites, choose the **Global** site.
Enable Telephony Integration	Select **Enable Telephony Integration** to allow Vocera and your phone system to communicate with each other. Selecting this checkbox enables several other fields in the Basic Information, Access Codes, Toll Info, and PIN pages. When the **Enable Telephony Integration** checkbox is cleared, those fields are disabled. You can disable telephony integration by clearing the checkbox and clicking **Save Changes**. Vocera saves your telephony settings, but disables communication to the PBX. **Important:** DO NOT select **Enable Telephony Integration** if the following if the site is a secondary site that uses the shared telephony server of another principal site. If you select **Enable Telephony Integration**, the site automatically becomes a principal site and is therefore removed from the list of secondary sites for all other principal sites.

Field	Description
Vocera Hunt Group Numbers	Specify the area code and phone numbers of the DID lines or hunt group you set up for the Vocera system in the **Vocera Hunt Group Numbers** fields. There are two hunt group number fields: • **Guest Access** – This number is for guest access to the Vocera system. When callers dial the Guest Access number, they are allowed to place a call but are not identified to the called person. Because guest users are not authenticated, they can call other users but they cannot issue voice commands. To use the **Guest Access** number with numeric pagers, enter an asterisk after the last digit of the phone number. When a user sends a numeric page, Vocera passes the value that you enter in this field to the pager, and then passes the user's desk extension to the pager. Some pagers display the asterisk as a hyphen, separating the desk extension from the Vocera number. • **Direct Access** – This number is for specially licensed user access to the Vocera system. This field is used only if Calling and Called Party Information is enabled on the PBX. Vocera uses the Caller ID feature to automatically authenticate users when they call the **Direct Access** phone number from their desk phone or cell phone.
Number of Lines	Specify the number of lines you want to provision for each telephony server in the **Number of Lines** field. Enter either of the following values, whichever is smaller: • The number of lines supported by your license. • The number of lines provisioned by the PBX for a single telephony server. If you are configuring a high availability array, enter the number of lines available to a single telephony server, not the total number of lines available to all servers. For example, if you have 2 VSTG servers and your license contains 48 lines specify 24 in this field to give each server 24 lines for a total of 48. The number of lines that you provision for each telephony server is decremented from the number of lines in your license.
Integration Type	Specify **IP** as the connection type for the PBX since **IP** specifies a VoIP connection to an IP PBX or VoIP gateway and is required for Vocera SIP Telephony Gateway. **Note:** The **Analog** and **Digital** fields are greyed out on the Vocera Admin Counsole user interface and are no longer supported.

Configuring IP and SIP Settings

If you are connecting Vocera to an IP PBX or a VoIP gateway, set the Integration Type field to IP, then specify settings in the following fields:

Table 9: IP and SIP settings

Field	Value
Signaling Protocol	Specify the signaling protocol that your IP PBX uses. Currently only SIP Version 2.0 is supported.

Field	Value
Call Signaling Address	Enter the call signaling address for your IP PBX or VoIP gateway. For PBX failover support, enter a comma-separated list (up to 256 characters) of call signaling addresses for two or more PBXs or gateways in order of preference. Enter each call signaling address in this format: `IP_Address:Port` The port is optional. If you do not specify a port, port 5060 (the default) is used. Vocera SIP Telephony Gateway uses only one PBX or gateway at a time. If you specify multiple call signaling addresses, Vocera SIP Telephony Gateway tries each PBX or gateway in the order specified and uses the first one that responds. If that PBX or gateway goes down, Vocera SIP Telephony Gateway switches to another one. The preference order of call signaling addresses is important. If the Vocera SIP Telephony Gateway is currently using the PBX for the second call signaling address, and then the PBX for the first call signaling address becomes active, Vocera SIP Telephony Gateway automatically switches to the first PBX. **Note:** The Vocera SIP Telephony Gateway uses the response to a SIP OPTIONS message to determine if the PBX or gateway is currently available. The OPTIONS message is sent every 30 seconds by default. For more information on how to configure Vocera SIP Telephony Gateway to use an OPTIONS message for keep-alive, see Detecting the Connection to the IP PBX on page 68. If the PBX or gateway is not configured to support SIP OPTIONS, then entering a second call signaling address has no effect. In some situations, using TCP as the signaling transport protocol reduces the length of time required for the VSTG to recognize that the current PBX is down and move to the next PBX in the list.
Calling Party Number	Enter the DID number, including the area code, of the Vocera trunk (the number of digits depends on the locale). Outgoing calls use this value as the caller ID. However, you can configure Vocera SIP Telephony Gateway to use caller information contained in the dial signal from the Vocera Voice Server as the caller ID. See Configuring Caller Information on page 66.
Enable Call Trace	Click **Enable Call Trace** to enable tracing for a number of calls specified in the Vocera SIP Telephony Gateway configuration file (`vgwproperties.txt`). The default number of calls traced is five. To view the trace, see the `vtg-dlog*.txt` log on the Vocera SIP Telephony Gateway.

Note: If you increase the number of lines and then save changes, it will cause the Vocera SIP Telephony Gateway to restart. If you decrease the number of lines, change the call signaling address, or change the calling party number and then save changes, those changes will be reflected in subsequent calls made through the Vocera SIP Telephony Gateway.

Configuring the Hunt Group Numbers

You need to configure telephone numbers that people can call to access the Vocera system. These telephone numbers are called the **Vocera Hunt Group Numbers**.

There are two hunt group number fields:

- **Guest Access** – This number is for guest access to the Vocera system.
- **Direct Access** – This number is for specially licensed user access to the Vocera system. This field is used only if Calling and Called Party Information is enabled on the PBX.

If you are integrating Vocera with an IP PBX, you specify Direct Inward Dialing (DID) numbers for the **Guest Access** and **Direct Access** numbers.

Note: Vocera provides additional options for configuring hunt groups in a deployment that implements multiple sites. See About Shared Telephony on page 18 for more information.

Calling the Guest Access Number

When a caller dials the **Guest Access** number, the Genie says, “Good morning. Say the full name of the person or group you want to reach or enter an extension.” Callers can then connect to a badge in any of the following ways:

- Speak the name of a person to connect to that user's badge.
- Enter a desk extension to connect to that user's badge.
- Speak the name of a group to connect to the badge of the first available group member.
- Speak the name of a Vocera address book entry.
- Enter the group's telephone extension to connect to the badge of the first available group member.
- Enter the number **555** to receive an additional Genie prompt that allows them to send a voice message to a badge.
- Enter the number **0** (for Operator) to connect to the badge of the first available member of the Operator group, if one exists.
- To switch to user access mode, press the star (*) key. The Genie may prompt for your first and last name, and then it may prompt for your phone access password.

Calling the Direct Access Number

If your Vocera system includes licenses that allow users to access the Genie from a phone, the Vocera System Administrator can enable that feature for certain users and grant the appropriate permission to access the Genie from a phone. For information about enabling user access to the Genie from a phone, see the Vocera Administration Guide.

When a caller dials the **Direct Access** number, the caller is automatically authenticated based on Caller ID, and the Genie says, “Good morning, [FirstName]. [Chime] Vocera." The caller can then say any of the supported commands.

Configuring Access Codes

The Access Codes page lets you specify your local area code and the access codes used by your PBX. An access code is a sequence of digits that the system must prepend to a telephone number in order to dial it. For example, many PBX systems require you to dial a 9 to get an outside line for a local call. In this situation, 9 is the local access code—it is the number that you prepend to a telephone number.

The access codes in use at your site are determined by they way your PBX is set up. When you configure Vocera, you need to identify these access codes so the telephony server can communicate properly with the PBX.

Your PBX may use many different access codes. For example, it may require different access codes to get an outside line for local calls, to get an outside line for toll calls, and to access your company's voicemail system.

Table 10: Access codes fields

Field	Description
Select Site	Use the **Select Site** field to choose a site to configure. If you do not have multiple sites, choose the **Global** site.
Local Area Code	Enter the area code of the region in which the Vocera Voice Server is installed in the **Local Area Code** field.
Omit Area Code when Dialing Locally	If your PBX requires you to dial local calls without using the area code, check **Omit Area Code when Dialing Locally**. By default, Vocera includes the area code in the dialing string, even when dialing a local number. Check this field if your PBX or locale requires you to omit the area code when dialing local calls.
Default Local Access Code	Use the **Default Local Access Code** field to specify the sequence of numbers you use to get an outside line. For example, a PBX might require you to dial a 0 or a 9 or an 8 to get an outside line. By default, Vocera prepends this access code to any number within the local area code.

Field	Description
Default Long-Distance Access Code	Use the **Default Long-Distance Access Code** field to specify the sequence of numbers you enter before placing a long distance call. For example, a PBX system might require you to dial a 9 to get an outside line and then dial a 1 before a long-distance telephone number. In this situation, the Default Long-Distance Access Code is 91. By default, Vocera prepends this access code to any number that includes an area code that is not the local area code.
Company Voicemail Access Code	Use the **Company Voicemail Access Code** field to specify the sequence of numbers you use to access the company's voice mail system. A typical entry includes X, then the sequence of digits that you dial to get into the voicemail system from an internal phone, and possibly special dialing characters such as the * or # to indicate the end of the sequence. See Adding the Voicemail Access Code on page 29 for details.
Access Code Exceptions	By default, numbers in the local area code use the Default Local Access Code and all others use the Default Long-Distance Access Code. Use the **Access Code Exceptions** table to specify exceptions to this policy, as described in Configuring Access Code Exceptions on page 30.

Use the following steps to configure Access Codes:

1. Click **Telephony** in the navigation bar.
2. Click the **Access Codes** tab to display the Access Codes page.
3. Specify access codes and exceptions.

   The list on the Access Codes page displays the telephone numbers that are exceptions to the access code rule—that is, all numbers within the local area code require the Default Local Access Code, and all other numbers require the Default Long-Distance Access Code, unless they appear in the list of exceptions on the Access Codes page.
4. Click **Save Changes**.

Exceptions to Access Codes

By default, Vocera assumes that any number within your local area code requires the Default Local Access Code and that any other number requires the Default Long-Distance Access Code. You use the list on the Access Codes page to specify exceptions to this rule.

For example, if your location has a toll-free area code in addition to your local area code, you may need to dial it with the Default Local Access Code instead of the Default Long-Distance Access Code. You can specify exceptions such as this, or any other exception, in the Access Codes page.

Adding the Voicemail Access Code

Use the **Company Voicemail Access Code** field on the Access Codes page to specify the sequence of characters you use to access the company's voice mail system. Vocera passes this sequence of characters to the PBX when users forward badge calls to their voice mailboxes.

The specific values you need to enter in this field depend upon both your PBX and the way in which it is set up. You always need to enter the actual access code required by your PBX; however, you may also need to enter the **%D** macro to pass a user's desk extension to the PBX, and possibly special dialing characters such as the ***** or **#** to indicate the end of the sequence.

Vocera interprets the data you enter in the **Company Voicemail Access Code** field in either of the following ways:

- If you do not explicitly enter the **%D** macro, Vocera automatically appends the user's desk extension to the end of the sequence you specify before passing it to the PBX.
- If you explicitly enter the **%D** macro, Vocera does not append anything to the sequence before passing it to the PBX.

To set the voicemail access code:

1. Click **Telephony** in the navigation bar.
2. Click the **Access Codes** tab to display the Access Codes page.
3. Enter an X to tell Vocera to treat the following sequence of digits as an extension, without prepending either an access code or an area code to them.
4. Enter the access code your voicemail system requires.

   This value is the sequence of digits that you dial to get into the voicemail system from an internal phone.
5. If your voicemail system typically requires you to pause before dialing the desk extension, and you are connecting to an analog PBX, enter one or two commas (, or ,,) to make Vocera pause briefly.
6. Do either of the following:
   - If your voicemail system requires only an access code and a desk extension to get into a voice mailbox, you do not need to enter anything else. Vocera automatically appends the voice mailbox desk extension when it submits the dialing sequence to the PBX.
   - Continue with the final step in this procedure.
7. If your voicemail system requires you to enter characters in addition to the desk extension, enter the **%D** macro.

   Vocera substitutes the voice mailbox desk extension for **%D** when it submits the dialing sequence to the PBX.
8. Enter any other special characters you need to dial after the desk extension to get into the voice mailbox.

   For example, some systems require you to enter an asterisk (***, also called a star**) or a pound sign (**#, also called a hash sign**) after the desk extension.
9. Click **Save Changes**.

   Vocera saves your voicemail access set up.

For example, suppose you need to dial **5555** to get into the voicemail system from an internal phone at your site, and then you need to enter the desk phone number followed by an asterisk. In this situation, you specify the following value in the **Company Voicemail Access Code** field:

X 5555 , , %D *

Configuring Access Code Exceptions

By default, Vocera uses the following rules to determine what access code to use with a telephone number:

- Any number within your local area code requires the Default Local Access Code.
- Any number that begins with a **0**, begins with an **X**, or has fewer than seven digits does not require an access code. Vocera treats numbers with fewer than seven digits as extensions.
- Any other number requires the Default Long-Distance Access Code.

If your organization uses any phone numbers that violate these rules, you must add entries that provide the access codes they require in the exception list. For example, you need to create an exception if an area code in addition to your local area code requires the Default Local Access Code instead of the Default Long-Distance Access Code.

Adding Access Code Exceptions

Use the Add Access Code Exception dialog box to add an entry to the list of exceptions on the Access Codes page. The Add Access Code Exception dialog box appears when you click the **Add** button on the Access Codes page.

You can add exceptions for an entire area code, for a specific prefix, and for a range of numbers in an area code.

To create an exception for an entire area code:

1. Enter the area code the exception applies to in the **Area Code** field.
2. Check **All Numbers in Area Code**.
3. Specify the access code that this area code requires in the **Use Access Code** field.
4. Do either of the following:
   - Click **Save** to save your entry and close the dialog box.
   - Click **Save & Continue** to save your entry and leave the dialog box open to create a new entry.

For example, suppose your local area code is 408 and your Default Local Access Code is 9. If numbers in the 650 area code are also considered local numbers, create an exception to prevent Vocera from using the Default Long-Distance Access Code for them, as follows:

1. Enter **650** in the **Area Code** field.
2. Check **All Numbers in Area Code**.
3. Enter **9** in the **Use Access Code** field.
4. Click **Save** to save your entry and close the dialog box.
   The 650 area code appears as an exception on the Access Codes page.

To create an exception for a specific prefix:

1. Enter the area code the exception applies to in the **Area Code** field.
2. Check **Numbers Starting With** and enter the prefix in the associated field.
3. Specify the access code in the **Use Access Code** field.
4. Do either of the following:
   - Click **Save** to save your entry and close the dialog box.
   - Click **Save & Continue** to save your entry and leave the dialog box open to create a new entry.

To create an exception for a specific range:

1. Enter the area code the exception applies to in the **Area Code** field.
2. Check **Numbers In Range** and enter the beginning and ending numbers in the associated fields.
   Use a seven-digit range in each field. To create an exception for a single number, enter the same number in both fields.
3. Specify the access code in the **Use Access Code** field.
4. Do either of the following:
   - Click **Save** to save your entry and close the dialog box.
   - Click **Save & Continue** to save your entry and leave the dialog box open to create a new entry.

Editing Access Code Exceptions

Use the Edit Access Code Table Entry dialog box to change an entry in the list of exceptions on the Access Codes page.

Use the following steps to change an entry in the exception list:

1. Click **Telephony** in the navigation bar.
2. Click the **Access Codes** tab to display the Access Codes page.
3. Choose an entry in the list of exceptions, then click **Edit**.
   The Edit Access Code Table Entry dialog box appears. Make any changes that are necessary to the entry.

Deleting Access Code Exceptions

Use the Access Codes page to remove an entry from the list of exceptions.

To remove an entry from the exception list:

1. Click **Telephony** in the navigation bar.
2. Click the **Access Codes** tab to display the Access Codes page.
3. Choose an entry in the list of exceptions, then click **Delete**.
   The entry is removed from the list.

Configuring Toll Information

By default, Vocera assumes that any number within your local area code is a toll-free number, and any number outside your local area code is a toll number. You can use the Toll Info page to specify exceptions to this rule. For example, many locations have an additional area code that is a toll-free calling area, or an exchange within the local area code that is a toll area. In Australia, for example, the area code 04 is reserved for mobile phone numbers, and calls to mobile phones are toll-free.

The distinction between toll-free and toll numbers can be important, because Vocera requires separate permissions for making toll calls, forwarding calls to toll numbers, making toll-free calls, and forwarding calls to toll-free numbers.

To specify whether specific area codes and ranges of phone numbers are toll-free, use the Toll Info page of the Telephony screen in the Administration Console.

Table 11: Toll Information Fields

Field	Description
Select Site	Use the **Select Site** field to choose a site to configure. If you do not have multiple sites, choose the **Global** site.
Toll Info Exceptions	By default, numbers in the local area code are considered toll-free, and others are considered to require toll-call permissions. Use the **Toll Info Exceptions** table to specify exceptions to this policy, as described in Configuring Toll Info Exceptions on page 33.

For example, many locations have an additional area code that is a toll-free calling area, or an exchange within the local area code that is a toll area. In Australia, for example, the area code 04 is reserved for mobile phone numbers, and calls to mobile phones are toll-free. You can specify exceptions such as this, or any other exception, with the Toll Info page.

The distinction between toll-free and toll numbers can be important, because Vocera requires separate permissions for making toll calls, forwarding calls to toll numbers, making toll-free calls, and forwarding calls to toll-free numbers.

To configure toll information:

1. Click **Telephony** in the navigation bar.
2. Click the **Toll Info** tab to display the Toll Info page.
   The list on the Toll Info page displays the telephone numbers that are exceptions to the toll-free rule—that is, all numbers within the local area code are toll-free numbers, and all numbers outside the local area code are toll numbers, unless they appear in the list of exceptions on the Toll Info page.
3. Add, edit, or delete toll info exceptions.
4. Click **Save Changes**.

Configuring Toll Info Exceptions

Use the Add/Edit Toll Info Exception dialog box to add an entry to the list of exceptions on the Toll Info page. You must add an exception to this list in either of the following situations:

- When a number or range of numbers in your local area code is a toll call.
- When a number or range of numbers outside your area code is a toll-free call.

Adding Toll Info Exceptions

You can create toll info exceptions for an entire area code, for a specific exchange, and for a range of numbers in an area code.

To create an exception for an entire area code:

1. Enter the area code the exception applies to in the **Area Code** field.
   Enter toll-free prefixes such as 800 and 888, as well as any area codes (such as 04 in Australia) that are toll-free in your dialing area.
2. Check **All Numbers in Area Code**.
3. Use the **Toll-Free?** field to specify a toll-free or a toll area code.
4. Do either of the following:
   - Click **Save** to save your entry and close the dialog box.
   - Click **Save & Continue** to save your entry and leave the dialog box open to create a new entry.

For example, to specify that a call to any 800 number is a toll free call:

1. Enter **800** in the **Area Code** field.
2. Check **All Numbers in Area Code**.
3. Check the **Toll-Free?** field.
4. Click **Save** to save your entry and close the dialog box.
   The 800 prefix appears as a toll-free prefix in the list on the Toll Info page.

To create an exception for a specific exchange:

1. Enter the area code the exception applies to in the **Area Code** field.
2. Check **Numbers Starting With** and enter the exchange in the associated field.
3. Use the **Toll-Free?** field to specify a toll-free or a toll exchange.
4. Do either of the following:
   - Click **Save** to save your entry and close the dialog box.
   - Click **Save & Continue** to save your entry and leave the dialog box open to create a new entry.

For example, to specify that calls to the 427 exchange in your local 408 area code are toll calls:

1. Enter 408 in the **Area Code** field.
2. Check **Numbers Starting With** and enter **427** in the associated field.
3. Uncheck the **Toll-Free?** field.
4. Click **Save** to save your entry and close the dialog box.
   The 427 exchange appears as a toll exchange in the list on the Toll Info page.

To create an exception for a range of numbers:

1. Enter the area code the exception applies to in the **Area Code** field.
2. Check **Numbers In Range** and enter the beginning and ending numbers in the associated fields.
   Use a seven-digit range in each field. To create an exception for a single number, enter the same number in both fields.
3. Use the **Toll-Free?** field to specify whether you are adding a toll-free or a toll exchange.

4. Do either of the following:
   - Click **Save** to save your entry and close the dialog box.
   - Click **Save & Continue** to save your entry and leave the dialog box open to create a new entry.

For example, to specify that the first 5000 numbers in the 427 exchange in your local 408 area code are toll calls:

1. Enter `408` in the **Area Code** field.
2. Check **Numbers In Range** and enter `427-0000` in the beginning field and `427-4999` in the ending field, so the range appears as **427-0000 To 427-4999**.
3. Uncheck the **Toll-Free?** field.
4. Click **Save** to save your entry and close the dialog box.
   The 427-0000 to 427-4999 range appears as a range of toll numbers in the list on the Toll Info page.

Editing Toll Info Exceptions

Use the Edit Toll Table Entry dialog box to change an entry in the list of exceptions on the Toll Info page.

Use the following steps to change an entry in the exception list:

1. Click **Telephony** in the navigation bar.
2. Click the **Toll Info** tab to display the Toll Info page.
3. Select an entry in the list of exceptions, then click **Edit**.
   The Edit Toll Table Entry dialog box appears. Make any changes that are necessary to the entry. See Configuring Toll Info Exceptions on page 33 for complete information.

Deleting Toll Information Exceptions

Use the Toll Info page to delete an entry from the list of exceptions.

Use the following steps to delete a toll info exception:

1. Click **Telephony** in the navigation bar.
2. Click the **Toll Info** tab to display the Toll Info page.
3. Select an entry in the list of exceptions, then click **Delete**.
   The entry is removed from the list.

Configuring Direct Inward Dialing

Your PBX administrator may reserve one or more groups of DID (Direct Inward Dialing) extensions for Vocera badges to use. When an outside caller dials a number within a specified DID range, the call goes directly to the badge of the associated user. Otherwise, the Genie prompts the caller to say the full name of the person or group, or enter an extension.

Use the DID page of the Telephony screen to specify the range of DID extensions that are available for use by Vocera users. If your PBX administrator provides a hunt group number as part of the DID range, do not include it in the range of DID extensions you specify here. The extensions on this page are for use by users only.

Table 12: DID information fields

Field	Description
Select Site	Use the **Select Site** field to choose a site to configure: • If you are configuring telephony for a site that has its own telephony server, specify the name of that site. • If you are sharing a telephony server among multiple sites, specify the principal site. All the DID ranges that you specify form a single pool that the principal site shares with any secondary sites. See About Shared Telephony on page 18 for information about the principal site.
Direct Inward Dialing	View the prefixes and range of extensions available for Vocera user DID numbers.

About Direct Inward Dialing

In traditional telecommunications, Direct Inward Dialing (DID, or DDI in Europe) is the ability of a person outside an organization to call an internal PBX extension without going through an operator or intermediate interface of any kind. In Vocera, DID is the ability of a caller anywhere to place a telephone call directly either to a user's badge or to a group, without going through the hunt group Genie or any speech recognition prompts.

Vocera supports DID if you are using Vocera SIP Telephony Gateway and SIP signaling protocol. This feature is powerful because it allows callers who are not aware of Vocera or its features to contact users directly on their badges. DID extends the benefits of Vocera to telephone callers who do not necessarily even belong to the organization that is deploying Vocera.

To enable DID, your PBX administrator must reserve a range of DID numbers for Vocera to use, and you must identify that range to Vocera. Use the DID Info page of the Telephony screen in the Administration Console to specify the range of DID numbers reserved for Vocera.

Tip: The DID numbers that you specify must be full 10-digit telephone numbers with area code in the US locale (or full numbers with city and region codes, in other locales).

If your PBX administrator provides a hunt group number as part of the DID range, enter it as the hunt number in Vocera, but do not include it in the range of DID numbers that you configure on the DID info page. User and group profiles may be assigned the DID numbers that you specify in the Administration Console, and you do not want a user or group to have the same extension as the hunt number.

If an incoming call arrives on a number that is within the specified DID range, but the number is not assigned, Vocera automatically directs the call to the hunt group Genie.

When multiple sites are sharing a PBX, they also share the single pool of DID numbers that are enabled in the primary site. You cannot distribute different ranges of DID numbers to individual sites that share a PBX.

When multiple sites are using different PBXs, each PBX may provide a different range of DID numbers, or even none at all. The way each PBX is configured determines whether its associated sites have access to DID.

Note: DID numbers may be more expensive and more difficult to obtain than other PBX extensions. You do not need to have a dedicated DID number for every badge to receive some of their benefits. See Configuring Dynamic Extensions on page 39.

See Configuring Direct Inward Dialing on page 34 for complete information on setting up DID.

Adding and Editing DID Information

When you add or edit DID information, you specify a prefix and the range of phone numbers to use for direct inward dialing.

Your PBX administrator may provide you with discontinuous ranges of DID extensions or even groups of DID extensions with different prefixes. Enter each range separately until they all appear in the list on the DID Info page of the Telephony screen.

For example, your PBX administrator may supply 100 DID extensions with the following ranges:

- (215) 995-4150 through (215) 995-4199
- (215) 885-6880 through (215) 885-6899
- (215) 885-6920 through (215) 885-6949

You can enter each range separately in the Add DID Range Entry dialog box to make them all available to Vocera.

To add or edit DID information:

1. Click **Telephony** in the navigation bar to display the Basic Info page.
2. Click the **DID Info** tab to display the Direct Inward Dialing (DID) page.
3. Specify the site these DID extensions are associated with in the **Select Site** field.
4. Click **Add** to create a new range of DID extensions, or choose a range in the list and click **Edit** to edit an existing range.
   The Add/Edit DID Range dialog box appears.
5. Enter the area code and prefix assigned to the range in the **Prefix** field.

   For example, if the local area code of the PBX is 408, and the corporate prefix for all extensions is 790, you typically enter `(408)-790`. In some situations, your PBX administrator may assign a different prefix for you to use.

   To provide maximum flexibility, Vocera does not check the value you enter in this field. If necessary, you may enter country and city codes, as well as extensions whose length is shorter or longer than four digits. For example, if your deployment has five-digit extensions, you may want to enter a prefix such as `(408)-79`.

   Important: Enter the area code and full prefix that make a complete dialing string when combined with a value in the range of extensions. Vocera combines the extension and the value in the **Prefix** field to create a call-back number for paging.
6. In the **Match** section, choose one of the following options to define the range of DID numbers:
   - Choose **All Desk Extensions with Prefix** to use the entire range of numbers represented by the value in the **Prefix** field.
     Examples:
     - If the value in the **Prefix** field is `(408)-790`, you are assigning the range `(408)-790-0000` through `(408)-790-9999` as DID extensions. The extensions available for assignment to Vocera users and groups are 000 through 999, 0000 through 9999, or 00000 through 99999.
     - If the value in the **Prefix** field is `5`, you are assigning any number that starts with a "5". The extensions available for assignment to Vocera users and groups are 500 through 599, 5000 through 5999, or 50000 through 59999.
   - Choose **Desk Extensions Starting With** and specify a starting value to use a subset of the range of numbers represented by the value in the **Prefix** field.
     Examples:
     - If the value in the **Prefix** field is `(408)-790`, and you enter `8` in the **Desk Extensions Starting With** field, you are assigning the range `(408)-790-8000` through `(408)-790-8999` as DID extensions. The extensions available for assignment to Vocera users and groups are 8000 through 8999.

    - If the value in the **Prefix** field is `(408)-790`, and you enter **94** in the **Desk Extensions Starting With** field, you are assigning the range `(408)-790-9400` through `(408)-790-9499` as DID extensions. The extensions available for assignment to Vocera users and groups are 9400 through 9499.
    - If your PBX passes 59xx to Vocera, enter `5` in the **Prefix** field and `9` in the **Desk Extensions Starting With** field. This means you are assigning the range `5900` through `5999` as DID extensions. The extensions available for assignment to Vocera users and groups are 900 through 999.
  - Choose **Desk Extensions In Range** and enter beginning and ending values value to specify a range of phone numbers within the set represented by the value in the **Prefix** field. This is the most typical situation.

    Examples:
    - If the value in the **Prefix** field is `(408)-790`, and you enter `8000 To 8999` in the **Desk Extensions In Range** field, you are assigning the range `(408)-790-8000` through `(408)-790-8999` as DID extensions. The extensions available for assignment to Vocera users and groups are 8000 through 8999.
    - If the value in the **Prefix** field is `5`, and you enter `501 To 549` in the **Desk Extensions In Range** field, you are assigning the range `5501` through `5549` as DID extensions. The extensions available for assignment to Vocera users and groups are 501 through 549.
7. Do either of the following:
   - Click **Add** to add your entry to the list and close the dialog box.
   - Click **Add & Continue** to add your entry to the list and leave the dialog box open to create a new range.

Deleting DID Information

Use the DID Info page to delete an entry from the list of direct inward dialing (DID, also called DDI in Europe) information.

To delete DID information:

1. Click **Telephony** in the navigation bar to display the Basic Info page.
2. Click the **DID Info** tab to display the Direct Inward Dialing (DID) page.
3. Select an entry in the list of DID information, then click **Delete**.
   A dialog box asks you to confirm the deletion.
4. Click **OK**.
   The entry is removed from the list.

Configuring Telephony PINs

A telephony PIN (Personal Identification Number) allows an organization to authorize telephone usage and to distribute telephone costs among different users, departments, or sites. Some organizations use the term FAC (Forced Authorization Code or Forced Access Code) to describe this feature.

For example, a company might require employees to enter a PIN along with a phone number to make a long distance or toll call. Vocera's telephony PIN feature automatically adds a PIN to the dialing sequence when a user places a call that requires one. In addition to long distance and toll calls, a PIN is also used for long distance forwarding, transferring, and paging.

Note: A user cannot make toll calls—and telephony PINs have no effect—unless he or she belongs to a group that allows toll calls.

Use the PIN page of the Telephony screen in the Administration Console to configure PINs.

Telephony PINs can be assigned at one or more of the following levels, listed in descending order of precedence:

1. **User profile**. If a user's profile specifies a telephony PIN, it is used each time the user places a call that requires a PIN.
2. **Department group**. If a telephony PIN is not specified in the user's profile, but the user belongs to department group to which a PIN has been assigned, then that PIN is used.

   When a user belongs to more than one department that has a telephony PIN assigned, a PIN is chosen at random from among those departments. Thus, costs are shared evenly among the user's various departments. To override this behavior, specify a PIN in the user's profile; for example, you could enter the PIN of the department to which the user's long-distance calls are billed.
3. **Site**. If a telephony PIN is not specified in the user's profile and the user does not belong to a department group that has a PIN, then the PIN specified for the user's site is used.

If there is no user PIN, no department PIN, and no site PIN, then no telephony PIN is used.

About Telephony PIN Fields

A telephony PIN (Personal Identification Number) allows an organization to authorize telephone usage and to distribute telephone costs among different users, departments, or sites. Some organizations use the term FAC (Forced Authorization Code or Forced Access Code) to describe this feature. Use fields in the PIN page to configure telephony PINs.

Table 13: PIN fields

Field	Maximum Length	Description
Select Site	n/a	Use the **Select Site** field to choose a site to configure. If you do not have multiple sites, choose the **Global** site.
PIN for Long Distance Calls	75	Use the **PIN for Long Distance Calls** field to specify a PIN for a site. If a telephony PIN is not specified in the user's profile and the user does not belong to a department group that has a PIN, then the site PIN is used. The site-level telephony PIN is used for long distance numbers specified in address book entries, as well. It is also used for group forwarding numbers, unless the group is department group with a PIN number specified, in which case the department group PIN is used.
PIN Template	75	Use the **PIN Template** field to specify a template for adding a PIN to a dialing sequence. When a dialing sequence includes a PIN, this value defines the format that the Vocera system uses to send it to the PBX. Every site that has its own PBX can define a PIN and a PIN template. Sites that share a PBX use the PIN and PIN template defined for the Global site. A PIN template can include digits, special characters, and PIN macros.

Each PBX has different rules for adding a PIN to a dialing sequence. Some require the phone number followed by the PIN. Some require the PIN before the phone number. Some require an access code for an outside line, or a feature code to indicate that a number is a PIN. Some require a separator character between the PIN and the number. A telephony PIN template specifies and formats the information in a PIN.

Specifying Telephony PIN Information For a Site

1. Click **Telephony** in the navigation bar.
2. Click the Basic Info tab to display the **Basic Info** page. In this page, the **Enable Telephony Integration** check box must be checked before you can enter information in the PIN page.
3. Click the **PIN** tab to display a page where you can enter a site-level PIN and define a template that specifies how to integrate a PIN into a dialing sequence.
4. Use the **Select Site** field to choose the site you want to configure.

Every site that has its own telephony server can specify a PIN and a PIN template. Sites that share a telephony server use the PIN and PIN template defined for the **Global** site. To configure a single-site environment, choose the **Global** site.

5. (Optional) Type a number in the **PIN for Long Distance Calls** field to define a site-level PIN.
6. Type numbers, formatting characters (for example, dashes or parentheses) special dialing characters (for example, commas or ampersands), and PIN macros in the **PIN Template** field. This template defines the format of all PINs, whether they are defined at the user, department group, or site level.

   If no PIN template is specified, the Vocera system applies one of the following default templates, depending on the type of PBX:

Table 14: Default PIN templates

PBX type	Default template	Description
IP	%N %P	Access code, phone number, and PIN.

Configuring Dynamic Extensions

To allow Vocera users to receive paging call-backs on their Vocera device, each user must have a unique extension entered in their Vocera profile. You must enter a value in either the **Vocera Extension** field or the **Desk Phone or Extension** field for each user. You can assign these values manually, or you can let Vocera assign them as dynamic extensions.

About Dynamic Extensions

Dynamic extensions are artificial telephone numbers that Vocera associates with users automatically, on an as-needed basis, if they need a number to enable a paging call-back on the badge. You can use dynamic extensions in either of the following situations:

- Vocera users do not have actual desk extensions and you want Vocera to assign an extension to users automatically rather than use the **Vocera Extension** field.
- You are using DID, but you don't have enough DID numbers to dedicate one to each Vocera user.

  You can use dynamic extensions to share a small amount of DID numbers among a greater number of Vocera users, enabling recipients of a numeric page to place a return call directly to a user's device, without going through the hunt group Genie.

Use the Dynamic Extensions page of the Telephony screen to enable and configure dynamic extensions. Vocera determines which of the above two situations apply based on the **Extension Range** you enter:

- If the range of numbers you enter for dynamic extensions is equal to or a subset of the range you entered for DID, Vocera assumes you are distributing DID numbers among your Vocera users.
- If the range of numbers you enter for dynamic extensions is outside the range of numbers you entered for DID, Vocera assumes that you want to assign desk extensions to users independently of DID.

Tip:

If you are using DID, set the dynamic extension range to be the same as the DID range. If the dynamic extension range is a subset of the DID range, some DIDs will not be used.

If you enter a range of numbers that include both DID numbers and non-DID numbers, Vocera still distributes all of them on an as-needed basis, starting from the beginning of the range. However, users may unpredictably have DID numbers some times and non-DID numbers other times. Vocera recommends that you avoid this configuration unless you need to use it to solve a specific communication problem.

You do not need to keep track of which users have which dynamic extensions; Vocera automatically tracks and allocates all numbers in the dynamic range. For example, if a user is deleted from the system, Vocera automatically returns that dynamic extension to the pool of available extensions.

If a user already has a desk extension or a Vocera extension, Vocera will never assign him or her a dynamic extension. Instead, Vocera will use the Vocera extension or the desk extension for paging or DID. In a mixed environment where some users have desk extensions and others don't, Vocera assigns dynamic extensions on demand to anyone who does not have either a desk extension or a Vocera extension.

Specifying Dynamic Extensions for a Site

The Dynamic Extensions page of the Telephony screen lets you configure Vocera to supply telephone extensions on demand to users who need them. Dynamic extensions affect only users whose profile does not include a Vocera extension or a desk extension. See the Vocera Voice Server Administration Console Guide for information about phone fields for users.

Vocera assigns dynamic extensions to users in a manner analogous to a DHCP server assigning IP addresses to client computers. See Configuring Dynamic Extensions on page 39 for complete information.

Use the following steps to specify dynamic extensions for a site:

1. Click **Telephony** in the navigation bar.
2. Click the Basic Info tab to display the **Basic Info** page. In this page, the **Enable Telephony Integration** check box must be checked before you can enter information in the Dynamic Extensions page.
3. Use the **Select Site** field to choose the site you want to configure.
4. Select the **Enable Dynamic Extensions** check box.
5. Enter values in the **First Extension** field and the **Last Extension** field to specify a range of phone numbers to use as dynamic extensions.
   - If the range is equal to or a subset of the range you entered for DID, Vocera assumes you are distributing DID extensions among your Vocera users.

     Tip: If you are using DID, set the dynamic extension range to be the same as the DID range. If the dynamic extension range is a subset of the DID range, some DIDs will not be used.

   - If the range is not a subset of the range you entered for DID, Vocera assumes that you want to assign desk extensions to users independently of DID.
6. Specify a value in the **Assignment Type** field in either of the following ways:
   - Choose **Permanent** to assign Vocera users extensions that do not expire.

     Permanent is useful when Vocera users do not have actual desk extensions, but you want them to have a unique identifier that allows recipients of a numeric page to place a return call to the badge.
   - Choose **Temporary** and assign a **Lease Duration** value to specify the minimum amount of time that an extension will be assigned to a user. **Temporary** is useful when you want to share a small number of DID extensions among a larger number of Vocera users.

     By default, the lease is set to seven days to allow safe paging callbacks several days later. If you don't have enough DID numbers for all Vocera users, you can set the lease duration to hours instead of days so that numbers can be reallocated as needed.
7. Click **Save Changes**.

Duration of Dynamic Extensions

When you configure dynamic extensions, you determine whether users will lease them for a specified duration or keep them permanently. When an assigned extension reaches the end of its lease, it expires and can be reassigned to another user.

However, Vocera avoids reassigning a dynamic extension as long as possible, even after it expires, in a manner analogous to a DHCP server assigning IP addresses:

- You specify a range of extensions to populate the pool of available values.

- When extensions are leased, the duration specifies the minimum amount of time that an extension will be assigned to a user. By default, the lease is set to seven days to allow safe paging callbacks several days later.
- If a user has a lease that has not expired and needs an extension again (for example, when sending another page), Vocera automatically renews the lease on the same extension for a new default duration.
- If a user has a lease that has expired, Vocera will not assign that extension to another user until all other available extensions have been exhausted.
- If a user with an expired lease requests an extension again, Vocera will assign the same lease, if it is still available.

As Vocera assigns dynamic extensions, they appear in the **Dynamic Extension** field on the Phone page of the Add/Edit User dialog box. The **Dynamic Extension** field is read-only and is displayed for informational purposes only. As extensions are renewed and expire, Vocera automatically updates this field.

Because extensions are assigned on-demand, the **Dynamic Extension** field may be empty even after you enable the dynamic extensions feature. Similarly, the **Dynamic Extension** field will continue to display an expired number that has not been reassigned, indicating that the user will keep the number as long as it is available.

Sites and Dynamic Extensions

If multiple sites are sharing a telephony server, they also share the single pool of dynamic extensions that are enabled in the primary site. You cannot distribute different ranges of dynamic extensions to individual sites.

If multiple sites are using different telephony servers, you may assign each site a different range of dynamic extensions, or even none at all.

Configuring Shared Telephony

When you configure two or more sites to share a telephony server, enable telephony for one site only. The site for which telephony is enabled is considered the principal site. Sites that use the shared telephony server of a principal site are called secondary sites.

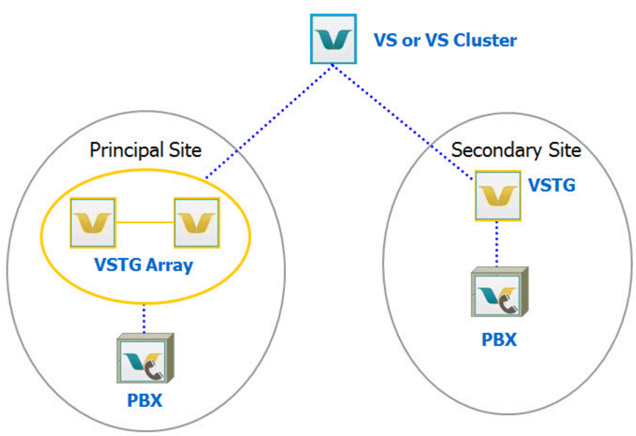


Figure 11: Telephony shared from principal to secondary site

Do not enable telephony for secondary sites that use the telephony server of a principal site. Instead, use the **Telephony > Sharing** page of the Administration Console to configure the principal site to share the telephony server with the other sites.

On the Sharing page of a principal site's Telephony screen, click the Add button to display the Add Shared Telephony Info dialog box. Specify the name of the site that is sharing the principal's telephony server, then do one of the following:

- **If the sites have the same hunt group numbers:**
  Leave the **Guest Access Number**, **Direct Access Number**, and **Reserve Range of Lines** fields blank.
  This configuration specifies that Vocera will search the combined grammars of the principal and sharing site when incoming callers respond to the hunt group Genie.
- **If the sites have different hunt group numbers:**
  If using Vocera SIP Telephony Gateway for telephony integration, enter values in the **Guest Access Number** and **Direct Access Number** fields. Leave the **Reserve Range of Lines** fields blank.
  This configuration specifies that Vocera will search only the grammars for a site as specified by the hunt group numbers or line numbers when incoming callers respond to the hunt group Genie.

The Sharing page of the Telephony screen lets you configure the information necessary to allow multiple sites to share a telephony server.

Table 15: Sharing fields

Field	Description
Select Site	Use the **Select Site** field to specify the name of the principal site. When you configure two or more sites to share a telephony server, the site for which telephony is enabled is considered a “principal site.”
Other Sites Sharing This Telephony Server	Displays each site that is using the shared telephony server of the principal site.

Adding and Editing Telephony Sharing Information

When you add or edit telephony sharing information, you specify the names of the secondary sites that are using the principal site's shared telephony server. You also optionally specify the hunt group numbers and trunk lines used by the secondary sites.

To add or edit telephony sharing information:

1. Click **Telephony** in the navigation bar to display the Basic Info page.
2. Click the Sharing tab to display the **Sharing Telephony Server Across Sites** page.
3. Specify the principal site in the **Select Site** field.

   Note: You must enable telephony for the principal site before you share it. Use the Basic Info page of the telephony screen to enable the telephony integration, as described in Configuring Basic Information on page 25.
4. Click **Add** to specify sharing information for a secondary site, or choose a site in the list and click **Edit** to modify existing sharing information.
   The Add/Edit Shared Telephony Site dialog box appears.
5. Enter or edit the following information:

   Table 16: Shared telephony information fields

Field	Description
Site	Specify a secondary site that will use the shared telephony server of the principal site. **Important:** The site you select should not be a principal site with telephony enabled. If you select a principal site with telephony enabled, the system automatically disables telephony for that site.
Reserved Range of Lines for Incoming Calls	This field is no longer supported. **Note:** For Vocera SIP Telephony Gateway, leave the **Reserved Lines** field blank.

Field	Description
Guest Access Number	Optionally specify the area code and phone number of a DID line or hunt group for this site. This number is for guest access to the Vocera system. When callers dial the **Guest Access Number**, they are allowed to place a call but are not identified to the called person. Because guest users are not authenticated, they can call other users but they cannot issue voice commands. • If you want the secondary site to have the same Guest Access number as the primary site, leave this field blank or enter the primary site's Guest Access number. • If you want the secondary site to have a different Guest Access number, coordinate with your PBX administrator, then enter the number.
Direct Access Number	Optionally specify the area code and phone number of a DID line for this site. This number is for specially licensed user access to the Vocera system. This field is used only if your Vocera system has a digital or IP connection to the PBX, you have selected ISDN or SIP signaling protocol, and Calling and Called Party Information is enabled on the PBX. Vocera uses the Caller ID feature to automatically authenticate users when they call the Direct Access number from their desk phone or cell phone.
Tie Line Prefix	Specify the prefix of the dial string used to place calls through the tie line to the selected site that is sharing the principal's telephony server. Alternatively, this field could also be used to specify a prefix for Direct Inward Dialing (DID) numbers at the selected site. For tie lines, enter the tie prefix plus the tie line. For example, if the tie prefix is 8 and the tie line is 257, enter `8-257`. For DID numbers, identify the DID prefix by determining the constant digits that become the prefix to an extension to produce a full DID number. For example, if the format of your DID numbers is 408-882-*nnnn*, the DID prefix is `408-882`. In the US locale, the full DID number must be a 10-digit telephone number that includes the area code. In other locales, full DID numbers include city and region codes. The Vocera Voice Server prepends the Tie Line Prefix to the extension dialed to generate the complete dial string for the selected site. If the selected site does not have its own PBX and a tie line or DID numbers, leave this field blank. **Note:** The Tie Line Prefix is used for all extensions dialed for the selected site. Only one Tie Line Prefix can be used per shared site.
Calling Party Number	Enter the DID number, including the area code, of the Vocera trunk (the number of digits depends on the locale). Outgoing calls use this value as the caller ID. However, you can configure Vocera SIP Telephony Gateway to use caller information contained in the dial signal from the Vocera Voice Server as the caller ID. When this field is filled in, the Vocera site telephony caller ID appears on the client device of the target (called user) making it easier for them to return the call.

6. Do either of the following:
   - Click **Add** to add this site to the list and close the dialog box.
   - Click **Add & Continue** to add this site to the list and leave the dialog box open to create a new range.

Deleting Telephony Sharing Information

Use the Sharing page to delete an entry from the list of sites that share a telephony server.

Use the following steps to delete telephony server sharing information:

1. Click **Telephony** in the navigation bar to display the Basic Info page.
2. Click the Sharing tab to display the **Sharing Telephony Server Across Sites** page.
3. Specify the principal site in the **Select Site** field.
4. Choose an entry in the list of sites, then click **Delete**.
   A dialog box prompts you to confirm your decision.
5. Click **OK**.
   The entry is removed from the list.

Configuring Cisco Integration

Use fields on the Cisco page to specify information to allow Vocera to integrate with Cisco Unified Communications Manager (CUCM) and Cisco wireless IP phones (7921G, 7925G, and 7926G).

Important: Vocera integration with CUCM currently supports only one CUCM per site. Do NOT check the **Enable Cisco Integration** checkbox until after the CUCM and the Cisco wireless IP phones have been properly configured for Vocera Connect for Cisco.

Table 17: Cisco information

Field	Value
Site	Select a site to use for this Cisco Unified Communications Manager (CUCM). You can use the Sharing tab to share this Cisco configuration with other sites.
Enable Cisco Integration	Make sure this box is checked to enable integration with CUCM. **Important:** Integration with CUCM requires Vocera Connect for Cisco client application licenses. Otherwise, users of Cisco wireless IP phones will not be able to connect to the Vocera Voice Server. To check your current Vocera Voice Server licenses in the Administration Console, see the Vocera Voice Server Administration Console Guide. To obtain additional licenses, contact Vocera.
Access Number	Enter the voice access number for CUCM. This number should match the route pattern/number for the Vocera SIP trunk. You can find route patterns in CUCM Console by choosing **Call Routing > Route/Hunt > Route Pattern**. This number may be different from the outgoing Calling Party Number entered for the Vocera SIP Telephony Gateway on the **Telephony > Basic Info** page, which is used for Caller ID purposes.
CUCM Information	**IP Address** – Enter the IP address of the CUCM in dotted-decimal notation (for example, 192.168.15.10). **Login ID** – Enter the Vocera application user ID for CUCM. **Password** – Enter the Vocera application user password for CUCM. **Re-enter Password** – Re-type the same password you entered in the **Password** field.
Vocera Line Range	Specify the first line and last line used for the internal range of Vocera lines. **Important:** Vocera supports only one range of lines for Cisco integration. However, the lines are not real DID numbers. Go ahead and make the range large enough to accommodate future growth. You must assign 2 Vocera lines for each phone; in other words, for 50 phones you will need 100 lines.
Enable Extension Mobility	Check this box to enable the Extension Mobility service on the phones. With Extension Mobility, users can access their phone configuration from other Cisco Unified Wireless IP Phones. If Extension Mobility is enabled, it will automatically log users into Vocera when they are authenticated on Cisco wireless IP phones; separate Vocera login is not needed. **Important:** Additional configuration is required on CUCM devices to support Extension Mobility. Do NOT check this box until the CUCM devices have been configured accordingly.

Working with Pagers

If your site has the telephony integration option, Vocera users can issue voice commands to send numeric pages to anyone with a pager. For example, a Vocera user can speak the command "Page Dr. Shostak" to send a numeric page to someone who is either another user or an address book entry. Similarly, the "Dial a Pager Number" command allows users to send a numeric page to any arbitrary number.

Vocera supports both inside and outside pagers:

- An inside pager is used with a service that allows employees to send each other numeric pages internally. These pages typically go through the company's PBX, preventing outside users from sending pages. The numbers employees use to send an inside page are often fewer than seven digits.
- An outside pager is used with a service that allows employees to send numeric pages to full-length phone numbers.

Templates in the Vocera Voice Server's **\vocera\server\properties.txt** file define the dialing patterns Vocera uses to call inside and outside pager numbers. These templates also determine the way any callback information is formatted on the pager's display.

About Vocera Paging

To allow Vocera users to receive paging call-backs on their Vocera device, each user must have a unique extension entered in their Vocera profile. You must enter a value in either the **Vocera Extension** field or the **Desk Phone or Extension** field for each user.

If users do not have actual desk extensions and you want Vocera to assign an extension to users automatically rather than use the **Vocera Extension** field, you can use the Dynamic Extension feature to assign extensions to users. See Configuring Dynamic Extensions on page 39.

When a Vocera user issues one of the "Page" commands, Vocera dials the pager number, pauses briefly, and then passes the pager a formatted string to display. The information the recipient sees depends on the type of integration:

Table 18: How Vocera paging works

Type of Integration	What the Page Recipient Sees	What Page Recipient Does
IP where user has DID number	DID number of the Vocera user	• Dials the DID number of the Vocera user. Call is connected directly to the user's Vocera device without any Genie prompts.
IP where user does not have DID number	DID number of the Vocera system, followed by the user's extension	1. Dials the DID number of the Vocera system. 2. At the telephony Genie prompt, enters the user's extension. Call is connected to the user's Vocera device.

Regardless of the integration type, Vocera always routes the return call to the user's Vocera device, not the user's extension, because the callback went through the telephony server. Vocera uses the user's extension only to identify the user and route the call appropriately.

Pagers and DID Numbers

If possible, assign DID numbers to users who frequently send pages. When users have DID numbers, recipients of pages can return calls directly to the Vocera device without going through the Vocera telephony Genie.

DID numbers provide more convenient paging callbacks for recipients. In addition, some pagers cannot properly format a message containing a phone number with more than 10 digits, so DID numbers may be less confusing to the recipient of the page.

See Configuring Dynamic Extensions on page 39.

Paging Progress Indicator

Due to the nature of the SIP protocol Vocera users will not hear DTMF tones when sending a page

Configuring Paging

Use Administration Console to configure paging.

To allow paging interactions to take place, provide the configuration information in the Administration Console.

To enable users to send pages:

1. Specify a Vocera extension or a desk phone number in each user's profile. Use either an actual desk phone number, a unique number that you enter manually, or an artificial number that Vocera provides through dynamic extensions.

   The Vocera extension, desk phone, or dynamic extension allows the user to receive a callback on a Vocera device from the recipient of the page.
2. Specify the email address in each user's profile to integrate the user with Vocera Messaging Platform (VMP). When the email address is not configured, the page is not sent as an alert through VMP.
3. Users do not require explicit permission to send pages to other users. Any user can send a page to any other user who has a pager number defined.

To enable users to receive pages:

1. Specify a pager number in the profile of each user who has a pager.
2. Assign users who have pagers either the **Have Toll-Free Pager Number** or the **Have Toll Pager Number** permission.

   Users who have permission to receive numeric pages can use the following voice commands to specify whether they want to receive pages:
   - Enable pages
   - Disable pages
3. Specify a pager number for address book entries who have pagers.

   No permissions are required for address book entries.

Pagers and Subscriber IDs

Some paging services provide subscriber IDs to distinguish among different individuals. In this situation, subscribers typically share a single phone number issued by the paging service, and individual users are identified by their unique subscriber IDs.

For example, to send a numeric page to a person who uses this system you could call a phone number such as 1-800-555-1111, listen to a message prompting you to provide a subscriber ID, and then enter an ID such as 4545. Some services also require you to enter a special character such as the pound symbol (#, also called a hash symbol) to indicate the end of the ID.

When you specify the pager phone number for users or address book entries that have such a paging service, you must enter both the phone number of the paging service and the subscriber ID in the **Pager** field of the Administration Console. Use a semicolon to separate the toll-free number from the subscriber ID, and provide a special character to terminate the subscriber ID, if required. (The semicolon causes Vocera to pause until the pager is ready to receive the numbers to display.)

For example, if the number of the paging service is **(800) 555-1111,** the subscriber ID of the recipient is **4545**, and the paging service requires a pound (hash) symbol to terminate the subscriber ID, enter the following value in the **Pager** field:

(800) 555-1111 ; 4545 #

When a Vocera user issues the Page command, Vocera dials the pager number, waits until the connection is established, and then passes the service the subscriber ID of the person being paged followed by the pound (hash) symbol to indicate the end of the PIN. At the end of this sequence, Vocera automatically passes the pager the hunt group/DID number of the Vocera system followed by the desk extension of the user who called, or simply the user's DID number, if he or she has one. The pager displays everything following the pound symbol.

The pager's owner returns the call by doing any of the following:

- Dialing the hunt group number and then entering the Vocera user's extension at the telephony Genie prompt.
- Dialing the DID number of the Vocera system and then entering the Vocera user's extension at the telephony Genie prompt.
- Dialing the user's DID number.

In all cases, Vocera then connects the return call to the user's Vocera device—not to the desk phone.

Customizing Pager Strings in the Properties File

Properties in `\vocera\server\properties.txt` allow administrators to configure the strings that the Vocera Voice Server uses to dial pagers and the strings that the pagers display. For example, some environments require trunk access codes to enable a call to an inside pager; other environments may want the pager to display only the extension of the Vocera hunt group or DID number, not the full dialing string. You edit these templates to allow Vocera to support the paging requirements of your environment. In a multi-site installation, these properties apply to all sites.

Note: If you modify the **Properties.txt** file, you must stop and start the Vocera Voice Server to load the properties into memory.

The following properties format the values passed to pagers.

Table 19: Pager properties

Property	Description
TelOutsidePageSetUp	Formats the string passed to a pager outside the Vocera system. The default value of this property is %N;%V%D. %N refers to the pager number Vocera will call, based on one of the following values: • An outside number specified in the voice command, "Dial a pager number." • The value entered in the Pager Number field of an address book entry • The value specified in Pager Number field of a user profile, if the number is not preceded by an X. %V refers to the Vocera hunt group number (analog integration) or DID number (digital integration). %D refers to the user's extension (either the **Desk Phone or Extension**, **Vocera Extension**, or dynamic extension, whichever applies). Administrators will not typically need to modify this default value.

Property	Description
TelInsidePageSetUp	Formats the string passed to a pager inside the Vocera system. The default value of this property is %N;%V%D. %N refers to the pager number Vocera will call, based on one of the following values: • A number specified in a voice command such as, "Page number 4321." • The value entered in the Pager Number field of an address book entry • The value specified in Pager Number field of a the user profile, if the number is 1 to 6 digits long, or 7 digits long and preceded by an X. %V refers to the Vocera hunt group or DID number. %D refers to the user's extension (either the **Desk Phone or Extension**, **Vocera Extension**, or dynamic extension, whichever applies). Administrators may want to modify the default value. Imagine the following scenario: • An organization's telephony system requires a user to dial 64 to get to the trunk. • Users want pagers to display only the 4-digit extension of the Vocera hunt number, not the full ten-digit number. For example, if the Vocera hunt number is 408-790-4170, the pager should display 4170. • The paging system uses an asterisk to indicate a call-back number. In this situation, an administrator would provide the following value for this property: 64%N;4170*%D
TelOutsidePageSetUpForDialIn	Formats the string passed to an outside pager by a person calling into the Vocera hunt group. The default value is %N;%X. %N refers to the pager number Vocera will call (the value entered in the Pager Number field of the user profile or address book entry). %X refers to the call-back number the user enters when prompted by the Genie. Administrators will not typically need to modify this default.
TelInsidePageSetUpForDialIn	Formats the string passed to an inside pager when a person calls into the Vocera hunt group. The default value is %N;%X. %N refers to the pager number Vocera will call (the value entered in the Pager Number field of the user profile or address book entry). %X refers to the call-back number the user enters when prompted by the Genie. Administrators will not typically need to modify this default value.
TelOutsidePageSetUpForDID	Formats the string passed to an outside pager by a Vocera user placing the page who is assigned a DID in the Vocera system. The default value is %N;%I. %N refers to the pager number Vocera will call (the value entered in the Pager Number field of the user profile or address book entry). %I represents the full DID number that is used as the call-back number. Administrators will not typically need to modify this default.
TelInsidePageSetUpForDID	Formats the string passed to an inside pager by a Vocera user placing the page who is assigned a DID in the Vocera system. The default value is %N;%I. %N refers to the pager number Vocera will call (the value entered in the Pager Number field of the user profile or address book entry). %I represents the full DID number that is used as the call-back number. Administrators will not typically need to modify this default.

The following figures show how these properties can be used.

The first figure shows the flow of events that occur when a Vocera user pages someone whose pager number is outside the Vocera system. This example uses the default value of the **TelOutsidePageSetUp** property.

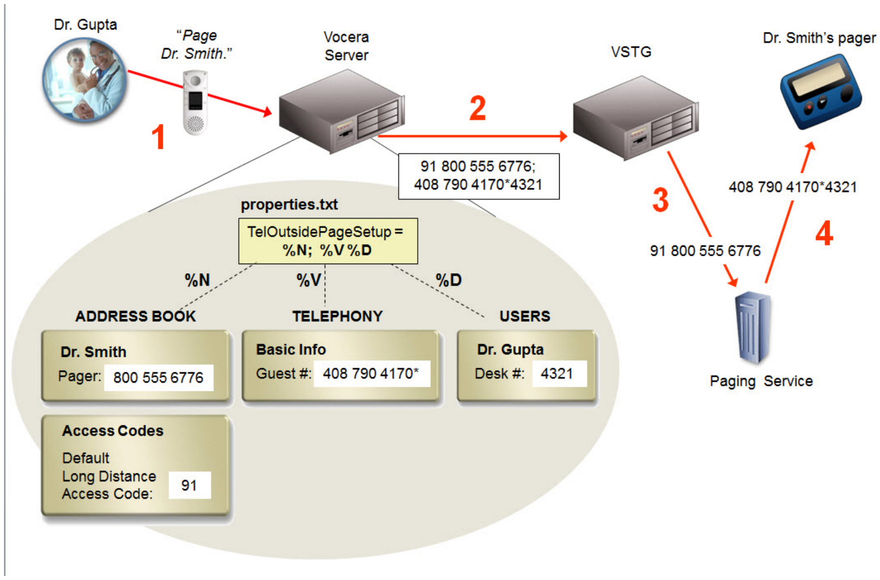


Figure 12: Paging an outside number

The next figure shows the flow of events that occur when a Vocera user pages someone whose pager number is inside the Vocera system. In this example, the value of **TelInsidePageSetUp** has been changed to meet the following criteria:

- The organization's telephony system requires a user to dial 64 to get to the trunk.
- Users want pagers to display only the 4-digit extension of the Vocera hunt number, not the full ten-digit number. For example, if the Vocera hunt number is 408-790-4170, the pager should display 4170.
- The paging system uses an asterisk to indicate a call-back number.

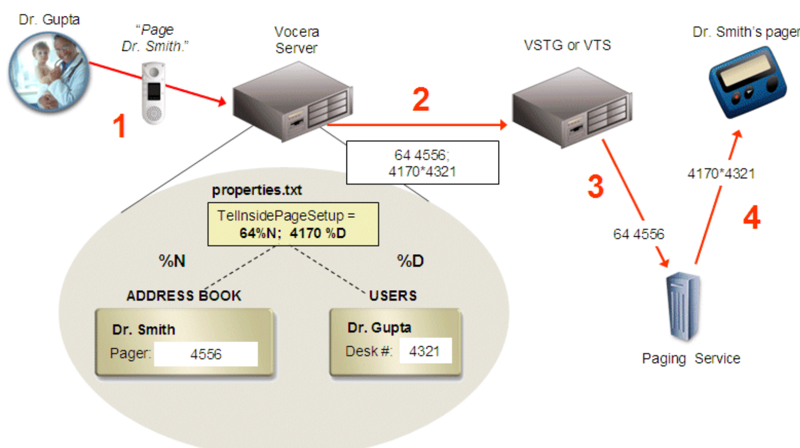


Figure 13: Paging an inside number

Specifying Fixed-Length Numbers

As a best practice, you should constrain telephone extensions and pager numbers to a fixed length. Fixed-length numbers can improve speech recognition, because they eliminate from the grammar all possibilities that are not of the specified length. The following properties in `\vocera\server\properties.txt` specify the number of digits in fixed-length phone extensions and pager numbers.

Table 20: Telephony properties for fixed-length numbers

Property	Description
TelExtensionLength	Specifies the number of digits in an extension. The default value is 0, which means that a variable length is accepted. A value of 1 through 7 specifies an extension of the given length.
TelPagerNumberLength	Specifies the number of digits in an internal pager system number. The default value is 0, which means that a variable length is accepted. A value of 1 through 7 specifies a pager number of the given length. It can be used in conjunction with the Page Number voice command (for example, "Page number 1234").

Note: If you modify the `Properties.txt` file, you must stop and start the Vocera Voice Server to load the properties into memory.

Configuring VMI Telephony Properties

If your Vocera installation includes Vocera SIP Telephony Gateway for telephony integration, you can configure trunk access codes (or TACs) to specify how specific dial strings are processed to define how a Vocera badge interacts with telephony equipment via a VMI client application.

These properties are designed for use with VMI applications in the following situations:

- You need to adjust the badge volume for calls from other devices. For example, if badge users are having trouble hearing calls from bedside speakers in a nurse call system, these properties can help.
- A system requires a special key sequence to end a device-to-badge call after the badge user hangs up.

The collection of VMI telephony properties must be complete. If you comment out one property in the collection, you must comment out the entire collection. You can, however, specify one or more empty values for a property in this collection. By default, the VMI telephony properties are undefined and therefore disabled. The following table describes these properties.

Table 21: VMI telephony properties

Property	Description
TelVMIDeviceTAC	Specifies a trunk access code (TAC) to identify a device (such as a nurse call system) that connects to a PBX to communicate with a Vocera badge. By default, this property value is not defined. When it is defined, this value activates the gain specified by the corresponding TelVMIRxGain property, and the macro defined by the corresponding TelVMIHangUpMacro property. The TAC for any given device is set by the PBX administrator. To specify TACs for multiple devices, use a forward slash (/) as a separator character. White space is ignored. You can specify up to 50 TACs. If two or more TACs begin with the same sequence of characters, list them in descending order of length. For example, each of the following TACs begins with the sequence 12: 12, 123, and 12345. In the properties file, you would list them in the following order: `12345 / 123 / 12`
TelVMIRxGain	Specifies how much gain is added when a badge user chooses the callback option to respond to a VMI message. By increasing or decreasing this value, you increase or decrease the sound level (volume) of the badge speaker in 6 dB increments. For example, a value of 3 increases the volume by 18 dB (3 * 6 = 18). Valid values range from 0 to 6, inclusive. By default, this property value is not defined. Optimum values should be determined in the field by trial and error. The specified gain is applied only if the corresponding TelVMIDeviceTAC property is defined. The gain is removed when the call ends. To specify gains for multiple devices, use the forward slash (/) as a separator character. White space is ignored. You can specify up to 50 gain values.
TelVMIHangUpMacro	Specifies a sequence to dial when a Vocera badge ends a call initiated using the callback option in response to a VMI message. This property is especially useful when interacting with a device that connects to a PBX via an analog line. The required sequence varies depending on the device. For example, nurse call systems from different vendors require different hang-up sequences. Consult the device documentation for details. The specified sequence is dialed only if the corresponding TelVMIDeviceTAC property is defined. To specify more than one macro, use the forward slash (/) as a separator character. White space is ignored. You can specify up to 50 macros.

When you specify more than one value for any of these properties, the order is important:

- If two or more TACs begin with the same sequence of characters, list them in descending order of length when you specify values for the TelVMIDeviceTAC property.

  Vocera's parser processes a dial string from left to right, and when it finds a sequence of digits that matches a value specified for TelVMIDeviceTAC, it interprets that sequence as the TAC portion of the dial string. Therefore, given a dial string of 1234914087904100 and two TelVMIDeviceTAC property values listed in the order 12/1234, the parser interprets the first match, 12, as the TAC. However, when the same property values are listed in the order 1234/12, the first match is 1234.
- The TelVMIRxGain and TelVMIHangUpMacro values are associated with a TelVMIDeviceTAC value, so you must list all property values in the same order. That is, the first TelVMIDeviceTAC value corresponds to the first TelVMIRxGain value and the first TelVMIHangUpMacro value, and so on.

For example, suppose a VMI client application interacts with a nurse call system made by company NC1, a blood pressure monitoring system made by company BP, and another nurse call system made by company NC2. The following code lists some sample values for this scenario:

```
#                     NC1    BP     NC2
#-----------------------------------
```

```
TelVMIDeviceTAC   = 835 / 7812 / 781
TelVMIRxGain      =  4  /      / 2
TelVMIHangUpMacro = ##  /      / *9*
```

In this example, a gain value of 4 and the hang-up macro ## are defined for nurse call system NC1, which has the TAC 835. Similarly, a gain value of 2 and the hang-up macro *9* are defined for nurse call system NC2, which has the TAC 781. However, the blood pressure monitor BP, which has the TAC 7812, does not define a gain value or a hang-up macro. In the properties file, such "empty" values can either be omitted or specified explicitly with spaces. Also, the TAC for BP is listed before the TAC for NC2 because both TACs begin with the sequence 781 and the TAC for BP is longer than the TAC for NC2. Listing the TACs in this order ensures that the Vocera parser will extract them correctly from a dial string.

Additional Setup

The Telephony pages in the Administration Console provide the basic information that enables Vocera and the PBX to communicate with each other. You can take full advantage of the telephony integration by performing additional setup tasks.

Additional Telephony Configuration Tasks

Use the Administration Console to set up permissions and phone numbers as follows:

- Specify which groups have calling and forwarding permissions.
- Specify the desk phone extension, Vocera extension, home phone number, and cell phone number of Vocera users.
- Specify which users are enabled to access the Genie from a phone, and specify the default phone password for those users.
- Specify a range of phone numbers to use for dynamic extensions. This feature is useful when Vocera users do not have physical desk phones but still need to supply a call-back number when paging someone. See Configuring Dynamic Extensions on page 39 for more information.
- See Working with Pagers on page 45 for details about configuring Vocera to work with pagers.
- Specify the telephone extension and forwarding information for groups.
- Create address book entries that all users can share.

See the Vocera Voice Server Administration Console Guide for complete information.

Additional Telephony Configuration by Users

Users can create or modify their own phone numbers in the User Console as follows:

- Specify their desk phone extension, home phone number, cell phone number, and pager number.
- Specify a telephone number for forwarding badge calls.
- Specify telephone numbers to contact outside buddies.
- Specify the telephone extension and forwarding information for groups that they manage.
- Specify the phone password used to authenticate users when they access the Genie from a phone.

See the Vocera Voice Server User Console Guide for complete information.

Customizing the Prefix Used for Urgent Broadcasts

By default, Vocera uses the prefix 666 for urgent broadcasts. A user can enter the sequence 666 followed by a group's telephone extension to make an urgent broadcast to the badges of all members of the group.

To change the prefix for urgent broadcasts, add the following property to the **\vocera\server\properties.txt** file:

Table 22: TelBroadcast property

Property	Description
TelBroadcast	Prefix used for urgent broadcasts to the desk extension for a group. The default value is 666.

Note: If you modify the `Properties.txt` file, you must stop and start the Vocera Voice Server to load the properties into memory.

For example, the following entry in **\vocera\server\properties.txt** sets the prefix for urgent broadcasts to 557:

```
# Prefix for urgent broadcasts
TelBroadcast = 557
```

Figure 14: TelBroadcast example

Configuring VSTG and VCG Properties

You can configure telephony settings for the Vocera Voice Server and Vocera SIP Telephony Gateway in the Administration Console. See Configuring Telephony on page 25.

Note: The Administration Console is not used to configure the Vocera Client Gateway, which is configured solely in its properties file.

For advanced configuration of Vocera SIP Telephony Gateway and Vocera Client Gateway, you can edit a properties file called `vgwproperties.txt`. You can use any text editor to modify the file. If you make any changes, you must restart the Vocera SIP Telephony Gateway or Vocera Client Gateway to load the properties into memory.

Vocera SIP Telephony Gateway and Vocera Client Gateway Architecture

Vocera SIP Telephony Gateway is a SIP telephony gateway between the Vocera Voice Server and an IP PBX or a VoIP gateway.

Vocera Client Gateway supports VCS Client , providing a signaling and multimedia gateway from the phones to the Vocera Voice Server for all calls. Vocera Client Gateway also provides a tunnel for application data between the Vocera smartphone and the Vocera Voice Server.

Because Vocera SIP Telephony Gateway and Vocera Client Gateway are both SIP gateways, they share many of the same components, including the same properties file for configuration.

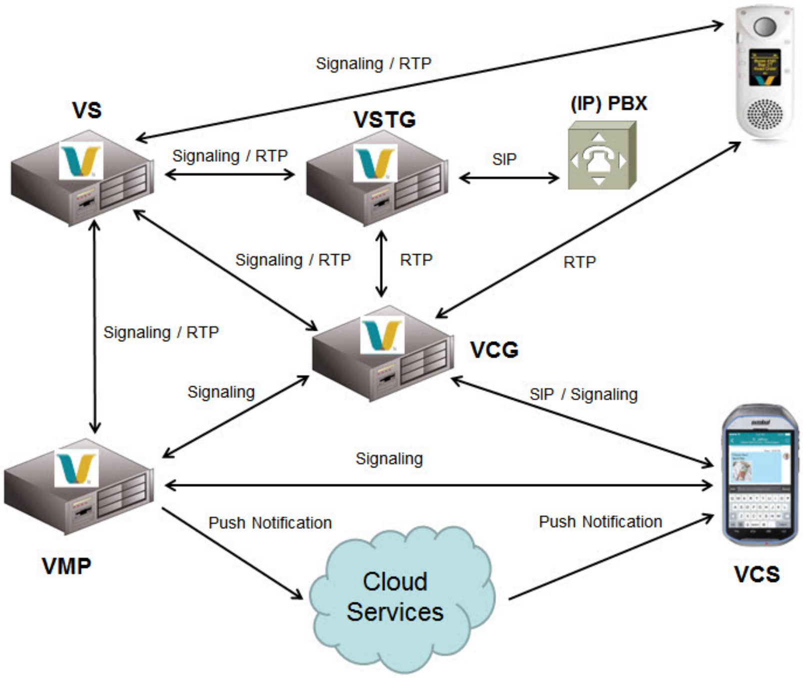


Figure 15: Vocera SIP Telephony Gateway and Vocera Client Gateway architecture

About vgwproperties.txt

The `vgwproperties.txt` file lets you set properties that control the behavior of the Vocera SIP Telephony Gateway or Vocera Client Gateway. The file is located in the `\vocera\telephony\vgw` folder on the computer where Vocera SIP Telephony Gateway or Vocera Client Gateway is installed. It is a simple text file, and you can use any text editor, such as Wordpad, to edit it.

This section describes how to set properties in `vgwproperties.txt` for both Vocera SIP Telephony Gateway and Vocera Client Gateway. Generally (although there are exceptions), property names have the following prefixes that help you identify whether a property applies to Vocera SIP Telephony Gateway, Vocera Client Gateway, or both:

Table 23: Prefixes for property names

Prefix	Description
VTG	A property of Vocera SIP Telephony Gateway
VCG	A property of Vocera Client Gateway
VGW	A property of both Vocera SIP Telephony Gateway and Vocera Client Gateway

Important: For Vocera Client Gateway, you should only modify properties related to Vocera Client Gateway ports, logging capabilities, and jitter tolerance. For all other Vocera Client Gateway properties, the default settings are recommended.

Modifying vgwproperties.txt

To modify `vgwproperties.txt`:

1. On the Vocera SIP Telephony Gateway or Vocera Client Gateway computer, open the `\vocera\telephony\vgw\vgwproperties.txt` file in a text editor.
2. Make changes to properties.

3. Save the `vgwproperties.txt` file.
4. Stop the Vocera SIP Telephony Gateway or Vocera Client Gateway and start it again. In the Control Panel for Vocera SIP Telephony Gateway or Vocera Client Gateway, choose **Run > Stop**, and then choose **Run > Start**.

Configuring Ports

This section describes how to configure ports for Vocera SIP Telephony Gateway and Vocera Client Gateway.

Vocera SIP Telephony Gateway Ports

When you open the `vgwproperties.txt` file, you can set the following properties to configure Vocera SIP Telephony Gateway ports:

Table 24: Vocera SIP Telephony Gateway port properties

Property	Description
VTGCallSignalingPort	Sets which port that Vocera SIP Telephony Gateway listens on for SIP signaling. The default is 5060.
VTGBaseAudioPortNo	Sets the base UDP port for receiving audio packets from Vocera badges and smartphones. The default is 5300. By default, Vocera SIP Telephony Gateway opens up to 256 UDP ports to receive audio from Vocera badges and smartphones. **Important:** If you change this property to something other than 5300, you must also add the following property to the `\vocera\server\properties.txt` file on the Vocera Voice Server, and then stop and start the server to load the property: `IPBasePhonePortNo = PortNumber` where *PortNumber* is equal to the port specified for `VTGBaseAudioPortNo`.
VTGRTPBasePort	Sets the base UDP port for receiving audio packets from the IP PBX or VoIP gateway. The default is 8700. Each client session uses an RTP and an RTCP port. Therefore, the total number of UDP ports for Vocera Voice Server or Vocera Client Gateway to Vocera SIP Telephony Gateway audio is two times the number of **VTGMaxChannelsSupported**, which is set to 256 by default, resulting in 512 ports allocated by default.

Vocera Client Gateway Ports

When you open the `vgwproperties.txt` file, you can set the following properties to configure Vocera Client Gateway ports:

Table 25: Vocera Client Gateway port properties

Property	Description
VCGCallSignalingPort	Sets the port to use for SIP signaling from Vocera Smartphones. The default is 5060.
VCGBaseAudioPortNo	Sets the base UDP port for receiving audio packets from Vocera Voice Server, Telephony Server, or Vocera badges. The default base UDP port is 6300. By default, Vocera Client Gateway opens up to 256 UDP ports to receive audio from the Vocera Voice Server, Telephony Server, or Vocera badges.

Property	Description
VCGRTPBasePort	Sets the base UDP port for receiving audio packets from Vocera Smartphones. The default is 7700. Each client session uses an RTP and an RTCP port. Therefore, the total number of UDP ports for Smartphone to Vocera Client Gateway audio is two times the number of **VCGMaxChannelsSupported**, which is set to 256 by default, resulting in 512 ports allocated by default.

Configuring Logging

In the `vgwproperties.txt` file, Vocera SIP Telephony Gateway and Vocera Client Gateway provide several properties that let you control logging.

Table 26: Logging properties

Property	Description
PJSIPLogLevel	Controls the level of logging from the PJSIP stack. Specify a value from 1 (least amount of logging) to 10 (greatest amount of logging). Generally, you should set this to 5 or lower unless you are experiencing problems. The default is 4.
VCGLogOPTIONSPayload	Specifies whether to log the payload in the OPTIONS message for a call. The default is FALSE. If you want to see what's being passed for debugging purposes, set it to TRUE.
VGWConsoleLoggingLevel	Sets the logging level for the console. Specify 0 to 8, 0 being the least amount of logging and 8 being the most. The default is 4 (DEBUG). 0 = FATAL 1 = ERROR 2 = WARNING 3 = INFO 4 = DEBUG 5 = PJSIP 6 = SIP_MSGS 7 = DEBUG_VERBOSE 8 = RTP_INFO If you specify 5 (PJSIP), the amount of PJSIP logging depends on the **PJSIPLogLevel** property.
VGWFileLoggingLevel	Sets the logging level for the log file. Specify 0 to 8, 0 being the least amount of logging and 8 being the most. See **VGWConsoleLoggingLevel** for the list of logging levels. The default is 6 (SIP_MSGS).
VGWMaxLinesPerLogfile	Sets the maximum size of each log file. The default is 70,000 lines.
VGWNumLogFilesToKeep	Sets the maximum number of log files to keep for the current gateway process. If you have plenty of disk space available and you do not want to delete older log files, you can set this property to 0. The default is 50. **Note:** Each time the gateway service is stopped and started, it will create up to **VGWNumLogFilesToKeep** new log files.

Property	Description
VGWMaxDaysToKeepLogfiles	Sets the maximum number of days to keep log files for the current gateway process. The default is 21 days. When the Vocera SIP Telephony Gateway or Vocera Client Gateway is started, it finds log files that have not been modified within the **VGWMaxDaysToKeepLogfiles** period, and it deletes those files. This keeps the logs directory from growing too large. To preserve all log files and not automatically delete them, set **VGWMaxDaysToKeepLogfiles** to 0.
VCGLogMaxLines	Sets the maximum size of each VCG console log file. This property does not affect the size of debug log files. The default is 100,000 lines.
VCGLogMaxFiles	Sets the maximum number of VCG console log files. The default is 100 files.
VTGLogMaxLines	Sets the maximum size of each VSTG console log file. This property does not affect the size of debug log files. The default is 100,000 lines.
VTGLogMaxFiles	Sets the maximum number of VSTG console log files. The default is 100 files.

Both Vocera SIP Telephony Gateway and Vocera Client Gateway maintain log files in their own `\vocera\logs` directory. The names of these files begin with the **vtg** and **vcg** prefixes, respectively. The names of debug-level logs, which have more detail than the console logs, begin with the **vtg-dlog** and **vcg-dlog** prefixes, respectively.

Each Vocera SIP Telephony Gateway and Vocera Client Gateway log statement has a preamble consisting of four sections:

- The date the event happened.
- The time the event happened.
- The ID of the thread generating the log statement and the level of the log statement (for example, INFO or DEBUG).
- The device ID (MAC address) in brackets.

Note: If a device is not associated with a log statement, 12 dashes appear in the brackets.

Configuring Jitter Tolerance and Jitter Buffer Settings

This section describes how to configure Vocera SIP Telephony Gateway and Vocera Client Gateway to handle jitter for RTP packets on the inbound or outbound Vocera side.

Jitter Tolerance

Jitter is the variation in the time between packets arriving, which could be caused by network congestion. The Vocera badge and smartphone both employ a jitter buffer to let the end user experience uninterrupted audio with very little sound distortion. Both the Vocera SIP Telephony Gateway and the Vocera Client Gateway are intermediate stops for audio packets. To avoid adding unnecessary latency, they do not employ a jitter buffer by default.

You can tune how the Vocera SIP Telephony Gateway and Vocera Client Gateway handle jitter from RTP packets by setting the jitter tolerance properties. The Vocera SIP Telephony Gateway and Vocera Client Gateway keep track of when it received the last RTP packet and when the next one should arrive. The jitter tolerance properties specify how long the gateway service should wait before it decides the next packet is not coming. If it determines the next packet is not coming, it inserts silence packets on the other side of the call. By modifying the jitter tolerance properties, you can ensure that the Vocera SIP Telephony Gateway and Vocera Client Gateway do not aggressively drop delayed audio packets and replace them with silence.

When you open the `vgwproperties.txt` file, you can set the following jitter tolerance properties:

Table 27: Jitter tolerance properties

Property	Description
VGWUseVRTPJitterTolerance	Enables or disables jitter tolerance for RTP packets. By default, this is set to TRUE, meaning jitter tolerance is enabled.
VGWVRTPJitterTolerance	Jitter tolerance for RTP packets sent from the Vocera Voice Server, Telephony Server, or Vocera badges. The default is 7 packetization periods. For RTP packets, each packetization period is 18 milliseconds, so 7 packetization periods is equal to 126 milliseconds.

Here's an example illustrating how the jitter tolerance properties work. Suppose Vocera Client Gateway received a VRTP packet p0 at time t0. The next packet, p1, should arrive at the following time:

$$t1 = t0 + (\text{packetization period} * \text{number of frames per packet})$$

The time at which Vocera Client Gateway stops waiting for packet p1 to arrive is defined by the following equation:

$$t\_giveup = t1 + (\text{packetization period} * \text{VGWVRTPJitterTolerance})$$

If packet p1 has not arrived by the t_giveup time, the Vocera Client Gateway will give up on packet p1 and insert silence into the buffer used to send RTP.

If packet p2 arrives while the gateway is waiting for packet p1, then silence will be inserted for packet p1. If packet p1 subsequently arrives, it will be discarded.

Important: If you modify the jitter tolerance properties, you must test the audio on a Smartphone-to-Smartphone call to ensure the audio quality is good and there are not too many dropped packets.

Jitter Buffer

Optionally, you can choose to employ a jitter buffer for the Vocera SIP Telephony Gateway or Vocera Client Gateway to prevent lost packets. However, a jitter buffer may increase the audio latency for calls.

When you open the `vgwproperties.txt` file, you can set the following jitter buffer properties:

Table 28: Jitter buffer properties

Property	Description
VGWUseRTPJitterBuffer	Enables a jitter buffer for RTP packets. By default, this is set to FALSE, meaning the jitter buffer is disabled.
VGWUseVRTPJitterBuffer	Enables a jitter buffer for RTP packets. By default, this is set to FALSE, meaning the jitter buffer is disabled.

Property	Description
RTPJitterMaxSize	Sets the maximum number of packets in the RTP jitter buffer. The default is 10.
RTPJitterUseFixed	Whether to used a fixed or adaptive jitter buffer. The default is TRUE (fixed).
RTPJitterPrefetch	The number of packets to pre-buffer. The default is 3.

Specifying the Companding Algorithm (mu-law or a-law)

By default, the Vocera SIP Telephony Gateway uses the G.711 mu-law companding algorithm, a standard for converting analog data into digital form using pulse code modulation in North America and Japan. For European locales, you can switch to the G.711 a-law companding algorithm by specifying the following property:

```
VTGUseALaw = true
```

For outbound traffic, Vocera SIP Telephony Gateway uses the audio codec specified by the `VTGUseALaw` property for negotiation with the IP PBX, and then it adjusts to the codec that the IP PBX offers. If `VTGUseALaw` is set to `FALSE` (the default), G.711 mu-law is used. Otherwise, G.711 a-law is used.

For inbound traffic, Vocera SIP Telephony Gateway converts the audio to G.711 mu-law codec before sending it to the Vocera Voice Server.

Note: Although there is a `VCGUseALaw` property for Vocera Client Gateway, you should not change its default setting from `False` to `True`. The Vocera Voice Server, smartphone, and badge all use G.711 mu-law.

Configuring Call Tracing

In the Vocera Administration Console, you can enable call tracing to debug calls going through the Vocera SIP Telephony Gateway. The following properties let you configure call tracing:

Table 29: Call tracing properties

Property	Description
VGWTraceCallsNumCallsToLog	When you click the **Enable Call Trace** button on the **Telephony > Basic Info** page in the Administration Console, the Vocera SIP Telephony Gateway logs all SIP messages for a specific number of calls set by this property. The default number of calls is 5. The log levels for both log files and the Vocera SIP Telephony Gateway console are set to LL_SIP_MSGS temporarily. Logging levels are restored to their previously set levels after the number of trace calls has been reached.
VGWTraceCallsAffectsConsoleToo	Whether call tracing affects console logging. The default is TRUE.

How the Vocera SIP Telephony Gateway Handles Paging Dial Strings

The Vocera SIP Telephony Gateway handles paging by explicitly setting timers for dial strings, and pause characters are used to control when DTMF is sent to the PBX.

The following figure shows the format of a valid paging dial string for Vocera SIP Telephony Gateway:

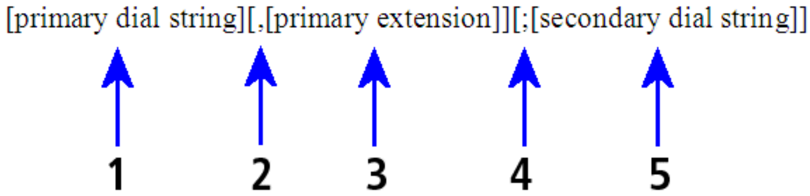


Figure 16: Paging dial string

1. **Primary Dial String** – a dialable phone number.
2. **Comma** – separates the primary dial string from the primary extension. This separating comma along with any commas in the primary extension will be added to the DTMF queue as pauses. The comma also distinguishes the primary dial string from the primary extension. The duration of a comma pause is configurable for each trunk access code; see Configuring Trunk Access Codes (TACs) on page 63.
3. **Primary Extension** – a string of DTMF digits to be sent to the PBX. It must start with at least one comma, and it cannot contain a semicolon.
4. **Semicolon** – a single semicolon for the Connect Pause. The semicolon introduces the delay before sending the secondary dial string. It also disginguishes between the primary dial string/extension and the secondary dial string. The duration of a semicolon pause is configurable for each trunk access code; see Configuring Trunk Access Codes (TACs) on page 63.
5. **Secondary Dial String** – a string of DTMF digits to be sent to the PBX. It must start with a semicolon. After that, any valid DTMF character or pause character (comma or semicolon) is valid.

Vocera SIP Telephony Gateway can accept a dial string of the form:

```
4085551234,,9898,,##;5559999,,,3232,,,5,,7,,555667777;;##*
```

Vocera SIP Telephony Gateway interprets this dial string as follows:

- Primary Dial String = `4085551234`
- Primary Extension = `9898,,##`
- Secondary Dial String = `5559999,,,3232,,,5,,7,,555667777;;##*`

The comma and semicolon serve two distinct functions in the dial string. They introduce pauses in sending DTMF, and they affect the state behavior of the call by demarking the primary dialed number, the primary extension, and the secondary dial string.

Configuring DTMF

Vocera SIP Telephony Gateway supports dual-tone multi-frequency (DTMF) signaling. You can press keys on your phone to send DTMF tones to the PBX. The DTMF signaling is transmitted in the RTP stream, but it is not encoded as part of the audio; it is included as a separate payload type. The volume, duration, and inter-digit duration of DTMF signals are all configurable. You can also configure Vocera SIP Telephony Gateway to not hang up the call until all the DTMF digits are sent.

The DTMF digits supported are 0 through 9, #, or *. For arbitrary pauses in a secondary dial string, you can add comma and semicolon to the DTMF queue. When the RTP sending thread pulls a comma or semicolon off the DTMF queue, it will stop sending DTMF for as long as the character represents. The duration of comma and semicolon pauses is configurable for each trunk access code; see Configuring Trunk Access Codes (TACs) on page 63.

The following properties let you configure handling of DTMF events:

Table 30: DTMF properties

Property	Description
VGWDTMFInterDigitInterval	Specifies the interval between DTMF digits in milliseconds. The default is 250.
VGWDTMFDuration	Specifies the duration of each DTMF digit in samples. For G.711, the sampling rate is 8000 samples/second, which means each sample is 1/8000 second (0.125 milliseconds). The default value for VGWDTMFDuration is 900 samples (112.5 milliseconds).
VGWDTMFVolume	Specifies the volume of DTMF signals in absolute dBm0 (the sign is removed). Enter a value between 36 and 0. The default is 16. If you enter a value outside the valid range, 16 is used.
VGWDTMFWaitToHangup	Whether to not disconnect the call until all DTMF digits in the queue have been sent. The default is FALSE, that is, the call is hung up without waiting for DTMF digits in the queue.

Handling DTMF in the RTP Payload

As mentioned above, Vocera SIP Telephony Gateway supports DTMF events as RTP payload as specified by RFC 2833.

If your IP PBX does not handle DTMF events as RTP payload and you do not use a media termination point (MTP) to convert out-of-band SIP Notify DTMF messages into RFC 2833 in-band (RTP), Vocera recommends that you connect to the PBX via a T1 ISDN-PRI circuit. This would connect to a media gateway (such as a Dialogic Media Gateway), and then to the Vocera SIP Telephony Gateway, as shown in the following figure.

Figure 17: VSTG connecting to PBX through Dialogic Media Gateway

To test whether your PBX handles DTMF in the RTP payload, connect a SIP softphone (such as x-lite, which you can download at http://www.counterpath.com/x-lite.html) to the PBX. To test outbound DTMF, attempt to use an outside service that uses DTMF, such as a paging service. To test inbound DTMF, use the SIP softphone to call your cell phone. Press some DTMF keys on the cell phone while you listen on the SIP softphone. If you can hear the DTMF tones, then they are NOT being converted correctly via RFC 2833.

Configuring SIP Provisional Message Reliability

To support interoperability with the PSTN such as hearing a ring back tone when you dial a number, the Vocera SIP Telephony Gateway needs to support SIP early media. This requires using the option tag "100rel", which enables reliable provisional responses. The Vocera SIP Telephony Gateway defaults to initially trying "100rel" from the following property:

```
VTGTry100Rel = true
```

If the SIP message returns a "420 Bad Extension" response, it will try the call again without requiring the "100rel" option tag.

To force calls to have early media (and a ring back tone) or otherwise fail, set the `VTGRequire100Rel` property to `TRUE`. If the SIP message returns a "420 Bad Extension" response, the call will fail.

Table 31: 100rel properties

Property	Description
VTGTry100Rel	Whether to support SIP early media by trying the "100rel" option tag with the SIP message. The default is TRUE, which means Vocera SIP Telephony Gateway intially tries "100rel". If the provisional response is not acknowledged, Vocera SIP Telephony Gateway will try the call again without requiring the "100rel" option tag.
VTGRequire100Rel	Whether to always require the "100rel" option tag to enable reliable provisional responses. The default is FALSE. When this property is set to TRUE and a provisional response is not acknowledged, the call will fail.

Configuring Ring Back Options

If your IP PBX does not support SIP early media or does not provide a ring back, you can configure Vocera SIP Telephony Gateway to use its own ring back tone by setting the following properties:

Table 32: Ring back options

Property	Description
VTGProvidedRingBackFilename	A ring back WAV file. Specify a location relative to the Vocera directory (<drive>:\vocera). The default file is `\telephony\vgw\ringback.wav`. The ring back tone will be played in a cycle until SDP negotiation completes, so a small WAV file is recommended. If you create a custom ring back file to use with Vocera SIP Telephony Gateway, the WAV file must have the following format: **Audio Format:** 16 bit Monophonic WAV PCM **Sampling Rate:** 8000 samples/second
VTGProvideRingBack	Whether to provide a ring back tone from the Vocera SIP Telephony Gateway rather than relying on the IP PBX to provide it. The default is TRUE.
VTGBufferRingBackData	Whether to buffer the ring back data to avoid constant disk access. The default is TRUE.
VTGRingBackBufferSize	The buffer size to allocate for the ring back tone. For G.711 mu-law, 1 second of audio is 8000 bytes. The default buffer size is 48000 bytes, or 6 seconds of ring back tone. The maximum buffer size is 128000 bytes, or 16 seconds of ring back tone.
VTGRingBackStartAtBeginning	Whether to always start the ring back tone at the beginning of the file. The default is TRUE. To start the ring back tone at the beginning, ring back data must be buffered (`VTGBufferRingBackData` = TRUE).

Configuring Trunk Access Codes (TACs)

To provide flexibility of paging support, you can specify global defaults that affect how Vocera SIP Telephony Gateway handles paging, or you can configure trunk access codes (or TACs) to specify how specific dial strings are processed. Each TAC is compared as a prefix to a dialed number, so you can use it to override entire classes of dialed numbers.

The TAC property values are entered into a matrix, with each value delimited by a forward slash (/) as a separator character. The first column contains the default value for each property. After the default value in each row, you can specify up to 64 TAC values, each delimited with a slash. White space characters (such as space or tab) are ignored.

When you specify more than one value for any of these properties, the order is important. If two or more TACs begin with the same sequence of characters, list them in descending order of length when you specify values for the `VTGTrunkAccessCode` property.

Vocera's parser processes a dial string from left to right, and when it finds a sequence of digits that matches a value specified for `VTGTrunkAccessCode`, it interprets that sequence as the TAC portion of the dial string. Therefore, given a dial string of `1234914087904100` and two `VTGTrunkAccessCode` property values listed in the order `12/1234`, the parser interprets the first match, `12`, as the TAC. However, when the same property values are listed in the order `1234/12`, the first match is `1234`.

The following table describes the TAC properties.

Table 33: Trunk access code properties

Property	Description
VTGTrunkAccessCode	Specifies a trunk access code (TAC) to identify a number dialed that passes through the Vocera SIP Telephony Gateway to the IP PBX to communicate with a Vocera device. The first value is "DEFAULTS", which cannot be changed. It is used to identify the first column in the matrix as default values. To add a TAC, type a forward slash ("/") followed by the access code. You can add up to 64 TACs.
VTGHangupMacro	Specifies a sequence to dial when a Vocera device ends a call initiated using the callback option in response to a VMI message or a page. The required sequence varies depending on the device. For example, nurse call systems and paging systems from different vendors require different hang-up sequences. Consult the device documentation for details. By default, this property value is not defined.
VTGPagingCommaDuration	Specifies the pause duration (in milliseconds) of each comma in the paging dial string. The default is 2000 ms, or 2 seconds.
VTGPagingSemicolonDuration	Specifies the pause duration (in milliseconds) of each semicolon in the paging dial string. The default is 3000 ms, or 3 seconds.
VTGPagingAddPrimaryExtOnSDPNegComplete	Whether to add the Primary Extension to the DTMF queue when SIP SDP offer/answer negotiation is complete. The default is TRUE.
VTGPagingAddPrimaryExtOnCONFIRMED	Whether to add the Primary Extension to the DTMF queue after the call is connected and the ACK is confirmed. The default is FALSE. The Primary Extension will be added to the DTMF queue at the first event that causes it to be added based on Vocera SIP Telephony Gateway properties. Consequently, if VTGPagingAddPrimaryExtOnSDPNegComplete is set to TRUE for a particular TAC, it takes precedence over the VTGPagingAddPrimaryExtOnCONFIRMED property if it is also set to TRUE.

Property	Description
VTGPagingAddSecondaryOnCONFIRMED	Whether to add the Secondary Dial String to the DTMF queue after the call is connected and the ACK is confirmed. The default is TRUE. The Primary Extension will be added to the DTMF queue at the first event that causes it to be added based on Vocera SIP Telephony Gateway properties. Consequently, if VTGPagingAddPrimaryExtOnSDPNegComplete is set to TRUE for a particular TAC, it takes precedence over the VTGPagingAddPrimaryExtOnCONFIRMED property if it is also set to TRUE.
VTGPagingAddSecondaryAfterPrimaryExt	Whether to add the Secondary Dial String to the DTMF queue immediately after the Primary Extension. The default is FALSE. In some cases, a call may not be explicitly answered (that is, it does not transition to CONNECTING and then CONFIRMED) until more digits are sent. If so, set the value of this property to TRUE.
VTGPagingAppendPoundToPageString	Whether to append a pound key (#) pound key to the end of the page dial string. The default is TRUE.
VTGGainVoceraToSIP	Adjusts the volume of audio sent from a Vocera device to a telephone. By increasing or decreasing this value, you increase or decrease the volume of the call in 3 dB increments. For example, a value of 3 increases the volume by 9 dB (3 * 3 = 9). Valid values range from -6 to 6, inclusive. The default is 0. The gain is removed when the call ends.
VTGGainSIPToVocera	Adjusts the volume of audio sent from a telephone to a Vocera device. By increasing or decreasing this value, you increase or decrease the volume of the call in 3 dB increments. For example, a value of 3 increases the volume by 9 dB (3 * 3 = 9). Valid values range from -6 to 6, inclusive. The default is 0. The gain is removed when the call ends.

Sample Vocera SIP Telephony Gateway Trunk Access Code Properties

Suppose Vocera devices interact with a paging system made by company PS, a blood pressure monitoring system made by company BP, and a nurse call system made by company NC. The following entries in `vgwproperties.txt` define the TACs for these systems:

```
#                                                     PS      BP     NC
#------------------------------------------------------------------------
VTGTrunkAccessCode                      = DEFAULTS / 835   / 7812 / 781
VTGHangupMacro                          =          / ##    /      / *9*
VTGPagingCommaDuration                  = 2000     /       / 1000 / 1000
VTGPagingSemicolonDuration              = 3000     /       / 2000 / 2000
VTGPagingAddPrimaryExtOnSDPNegComplete  = true     / false /      /
VTGPagingAddPrimaryExtOnCONFIRMED       = false    / true  /      /
VTGPagingAddSecondaryOnCONFIRMED        = true     / false /      /
VTGPagingAddSecondaryAfterPrimaryExt    = false    / true  /      /
VTGPagingAppendPoundToPageString        = true     / false /      /
VTGGainVoceraToSIP                      = 0        / 2     / 1    / 1
VTGGainSIPToVocera                      = 0        / 1     / 2    / 2
```

In this example, the hang-up macro ## is defined for PS, which has the TAC 835. Similarly, the hang-up macro *9* is defined for NC, which has the TAC 781. However, BP, which has the TAC 7812, does not define a hang-up macro (the value is filled with spaces). Also, the TAC for BP is listed before the TAC for NC because both TACs begin with the sequence 781 and the TAC for BP is longer than the TAC for NC. Listing the TACs in this order ensures that the Vocera parser will extract them correctly from a dial string. Different comma and semicolon duration and paging properties are specified for PS, whereas the other TACs use the default values for those properties.

Configuring Global Gain Control

You can set global properties that adjust the volume of audio for calls sent from a telephone to a Vocera device or from a Vocera device to a telephone. These global gain properties can be further offset by Trunk Access Code (TAC) configuration settings that also adjust the volume of calls.

Table 34: Global gain properties

Property	Description
VGWGlobalRxGain	Globally adjusts the volume of audio sent from a telephone to a Vocera device. By increasing or decreasing this value, you increase or decrease the volume of the call in 3 dB increments. For example, a value of 3 increases the volume by 9 dB (3 * 3 = 9). Valid values range from -6 to 6, inclusive. The default is 0. The gain value that you specify will be further offset by the VTGGainSIPToVocera property for a TAC.
VGWGlobalTxGain	Globally adjusts the volume of audio sent from a Vocera device to a telephone. By increasing or decreasing this value, you increase or decrease the volume of the call in 3 dB increments. For example, a value of 3 increases the volume by 9 dB (3 * 3 = 9). Valid values range from -6 to 6, inclusive. The default is 0. The gain value that you specify will be further offset by the VTGGainVoceraToSIP property for a TAC.

Configuring Caller Information

When calling out through Vocera SIP Telephony Gateway, caller information can be included in the From header of the INVITE message. For example, a From header might look something like:

From: Doctor Jankis <sip:4085559898@enterprise.com>;
tag=023874593485734

In this example, the display name is Doctor Jankis and the user part of the From URI (the calling party number) is 4085559898.

In the Vocera SIP Telephony Gateway, the display name and the calling party number values can be set in different ways depending on the Vocera configuration.

Table 35: CallerInfo properties

Property	Description
VTGUseDialCallerInfoInINVITE	Whether to use the caller information contained in the Dial signal from the Vocera Voice Server. The default is FALSE.
VTGFromURIUserPartInINVITE	The user part of the From URI to include in the INVITE message. If there is no Calling Party Number specified on the **Telephony > Basic Info** tab of the Administration Console, you can specify an alternative calling party number here. The default is 8005551212.

Property	Description
VTGFromHeaderDisplayNameInINVITE	The display name part of the From URI to include in the INVITE message. If the display name of the caller cannot be identified from the caller ID, this display name will be used instead. The default is "Vocera".

Caller Information in the Dial Signal from Vocera Voice Server

The Dial signal from the Vocera Voice Server can contain caller information that includes the display name and the number. If caller information is present in the Dial signal, it can be used in the INVITE message.

Caller information is included in the Dial signal from the Vocera Voice Server in the following two scenarios:

- The caller's Vocera extension (either the Desk Phone or Extension, Vocera Extension, or dynamic extension, whichever applies) is a Vocera DID number.
- The caller called into Vocera from a phone, and the system has the incoming caller ID information. In this case, the Vocera Voice Server passes through the caller ID.

Caller information includes the display name of the caller unless the display name cannot be identified from the caller ID.

Call Scenarios Involving Caller Information

In certain Vocera call scenarios, caller information might be confusing or cause unintended results. Consequently, you should weigh the frequency of these scenarios before you decide to enable dial signal caller information in the Vocera SIP Telephony Gateway.

- **Scenario 1:** Using a SIP-enabled desk phone with the extension 5818, Doctor Jankis calls the Vocera Direct Access hunt group number, and then uses the Genie to call a Vocera user. When the user answers the call on a badge, the badge displays 5818 on screen, as intended.
- **Scenario 2:** Using a SIP-enabled desk phone with the extension 5818, Doctor Jankis calls the Vocera Direct Access hunt group number, and then uses the Genie to call a Vocera user. Since the user is not logged in, the call is forwarded to the user's home phone. However, the home phone is set to not accept calls from phones whose caller ID is blocked or does not conform to standard U.S. 10-digit phone numbers. In this case, the call may not go through.
- **Scenario 3:** Using his cell phone, which blocks caller ID, Doctor Jankis calls the Vocera Direct Access hunt group number, and then uses the Genie to call a Vocera user. Since the user is not logged in, the call is forwarded to the user's home phone. However, the home phone is set to not accept calls from phones that block caller ID. In this case, the call may not go through.
- **Scenario 4:** Using his cell phone, Doctor Jankis calls the Vocera Direct Access hunt group number, and then uses the Genie to call a Vocera user. Since the user is not logged in, the call is forwarded to the user's cell phone. **Doctor Jankis's caller ID (his cell phone number) is displayed on the user's cell phone. This may not be what Doctor Jankis intended.**

If the results of any of these call scenarios are unintended and you would rather display the DID number of the Vocera trunk as the caller ID for all calls, then set the **VTGUseDialCallerInfoInINVITE** property to FALSE (the default setting).

Suppressing Dial Signal Caller Information

In some cases, you may want to NOT use caller information from the Vocera Voice Server Dial signal. To do so, set the following property:

```
VTGUseDialCallerInfoInINVITE = false
```

This property affects all calls.

If the Dial signal contains no caller information or the `VTGUseDialCallerInfoInINVITE` property is set to `false`, the Calling Party Number specified on the **Telephony > Basic Info**tab of the Administration Console is used as the user part of the From URI.

Configuring Calling Party Number Prefixes for Incoming Calls

The `vgwproperties.txt` file supports the following properties that can be used to add a prefix to the calling party number for incoming calls. These properties may be needed for PBXs that use E.164 numbering plans. The new prefixed number is the calling party number that the Vocera system uses to match for caller ID purposes for Vocera Access Anywhere.

These properties are commented out by default, but you can uncomment them and modify them. Enter the same number of values for each property. As with trunk access codes (TACs), use a forward slash (/) as a separator character. Both properties are limited to 256 characters.

Table 36: PrefixCallerID properties

Property	Description
VTGPrefixCallerIDIncomingINVITEMatch	Specify dialplan entries to match against a calling party number. Numbers match at the beginning of each specified dialplan entry, with each subsequent "x" character standing for an actual digit. You may enter dashes (which are ignored) for readability, but not parentheses. Vocera matches an incoming number with the first appropriate dialplan entry going from left to right.
VTGPrefixCallerIDIncomingINVITE	Specify the prefix for the corresponding dialplan entry.

```
VTGPrefixCallerIDIncomingINVITEMatch =  2xx-xxxx / 33x-xxxx / 408-xxx-xxxx /
                                        xxx-xxx-xxxx
VTGPrefixCallerIDIncomingINVITE      =  0 / 00 / 001 / 76
```

In the above example, the dialplan entries for the `VTGPrefixCallerIDIncomingINVITEMatch` property are shown on two lines to fit on this page, but they should appear on one line in the `vgwproperties.txt` file. In the example, the following prefixes are used:

- For 7-digit calling party numbers starting with "2", the prefix "0" is added.
- For 7-digit calling party numbers starting with "33", the prefix "00" is added.
- For 10-digit calling party numbers starting with "408", the prefix "001" is added.
- For all other 10-digit calling party numbers, the prefix "76" is added.

Detecting the Connection to the IP PBX

The Vocera SIP Telephony Gateway does not have a direct physical connection to the IP PBX (or VoIP gateway). However, some IP PBXs support using the OPTIONS message to ping the PBX to determine if the SIP trunk is up. The following parameters are for sending an OPTIONS message to the IP PBX to determine if the SIP trunk is still alive or not. If the connection goes down (perhaps due to a network problem), an email notification is sent to the Vocera administrator.

Note: If you have set up a VSTG server array and one of the servers stops responding, the Vocera Voice Server automatically redirects outbound calls to another available VSTG server for uninterrupted service. If the connection goes down (perhaps due to a network problem), an email notification is sent to the Vocera administrator.

Table 37: Connection status properties

Property	Description
VTGUseOPTIONSForKeepAlive	Whether to use the OPTIONS message to ping the IP PBX to determine if the SIP trunk is up. The default is TRUE. **Note:** You should only set this property to TRUE if your PBX supports using an OPTIONS message as a keep-alive mechanism.
VTGOPTIONSKeepAliveInterval	Specifies the time interval between pings of the IP PBX in seconds. The default is 30, but you can set it as low as 5.
VTGOPTIONSKeepAliveToUser	Specifies the number for the user part of the To header and for the Request URI. The default is "trunk_status". In most cases, you should not change this value. You can specify a null string (""). Do NOT change the default value to a real user number.
VTGUseOPTIONSKeepAliveText	Whether to include the text "keepalive" in the OPTIONS message payload. The default is TRUE, which is appropriate for most IP PBXs. **Note:** If you are connecting to a PBX using Dialogic Media Gateway, set this property to FALSE.

Configuring a Vocera SIP Telephony Gateway with Dual NICs

If your Vocera Voice Server and the PBX that Vocera SIP Telephony Gateway connects to are located on different subnets, your Vocera SIP Telephony Gateway requires dual network interface cards (NICs) to connect to both subnets. Additionally, you must configure the Vocera SIP Telephony Gateway so that it uses the correct IP address for SIP and RTP signaling. By default, the primary NIC address is used for SIP and RTP signaling.

The following figure shows a VSTG connecting to a Vocera Voice Server and an IP PBX on different VLANs.

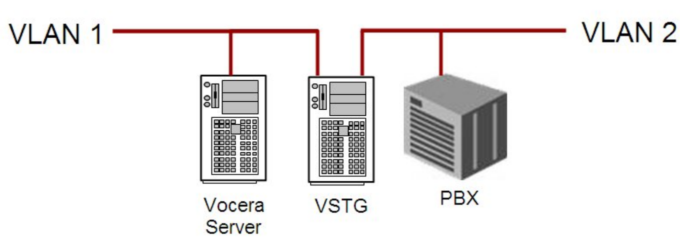


Figure 18: VSTG with dual NICs

Table 38: SIP and RTP interface properties

Property	Description
VTGSIPInterfaceAddress	If Vocera SIP Telephony Gateway should use a different IP address than the primary NIC address for SIP signaling, specify the IP address in dotted-decimal notation (for example, 192.168.15.10).
VTGRTPInterfaceAddress	If Vocera SIP Telephony Gateway should use a different IP address than the primary NIC address for RTP signaling, specify the IP address in dotted-decimal notation (for example, 192.168.15.10).

Note: By default, both the **VTGSIPInterfaceAddress** and **VTGRTPInterfaceAddress** properties are preceded by pound signs (#) in the `vgwproperties.txt` file. The pound signs denote comments, which means these properties are ignored by Vocera SIP Telephony Gateway. If you want to set these properties, make sure you remove the pound signs that precede the property names.

Overriding the Call Signaling Address to Connect to a Different IP-PBX

The **Call Signaling Address** for all Vocera SIP Telephony Gateway servers at a site is specified in the Administration Console on the **Telephony > Basic Info**page, as described in Configuring IP and SIP Settings on page 26.

You can use the **VTGRemoteCallSignalingAddr** property to override the call signaling address specified in the Administration Console, thereby letting the VSTG talk to a different IP-PBX or gateway, as shown in the following figure.

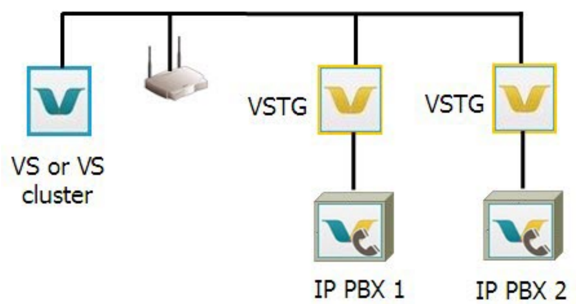


Figure 19: VSTG servers talking to different IP-PBXs

Note: If your Vocera system integrates with other internal systems that access telephony, such as an internal paging system or a nurse call system, make sure the IP-PBXs are configured consistently to support these systems.

Table 39: Remote call signaling address property

Property	Description
VTGRemoteCallSignalingAddr	Specifies the call signaling address of the IP-PBX or gateway used by this VSTG. This call signaling address overrides the one specified in the Administration Console. Enter the IP address in dotted-decimal notation (for example, 192.168.15.10). By default, port 5060 is used. If you need to change the port, enter the call signaling address in the form IP_Address:Port.

Using UDP, TCP, or TLS Transport to the IP PBX

By default, Vocera SIP Telephony Gateway uses UDP to send SIP packets to the IP PBX. You can configure Vocera SIP Telephony Gateway to use TCP or Transport Layer Security (TLS) transport instead. TLS provides encrypted communication with the IP PBX.

Table 40: SIP transport property

Property	Description
VTGSipTransport	Specifies the transport protocol used to send SIP packets to the IP PBX. Specify `udp` (the default), `tcp`, or `tls`. You can enter a comma-delimited list of values to support multiple transport types. The first value in the list denotes the transport protocol for outgoing calls. All values in the list denote supported protocols for incoming calls. Also, the calls can originate from different PBX trunks. If you specify TCP or UDP, both TCP and UDP are supported. Here's an example: `VTGSipTransport = tls, tcp` This means that all outgoing calls use TLS transport, but incoming calls can use TLS, TCP, or UDP transport. If you specify `tls` to encrypt SIP signaling between VSTG and the IP PBX using Transport Layer Security, you must install the VSTG self-signed certificate (server.crt) on your IP PBX, and you must configure your IP PBX for TLS.

Configuring TLS Transport

The SIP protocol provides no means of encryption. However, the TLS transport option lets you wrap the unencrypted SIP channel within TLS encryption to secure SIP signaling between VSTG and the PBX.

This section provides general instructions on how to configure TLS for an IP PBX. For information about specific PBXs that have been certified, contact Vocera Technical Support.

Note: TLS transport encrypts only the SIP signaling between the VSTG and the PBX. It does not encrypt the voice communication between SIP clients.

TLS Negotiation

Every VSTG generates its own self-signed certificate with its IP address and uses this certificate for TLS negotiation and encryption. The self-signed certificate for each VSTG needs to be loaded into the PBX certificate store for end-to-end TLS negotiation to succeed.

Both VSTG and the PBX act as a client or server depending on which one initiates a SIP INVITE:

- When VSTG acts as a TLS server, it binds and listens on TLS port 5061 for incoming TLS traffic. VSTG sends its self-signed certificate and the PBX accepts it by verifying that it is present in its certificate store.
- When VSTG acts as a TLS client, it makes a single TLS connection with the PBX and uses the same socket/connection for all SIP communication with the PBX. This means that TLS negotiation with the PBX happens only on the initial call for that VSTG session.

TLS Certificates

When you install VSTG, it uses OpenSSL to generate a private key and a self-signed certificate in the `\vocera\telephony\vgw` folder. The certificate (`server.crt`) is set to expire 5 years after the date it was created. Once the VSTG certificate exists, it is not updated by subsequent VSTG upgrades unless the IP address of the server has changed.

Updating TLS Certificates

If the 5-year VSTG certificate expires, you can generate a new one by running the `\vocera\telephony\certificate\cert.bat` batch file. After you generate the certificate, you must install it on your PBX.

TLS Configuration Tasks

To enable TLS transport between VSTG and the PBX, complete the following tasks:

1. Upload the TLS certificate from the following location on each Vocera SIP Telephony Gateway to the PBX:
   `\vocera\telephony\vgw\server.crt`
2. Configure TLS security for the Vocera SIP Telephony Gateway trunk on your PBX.
3. Set the following property in `vgwproperties.txt` on each Vocera SIP Telephony Gateway:
   VTGSIPTransport = tls
4. Restart each Vocera SIP Telephony Gateway.
5. Call the Vocera Hunt Group Number to make a test call. If you are able to make a call, the TLS connection with the PBX is being established successfully.

Configuring TLS for CUCM

This section provides instructions for configuring TLS for Cisco Unified Communications Manager (CUCM). For more details on CUCM configuration, see the Configuring a SIP Trunk Security Profile chapter in the CUCM Security Guide.

To upload the Vocera TLS certificate to CUCM:

1. Log into CUCM Operating System Administration.
2. Navigate to **Security > Certificate Management**.
   The Certificate List window displays.
3. Click **Upload Certificate**.
   The Upload Certificate dialog box opens.
4. In the **Certificate Name** field, select CallManager-trust.
5. Click **Browse**, navigate to the certificate file, and then click **Open**.
6. Click **Upload File**.

Note: When you upload each VSTG certificate to CUCM, the certificate is renamed to the IP address of the VSTG server and converted to PEM format.

To configure SIP trunk security on CUCM:

1. In CUCM Console, configure a SIP Trunk Security Profile for TLS.
   When you enter the **X.509 Subject Name** for the SIP Trunk Security Profile, enter the VSTG IP address. If you have a VSTG array, enter a comma-delimited list of VSTG IP addresses.
2. Apply the SIP Trunk Security Profile to the trunk in the Trunk Configuration window.

To update a VSTG certificate on CUCM:

1. Schedule VSTG and CUCM downtime.
2. Stop the VSTG server.
3. On the VSTG machine, run `\vocera\telephony\certificate\cert.bat` to generate a new certificate.
4. Log into CUCM Operating System Administration and remove the old VSTG certificate from CUCM.
5. Reset the SIP trunk.
6. Upload the new certificate to CUCM.
7. Reset the SIP trunk again.
8. Restart the CallManager Service, or restart the CUCM machine.
9. Start the VSTG server.

Enabling Multicast Support

The Vocera Smartphone with Firmware 2.2.6 or later (included with Vocera 4.1 SP6, Vocera 4.2 GA, or later) has been enhanced to support multicast transmissions for broadcast and push-to-talk conferences.

Table 41: Multicast property

Property	Description
VGWSupportMulticast	Specifies whether the gateway supports multicast transmissions. Otherwise, it converts multicast to unicast. The default is FALSE, meaning multicast support is disabled.

Configuring Auto-Answer Properties

The following properties let you control auto-answer for the gateway. You can choose to read the the auto-answer flag for the Dial signal from the Vocera Voice Server, or specify your own auto-answer value regardless what came from the Vocera Voice Server.

If either **VGWReadAutoAnswer** or **VGWForceAutoAnswer** is set to TRUE, the gateway includes an Answer-Mode header in the SIP INVITE message. **VGWForceAutoAnswer** has highest precedence. If set to TRUE, the value of Answer-Mode is set by **VGWForceAutoAnswerValue**.

Table 42: Auto-Answer properties

Property	Description
VGWReadAutoAnswer	Specifies whether the gateway reads the auto-answer flag for the Dial signal from the Vocera Voice Server. The default is TRUE.
VGWForceAutoAnswer	Forces auto-answer to a specific value regardless of what came from the Vocera Voice Server. The default is FALSE.
VGWForceAutoAnswerValue	Specifies the auto-answer value when **VGWForceAutoAnswer** is set to TRUE. The default is FALSE.

Configuring the VCS and VMP Interface

The Vocera Client Gateway provides support for SIP signaling over a TCP connection. This enables connections between the Vocera Client Gateway and Vocera Collaboration Suite clients. The Vocera Client Gateway also supports the API for the VMP Server, which enables the Vocera Client Gateway to send push notifications and escalate wakeup signals.

The following configurable parameters are included:

Table 43: VCS and VMP interface properties

Property	Description
VCGIOSClientPingInterval	The expected ping interval from a VCS iOS client. The default is 120 seconds. The Vocera Client Gateway sends a ping request to the client when this interval expires. If the client does not send a ping back, the Vocera Client Gateway sends a request again in the ping interval specified in this property.
VCGAndroidClientPingInterval	The expected ping interval from a VCS Android client. The default is 120 seconds. The Vocera Client Gateway sends a ping request to the client when this interval expires. If the client does not send a ping back, the Vocera Client Gateway sends a request again in the ping interval specified in this property.
VCGVMPConnectionPortNo	The connection port for the VMP API. The Vocera Client Gateway listens on this port, and the VMP Server opens a TCP connection to it. The default port is 5008.

Property	Description
VCGWakeupPingTimerInterval	The internal Vocera Client Gateway timer interval. This timer initiates ping pre-timeouts and escalates unanswered pings to the VMP Server, which then sends a wakeup signal using APNS or GCM. The default is 500 milliseconds.
VCGWakeupIOSPingWaitInterval	Specifies the length of time that the Vocera Client Gateway is to wait for a ping from the VMP Server after a ping request is sent to a VCS iOS client using a SIP TCP connection. The default is 1200 milliseconds. If no ping is received, the Vocera Client Gateway sends a wakeup signal to the VMP Server. If this property is set to 0, the Vocera Client Gateway sends an APNS wakeup signal right away, and does not wait for a ping request timeout.
VCGWakeupAndroidPingWaitInterval	Specifies the length of time that the Vocera Client Gateway is to wait for a ping from the VMP Server after a ping request is sent to a VCS Android client using a SIP TCP connection. The default is 1200 milliseconds. If no ping is received, the Vocera Client Gateway sends a wakeup signal to the VMP Server. If this property is set to 0, the Vocera Client Gateway sends a GCM wakeup signal right away, and does not wait for a ping request timeout.
VCGWakeupIOSCooldownInterval	The time interval during which repeated ping requests should not be sent to the same VCS iOS client. This keeps the client from being flooded with multiple ping requests. The default is 10000 milliseconds.
VCGWakeupAndroidCooldownInterval	The time interval during which repeated ping requests should not be sent to the same VCS Android client. This keeps the client from being flooded with multiple ping requests. The default is 10000 milliseconds.
VCGWakeupVerboseLogging	This option enables additional logging for troubleshooting. The default is false.

Entering Phone Numbers

Vocera allows you to enter various types of phone numbers.

For example, when you add a user to the Vocera system, you can specify the user's desk extension, cell phone number, pager number, and home phone number. Similarly, groups and address book entries also have phone numbers associated with them.

In Vocera, the value of a phone number can contain any of the following characters:

- Digits. Any of the following characters: 0123456789.
- Special dialing characters.
- Special dialing macros.
- PIN template macros.

Vocera ignores any other character that you enter in phone number fields. For example, you can enter **(408) 790-4100**, to make a number more readable, instead of **4087904100**. Vocera ignores the extra spaces, dashes, and parentheses when the number is actually dialed. However, Vocera may add access codes or area codes to numbers before dialing.

Note: Entering a number in a console field does not guarantee that a user will be able to call that number.

About Call Types

You can specify whether a group has permission to make various types of calls, and users acquire calling permissions through group membership.

To grant or revoke permissions for specific call types in the Administration Console, use the **Groups** screen > **Add/Edit** page > **Permissions** tab.

Vocera recognizes the following call types:

Table 44: Call types

Call Type	Description
Internal	A number on your side of the PBX. For example, a call to a desk extension or internal pager is an internal call. Vocera dials these numbers without adding any other codes. The Call Internal Numbers permission controls a group's ability to make internal calls.

Call Type	Description
Outside	A number on the other side of the PBX. For example, a call to a business or residence or service or outside pager is an outside call. There are two types of outside call, toll-free and toll: • By default, a call within the specified local area code is **toll-free**. Therefore, you must grant a group Call Toll-Free Numbers permission to enable members of that group to make local calls. Vocera omits or includes the area code when making a local call, depending on the value of the **Omit Area Code when Dialing Locally** field. Vocera also adds any access codes you need to get an outside line, such as a **9**. • By default, any other call is considered a **toll call**. For example, domestic or international long distance calls are toll calls. The Call Toll Numbers permission controls a group's ability to make toll calls. For domestic long distance calls, Vocera adds any access codes you need to get an outside line, such as a **9**, and any numbers you need to specify a long distance call, such as a **1** or a **0**. The format for an international long distance number depends on the locale of the Vocera Voice Server. Typically, you specify the complete dialing sequence, including access codes and country codes, as appropriate, and Vocera dials the string as-is.

You can override Vocera's default handling of internal and outside calls (for example, you can define an outside area code to be toll-free). Use the Telephony section of the Administration Console to customize this behavior.

Phone Number Rules

The following rules define how Vocera interprets a value in a phone number, pager number, or extension field in the Administration Console or the User Console.

These requirements include:

- A value that starts with the letter X (for example, X1234) represents an **internal number** (for example, a desk extension or an inside pager number).
- A value that starts with the letter Q (for example, Q901114087904100) represents a number to be interpreted **literally**. Vocera dials such numbers as-is, without adding any access codes or area codes.
- Vocera also interprets a value with 6 or fewer digits as an **internal number**. However, for clarity, it's best to type the letter X before such values to make the meaning explicit. On a Vocera system configured for the UK locale, you must type the letter Q before an outside number of 6 or fewer digits (for example, Q9100 specifies an access code and a short code service number) to make Vocera dial the number as-is.
- A value longer than the maximum length for the locale is also interpreted **literally**. The following table lists the maximum phone number length, including area code, for each supported locale.

  Table 45: Maximum phone number length per locale

AU	CA	GB	NZ	US
10 digits	10 digits	11 digits	11 digits	10 digits

- Some locales define a fixed length for telephone numbers. When a phone number field value is of this length, it represents a **local outside number**. For example, a Vocera system configured for the US locale interprets a 7-digit value as an outside number within the local area code. The following table lists the fixed length of local numbers, not including area code, defined for each supported locale.

  Table 46: Fixed length of local numbers per locale

AU	CA	GB	NZ	US
Not defined	7 digits	Not defined	7 digits	7 digits

- In any other case, the value represents an outside number. Vocera adds access codes and applies **long distance** and **toll call** rules as appropriate. For clarity, it's best to include the area code with any outside number, local as well as long distance.

Vocera uses the same rules to interpret phone numbers spoken through voice commands, with the following additional limitations:

- You cannot use special dialing characters in a voice command.
- You cannot specify an extension of seven or more digits in a voice command.
- You must include the area code when speaking an outside number.

Special Dialing Characters

A special dialing character is a non-numeric character that you can enter in a field in the Administration Console or the User Console that requires an access code, phone number, or extension.

For example, you can use an asterisk (*) to simulate pressing the star key on a touch-tone phone, or enter an X at the beginning of a number to tell Vocera to treat that number as an extension.

Vocera supports the following special dialing characters:

Table 47: Special dialing characters

Character	Effect
,	When connecting to an analog PBX, pauses for two seconds before dialing the next digit. Use a comma to force Vocera to pause briefly during a dialing sequence. Use multiple commas if you need to pause for more than two seconds. For example, suppose your system requires you to dial 9 as the local access code, but it is slow to establish an outside line. If you enter `9,` in the Default Local Access Code field, Vocera dials a 9 and then pauses to let the system establish the outside line before continuing with anything following in the dialing sequence. Do not use a comma when you are connecting to a digital PBX. The comma character is not recognized by a digital PBX, and it may prevent a connection. However, you can use commas in sequences issued after a connection is made. For example, you can use commas to the right of a semicolon.
;	Separates the data Vocera uses to connect a call from any data Vocera passes through after the call is established. Characters to the left of the semicolon are used to establish the connection, and characters to the right of the semicolon are passed through after the connection is made. For example, you may need to use a sequence of characters such as the following to forward calls to a pager: `Q 9, 1 (408) 555-1313 ; %V %D #` In this sequence, `Q 9, 1 (408) 555-1313` establishes the connection; the Q tells Vocera not to prepend an access code or area code, the 9 gets an outside line, and the remaining characters indicate the phone number to call. The %V %D # characters are pass-through values (the %V and %D are dialing macros, and the # is required by the pager to end the sequence). **Important:** For any dialing string that includes a semicolon (;), the Vocera Telephony Gateway server automatically appends a # to end the sequence.
&	Simulates pressing the flash key on a touch-tone telephone.
#	Simulates pressing the pound key (also called the hash key) on a touch-tone telephone.
*	Simulates pressing the star key on a touch-tone telephone.
X	Tells Vocera to treat the sequence of digits following this special dialing character as an extension, without prepending either an access code or an area code to them. Vocera ignores this character unless it is the first character of the number. This special dialing character is not case-sensitive.
Q	Tells Vocera to dial the sequence of digits following this special dialing character as a literal value, without prepending either an access code or an area code to them. Vocera ignores this character unless it is the first character of the number. This special dialing character is not case-sensitive.

Special Dialing Macros

A dialing macro represents a dialing sequence.

In data entry fields where you cannot enter a specific number—because the number varies with the user who accesses the feature—you can enter a dialing macro. Vocera replaces the macro with the actual number on demand.

Dialing macros are especially useful when editing Company Voicemail Access Codes and Address book entries. For example, the Company Voicemail Access Code field specifies the dialing sequence that Vocera uses to forward an incoming call to company voicemail. As part of the dialing sequence, you typically need to specify a desk phone extension to identify the voice mailbox you want to access. You cannot enter a specific desk extension in this field, because the number will vary depending on which user is forwarding calls. Instead, you use the **%D** macro as part of the dialing sequence. Vocera replaces that macro with the actual desk extension of the user who is forwarding calls.

Vocera supports the following dialing macros, listed in alphabetical order:

Table 48: Dialing macros

Macro	Effect
%C	Inserts the user's cell phone number into a data entry field. This macro expands to the value of the **Cell Phone** field of the Phone page in the Add/Edit User dialog box. A user can also enter or change this value in the User Console.
%D	Inserts the user's extension (either the **Desk Phone or Extension**, **Vocera Extension**, or dynamic extension, whichever applies) into a data entry field. You can enter or change the value of the **Desk Phone or Extension** field or the **Vocera Extension** field on the Phone page in the Add/Edit User dialog box. A user can also enter or change these values in the User Console.
%H	Inserts the user's home phone number into a data entry field. This macro expands to the value of the **Home Phone** field of the Phone page in the Add/Edit User dialog box. A user can also enter or change this value in the User Console.
%V	Inserts the Vocera hunt group or DID number into a data entry field. This macro expands to the value in the **Vocera Hunt Group Number** field on the Basic Info page of the Telephony screen.

PIN Template Macros

Each PBX has different rules for adding a PIN to a dialing sequence.

Some require the phone number followed by the PIN. Some require the PIN before the phone number. Some require an access code for an outside line, or a feature code to indicate that a number is a PIN. Some require a separator character between the PIN and the number. A telephony PIN template can use macros to specify and format the information in a PIN.

Vocera provides the following macros for specifying a PIN template:

Table 49: PIN template macros

Macro	Effect
%A	Expands to the value of the access code for the phone number being dialed.
%M	Expands to the value of the phone number being dialed.
%N	Expands to the value of the access code for the phone number being dialed, followed by the phone number. The %N macro is the equivalent of the %A macro followed by the %M macro.
%P	Expands to the value in one of the following fields, listed in descending order of precedence: • The **PIN for Long Distance Calls** field in the Phone page of the Add/Edit User dialog box. • The **PIN for Long Distance Calls** field in the Department page of the Add/Edit Group dialog box. • The **PIN for Long Distance Calls** field in the PIN page of the Telephony section.

The %A and %M macros are useful for inserting a PIN into the dialing sequence (for example, between the access code and the number) instead of appending it.

Example PIN Templates

The following table lists some example PIN templates, along with descriptions and the values sent by the Vocera system to the PBX.

The results are based on the following assumptions:

- The user belongs to a group that allows toll calls.
- The user's PIN is **1234**.
- The phone number **(213) 555-0945** is a long distance call.
- The long distance access code (if required) is **91**.
- The feature code for a PIN (if required) is ***88**.

Table 50: PIN template examples

PIN template	Result	Description
%N %P	912135550945 1234	Access code, phone number, PIN.
%M %P	2135550945 1234	Phone number, PIN.
%A, %M %P	91, 2135550945,1234	Access code, pause, phone number, PIN.
%P, %A %M	1234, 91 2135550945	PIN, pause, access code, phone number.
%A *88 %P %M	91 *88 1234 2135550945	Access code, feature code, PIN, phone number

How Vocera Builds a Dialing Sequence

When a user issues a voice command to dial a telephone number or forward a badge call to a telephone or to voice mail, Vocera sends the phone system a sequence of digits to dial.

In addition to the phone number itself, the sequence may contain the access codes needed to obtain an outside line (such as a **9**), to dial long distance (such as a **9** followed by a **1**), or to access company voicemail.

You do not enter these access codes as part of a phone number. You set up these access codes for your entire organization, and Vocera adds them to phone numbers as necessary before dialing. For example, the following figure shows the flow of events that occur when a badge user places a long distance call to a person who is listed in an address book entry.

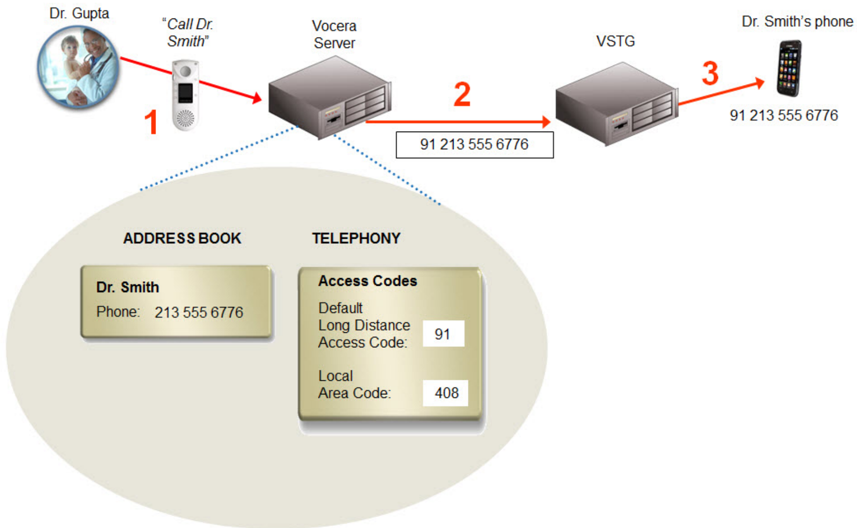


Figure 20: Placing a long distance call

In this situation, the following events occur:

1. Dr. Gupta tells the Genie to call Dr. Smith.
   The Vocera Voice Server finds Dr. Smith's telephone number in the address book, then adds long distance access codes to the dial string because Dr. Smith's area code is different from the local area code.
2. The Vocera Voice Server tells the Telephony server to dial the number.
3. The Telephony server dials the number.