

VoIP CONFIGURATION GUIDE FOR MULTI-LOCATION NETWORKS



INTRODUCTION

About this guide

This guide is designed to help you plan and configure a TalkSwitch multi-location network for Voice over IP (VoIP).

NOTE: For TalkSwitch provisioning with a VoIP service provider, please refer to the documentation located on our website www.talkswitch.com.

Who should use this guide

This guide is intended for customers who have:

- Purchased the TalkSwitch 48-CVA phone system and wish to configure it to use VoIP.
- Upgraded their existing TalkSwitch 48-CA phone system with a VoIP module (thereby converting it to a TalkSwitch 48-CVA) and wish to configure it to use VoIP.

Where to go for further information

The guides listed below, as well as other guides, can be found in the TalkSwitch folder in the Windows Start menu, on the TalkSwitch software CD, and in the support section of our website at www.talkswitch.com/support.

- For information on the initial installation and setup of a TalkSwitch phone system, refer to the *TalkSwitch Start Guide*.
- For additional information on VoIP functionality and advanced configuration, refer to the *TalkSwitch User Guide* for the TalkSwitch 48-CVA.
- For information on installing a VoIP module, refer to the *VoIP Module Installation Guide*.

Troubleshooting

The TalkSwitch system and the configuration software are designed for ease of use. However, if you encounter difficulties with the installation or configuration of your TalkSwitch:

- Consult the documents in the TalkSwitch folder on your PC or on the TalkSwitch software CD.
- Access additional TalkSwitch information online, including FAQs and Quick Guides, at www.talkswitch.com/support.

Contacting TalkSwitch Technical Support

- Contact your reseller.
- Email support@talkswitch.com, providing your company name and TalkSwitch product information.
- Call our support line at **1-866-393-9960** (toll free in continental North America) or at **(613) 725-2466**, Monday to Friday from 9:00 am to 5:00 pm EST.

BEFORE YOU BEGIN

Before you configure your TalkSwitch system to make and receive VoIP calls, review the network planning information provided in this section.

Ensure there is a TalkSwitch 48-CVA or VoIP gateway in each location

In order to create a multi-location VoIP network, a TalkSwitch 48-CVA or a VoIP gateway, such as a Mediatrix® 2102, is required at each location. The Mediatrix 2102 is a TalkSwitch-certified third-party SIP gateway that can be used by small branch offices or teleworkers.

Note: Check with TalkSwitch Support before purchasing third-party SIP devices to ensure they operate with TalkSwitch.

Each location in a multi-location network can support up to 4 TalkSwitch 48-CVA units, with a maximum of 32 local extensions and 16 VoIP lines per location. You can contact TalkSwitch Support for assistance in setting up large multi-location networks.

The following is an example of a multi-location network where two locations are equipped with TalkSwitch systems and a teleworker is equipped with a Mediatrix 2102 device. TalkSwitch and the Mediatrix 2102 route VoIP calls over the Internet, while local calls are routed over the traditional telephone network.

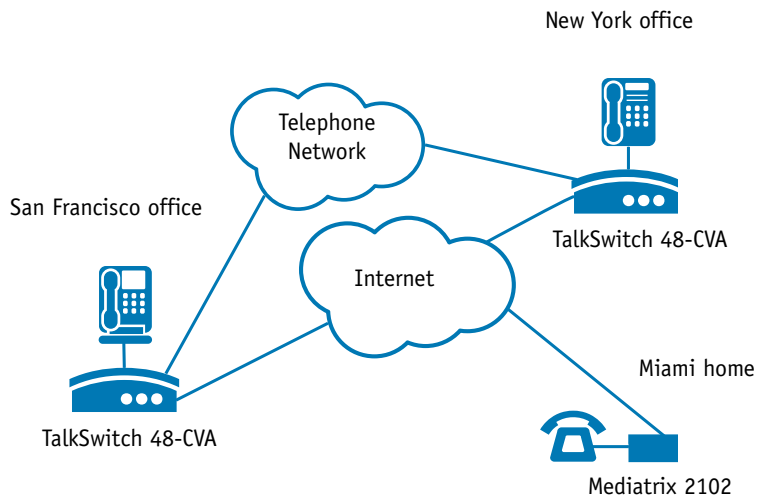


FIGURE 1. Multi-location network with VoIP calling

Ensure TalkSwitch is connected to a LAN and IP network

In order to make and receive VoIP calls, ensure the TalkSwitch system in each location is connected to a Local Area Network (LAN), consisting of a PC (for configuration purposes), Ethernet switch and router, or a router with an integrated switch (such as the Linksys BEFSR81).

Note: For optimal performance, you will need a reliable high-speed broadband Internet connection in each location. 'Lite' broadband connections are not suitable for simultaneous voice and data traffic.

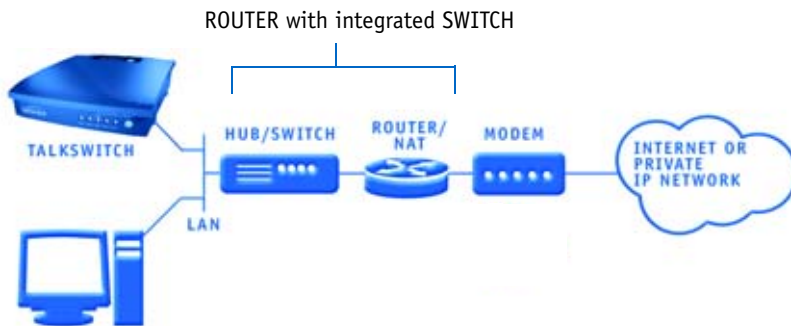


FIGURE 2. TalkSwitch network configuration

Uncertain of your VoIP call capacity? Visit our website at: www.talkswitch.com/voip/voip_capacity.php to perform a VoIP Bandwidth Test, which will determine your VoIP call capacity based on your current Internet connection speed.

Prioritizing voice traffic

If you plan on sharing your Internet connection among your computers and TalkSwitch (between data and voice), then it is critical that the voice traffic has priority over the data traffic. Some routers, such as the Linksys BEFSR81, support Quality of Service routing (QoS) for this purpose. If your router supports QoS, ensure that it is configured to provide priority to voice traffic (see your router documentation for configuration details).

Forwarding voice data through a router/firewall

In order to pass through a router's firewall, voice data received by your network must be mapped to the appropriate TalkSwitch unit. TalkSwitch uses UPnP (Universal Plug and Play) to automate the mapping of firewall ports for forwarding VoIP data. Many of today's routers support UPnP for port mapping/forwarding. If your router supports UPnP, ensure that it is enabled (see your router documentation for configuration details).

If a router does not support UPnP, you will need to configure it to map specific ports to TalkSwitch. The following describes which ports to map to TalkSwitch:

- To allow SIP Signaling data* to be forwarded to TalkSwitch, firewall port 5060 must be mapped to the TalkSwitch unit known as the Local Proxy. By default, this is the lowest numbered 48-CVA unit on a LAN. For example, if TalkSwitch unit 1 is a 48-CVA unit, then port 5060 should be mapped to this unit. SIP Signaling is done using UDP. Note: The IP Configuration window of the TalkSwitch Software indicates (with an asterix) which TalkSwitch unit is acting as the Local Proxy (see Step 1 of this guide — *Configure TalkSwitch to make VoIP calls*).
- In order for voice traffic to be forwarded to TalkSwitch, corresponding ports must be mapped to each TalkSwitch 48-CVA unit. The following table shows the firewall ports to map to each TalkSwitch 48-CVA unit on a LAN.

TABLE 1. Firewall ports^a required open for VoIP communication with TalkSwitch

Description	Type	CVA unit 1	CVA unit 2	CVA unit 3	TCVA unit 4
Voice Traffic	UDP	6000-6006	6010-6016	6020-6026	6030-6036

a. The port numbers shown are the reserved default port numbers. To re-assign these numbers, contact TalkSwitch Technical Support.

- If you plan on configuring TalkSwitch remotely, you will need to map port 9393 to the TalkSwitch unit that is being configured. For information on configuring TalkSwitch remotely, refer to the *TalkSwitch Remote Configuration* Quick Guide. Configuration is done using TCP.

For information on configuring a router for port forwarding, visit <http://www.portforward.com/routers.htm>.

TalkSwitch with DSL Internet service

If a location has DSL Internet service, ensure that the router's Point-to-Point Protocol over Ethernet (PPPoE) settings are configured for connecting to the Internet service. Refer to the documentation provided with the router and check with the Internet Service Provider (ISP) for configuration information.

* The Session Initiation Protocol (SIP) Signaling data initiates a VoIP session in an IP network.

Determine which TalkSwitch will act as the SIP server for your VoIP network

The TalkSwitch 48-CVA has a built-in Session Initiation Protocol (SIP) server to manage all SIP devices in a TalkSwitch multi-location network. The TalkSwitch system designated as the SIP server will contain the registration information for all other systems and SIP devices in the network. When making or receiving a VoIP call, a device will register with the SIP server, which will then route the call to the appropriate location.

Note: If interfacing TalkSwitch with a VoIP Service Provider, the Service Provider will act as the SIP server for your network.

In the following example, the TalkSwitch system in the San Francisco office is designated as the SIP server. When a VoIP call is placed from any location, the phone number is resolved by the SIP server and the call is routed to the appropriate office.

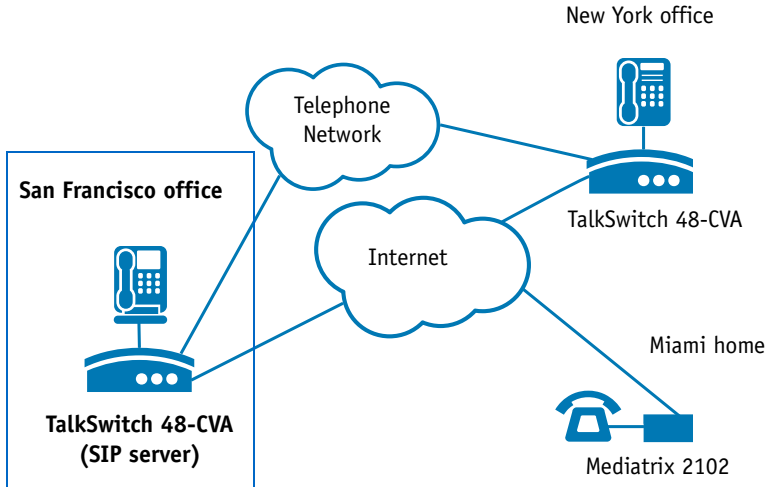


FIGURE 3. Multi-location network with designated SIP server

Which location and TalkSwitch system should I configure as the SIP server?

Any TalkSwitch 48-CVA system in a multi-location network can act as the SIP server. However, we recommend that the location have a **static**, rather than a dynamic, public IP address. Since a static IP address is fixed and does not change, a consistent connection can be kept among all locations and the SIP server.

If having a static IP address is not possible, you can obtain a Fully Qualified Domain Name (FQDN) from a Dynamic Domain Name Service (DDNS) provider. One such provider is DynDNS. A DDNS handles dynamic IP address changes and maps IP addresses to your FQDN, thereby making IP address changes transparent to TalkSwitch. Check which DDNS your router can support. For more information on DynDNS, visit:

<http://www.dyndns.org/services/dyndns>.

Note: Only one TalkSwitch system in a multi-location VoIP network should be designated as the SIP server.

Configuring TalkSwitch to act as the SIP server is described in Step 2 of this guide — *Configure TalkSwitch to act as the SIP Server*.

For more information about the SIP server and for information on optimizing your IP network for VoIP, refer to the *TalkSwitch Installation and User Guide* for the TalkSwitch 48-CVA.

Establish a dialing plan for your network

Since you are creating a private VoIP network, you must assign phone numbers to VoIP trunks (lines) in TalkSwitch. A form has been provided in this guide to help you record a dialing plan for your network (see Appendix A — *TalkSwitch VoIP Network Administration Form*).

Note: If interfacing TalkSwitch with a VoIP Service Provider, the Service Provider will assign VoIP phone numbers for your network.

Choosing phone numbers for VoIP lines

Each TalkSwitch 48-CVA unit supports 4 VoIP lines. Up to 3 phone numbers can be assigned to each of the 4 lines so that unique call handling scenarios can be configured for up to 12 numbers per unit. Phone numbers for VoIP lines can be 3 or more digits in length.

TalkSwitch supports **50 VoIP location numbers** (250 to 299) that permit direct dialing from any extension or Auto Attendant. Numbers outside of this range can also be assigned to VoIP lines and accessed using Hunt Groups. Note: Each phone number assigned to a VoIP line must be unique in a multi-location network. Assigning phone numbers to VoIP lines is described in Step 4 of the configuration section of this guide — *Assign Phone Numbers to VoIP Lines*.

In the following example, four VoIP location numbers are assigned to VoIP lines in San Francisco (251 to 254), four are assigned to VoIP lines in New York (261 to 264), and one is assigned directly from the residential office in Miami (271). These phone numbers can be dialed directly from an extension or from an Auto Attendant from any location within the network.

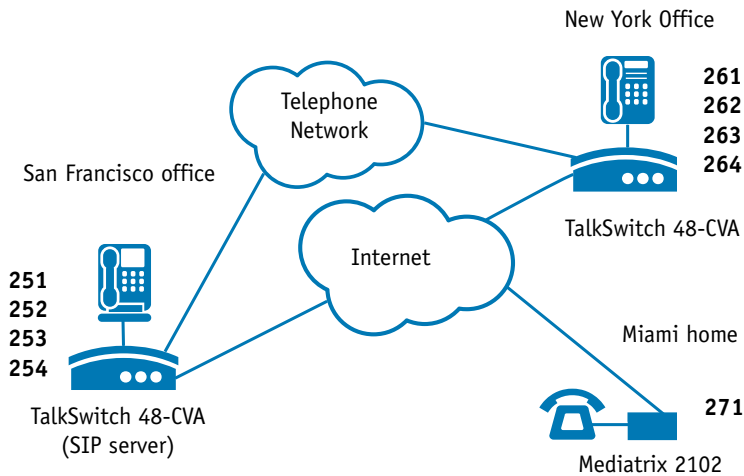


FIGURE 4. Multi-location network with assigned phone numbers

STEP 1 – CONFIGURE TALKSWITCH TO MAKE VOIP CALLS

Each TalkSwitch system in a multi-location network must be configured with both local IP and public IP address information. These addresses are used to direct VoIP calls to the appropriate location. Each TalkSwitch unit in a system will have a different local IP address.

Set the TalkSwitch local IP address

The following steps describe how to set the local IP address(es) of a TalkSwitch system:

1. Open the TalkSwitch configuration software by double-clicking the **TalkSwitch** icon on your Desktop or, from the Windows **Start** menu, select **Programs**, the **TalkSwitch** folder, and click **TalkSwitch**.
2. In the **TalkSwitch System Configuration** window, select **System Information** and then **IP Configuration** from the menu at the left.
3. TalkSwitch is factory configured to use DHCP (Dynamic Host Configuration Protocol) to automatically obtain its System IP Settings. If the location has a DHCP server, which is typically a function of the router, the fields on the screen will automatically be filled in with the correct IP information. Click **Use the following IP and DNS information** to 'lock in' these settings and ensure they are saved as the system IP settings.
4. If the location does not have a DHCP server, the System IP Settings fields will not be filled in. Click **Use the following IP and DNS information** and enter the IP address(es), Subnet mask, Default Gateway, Preferred DNS Server, and Alternate DNS Server information in the appropriate fields. For details on these settings refer to the *TalkSwitch Installation and User Guide* for the 48-CVA.

Note: **Unit 1 IP Address** designates the local IP address of TalkSwitch unit 1. If there are multiple units in a system, enter the IP addresses of all units in the system (Unit 2 IP Address, Unit 3 IP Address, Unit 4 IP Address).



Set the TalkSwitch public IP address

The public IP address of a TalkSwitch system is assigned by your Internet Service Provider. Depending on the type of service, you will either have a dynamic or static IP address. If you are unsure which type you have, contact your Internet Service Provider.

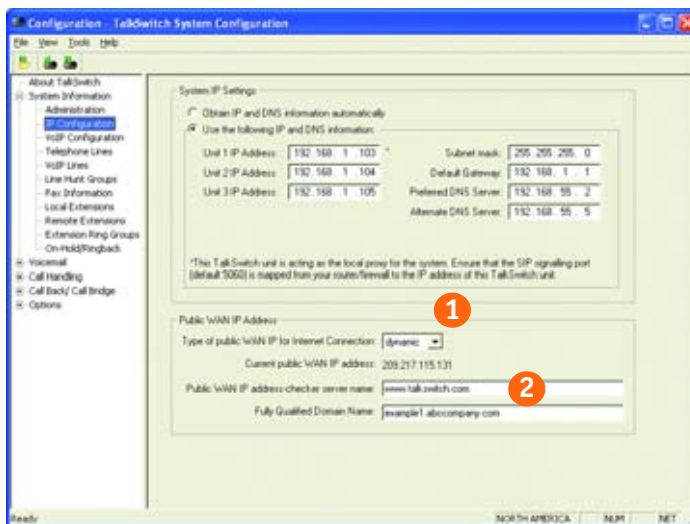
The following steps describe how to set the public IP address of a TalkSwitch system:

1. Specify the **Type of Public WAN IP for Internet Connection** by selecting either **dynamic** or **static** from the drop-down list.

If this TalkSwitch system will act as the **SIP server** for a network, we recommend having a static Public IP address. For more information on the SIP server, refer to *Determine which TalkSwitch system will act as the SIP server for your VoIP network* in the *Before you Begin* section of this guide.

2. If this TalkSwitch system will act as the SIP server for a network and has a dynamic Public IP address, enter the **Fully Qualified Domain Name (FQDN)** associated with the Public IP address. If the address is static, leave this field blank.

You can obtain a FQDN from a Dynamic Domain Name Service (DDNS) provider such as DynDNS (visit <http://www.dyndns.org/services/dyndns>). Check which DDNS your router can support.



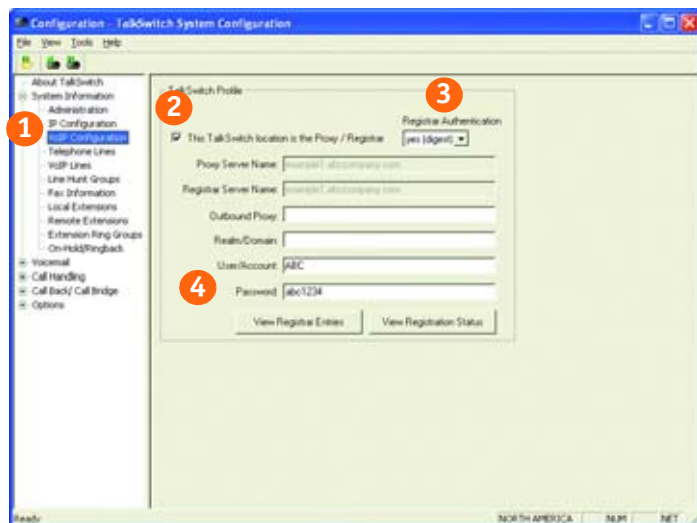
STEP 2 – CONFIGURE TALKSWITCH TO ACT AS THE SIP SERVER

The following steps describe how to configure a TalkSwitch system to act as the SIP server (Redirect Proxy/Registrar). Only one TalkSwitch system in a multi-location network can be configured as the SIP server. For information on the SIP server, see *Determine which TalkSwitch system will act as the SIP server for your VoIP network* in the *Before you Begin* section of this guide.

To configure TalkSwitch to act as the SIP server:

1. In the **TalkSwitch System Configuration** window, select **System Information** and then **VoIP Configuration** from the menu on the left.
2. Select **This TalkSwitch location is the Proxy/Registrar**. The Public IP address or Fully Qualified Domain Name will automatically be populated in the Proxy Server Name and Registrar Server Name fields.
3. To enable authentication, select **yes (digest)** from the drop-down list below **Registrar Authentication**. Using authentication will allow only authorized devices to access the network.
4. If Authentication has been enabled, enter a **User/Account** name (for example, ABC) and system **Password**.

Note: The **Outbound Proxy** and **Realm/Domain** fields are only required when the system is being provisioned with a VoIP service provider; refer to the documentation for configuring TalkSwitch with the specific service.



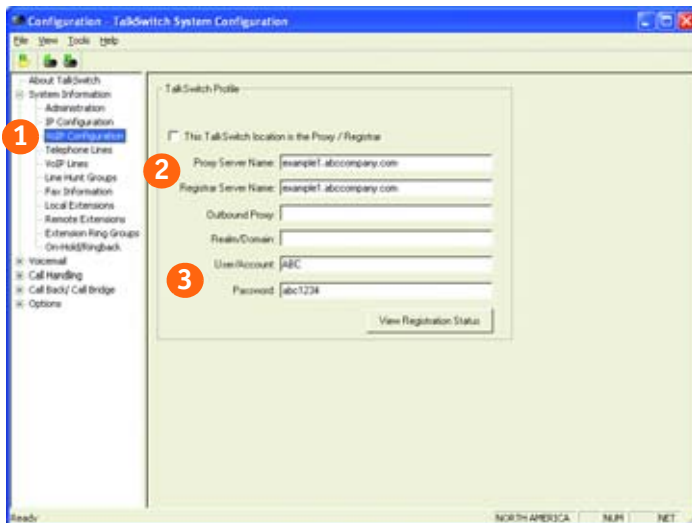
STEP 3 – CONFIGURE TALKSWITCH TO REGISTER WITH THE SIP SERVER

To make and receive VoIP calls, TalkSwitch systems must be configured to register with the system designated as the SIP server. This step describes how to configure a TalkSwitch system to register with the SIP server.

The information filled in for all systems should match the information filled in for the system designated as the SIP server in *Step 2 — Configure TalkSwitch to act as the SIP Server*.

1. In the **TalkSwitch System Configuration** window, select **System Information** and then **VoIP Configuration** from the menu on the left.
2. Enter the **Proxy Server Name** and **Registrar Server Name** in the appropriate fields. These should be the same as those specified for the SIP server.
3. If authentication was enabled for the SIP server, then enter the **User/Account** name and system **Password** in the appropriate fields. These should be the same as that specified for the SIP server.

Note: The **Outbound Proxy** and **Realm/Domain** fields are only required when the system is being provisioned with a VoIP service provider; refer to the documentation for configuring TalkSwitch with the specific service.



STEP 4 – ASSIGN PHONE NUMBERS TO VOIP LINES

The following steps describe how to assign phone numbers to VoIP lines. Phone numbers must be assigned to the VoIP lines of each TalkSwitch 48-CVA in a multi-location network. For information on VoIP phone numbers, see *Establish a dialing plan for your network* in the *Before you Begin* section of this guide.

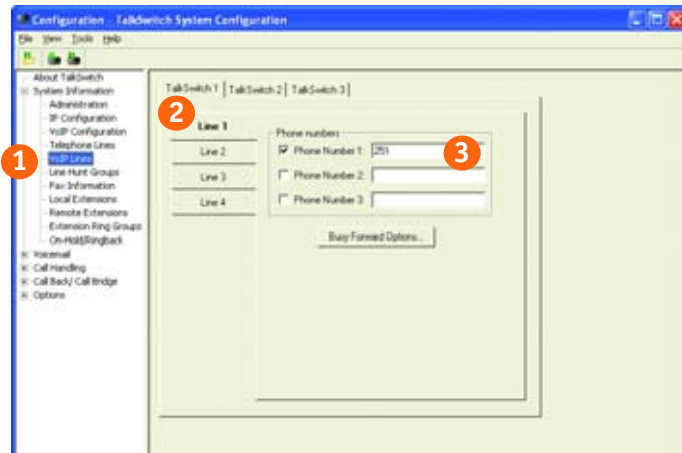
To assign phone numbers to VoIP lines:

1. In the **TalkSwitch System Configuration** window, select **System Information** and then **VoIP Lines** from the menu on the left.
2. Select **Line 1** and Click on **Phone Number 1**, a dialog box opens requesting you to enter a phone number.
3. Enter a unique phone number to be used by callers dialing this TalkSwitch location. To simplify inter-branch calling between TalkSwitch locations, we recommend using the TalkSwitch VoIP location numbers 250-299, as these numbers can be directly dialed from any extension or from an Auto Attendant.

Note: Ensure that each number assigned is unique in the network.

For each TalkSwitch 48-CVA unit in a system, you can assign up to three VoIP phone numbers per line, increasing the number of inbound VoIP call handling options available. You can set these additional numbers at any time.

4. Repeat the above steps for Lines 2, 3 and 4.
5. For systems with multiple units, repeat the process for each TalkSwitch 48-CVA unit in the system (click on the tabs labeled ‘TalkSwitch 1’, ‘TalkSwitch 2’, ‘TalkSwitch 3’, and ‘TalkSwitch 4’).



Assigning numbers outside of the VoIP location number range

Numbers outside of the VoIP location number range (outside of the 250 to 299 range) can be assigned to VoIP lines and accessed using Hunt Groups. Typically these numbers are assigned when most of the VoIP location numbers have already been allocated.

Using Hunt Groups to access VoIP lines

To use Hunt Groups to access VoIP lines, perform the following steps:

1. From the **System Information** menu, select **VoIP Lines**.
2. Assign phone numbers outside of the VoIP location number range to VoIP lines (for example, '4001' to VoIP Line 1, '4002' to line 2, '4003' to line 3, '4004' to line 4).



3. From the **System Information** menu, select **Line Hunt Groups** and assign a Hunt Group (for example '88') to access VoIP lines (select 'VoIP Lines' as the Line Type and ensure all VoIP lines are selected).

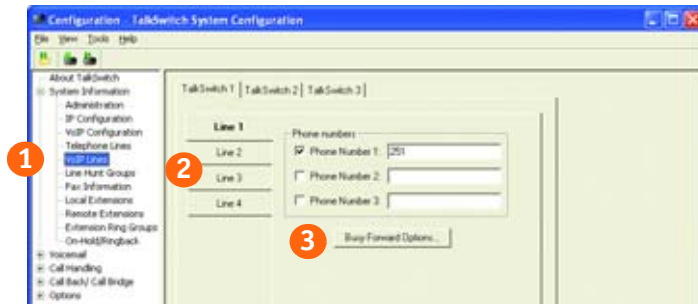


In this example, callers can dial '88' from any extension to get a VoIP dialtone. Then dial '4001' to '4004' to reach that location.

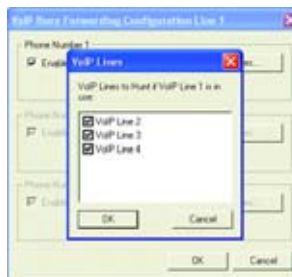
Using the Busy Forwarding feature for VoIP lines

The Busy Forwarding feature is similar to the phone company's 'Rollover/Busy Forwarding' service. To use the Busy Forwarding feature for VoIP lines:

1. From the **System Information** menu, select **VoIP Lines**.
2. Assign a VoIP location number such as '251' to VoIP Line 1. Next, assign numbers outside of the VoIP location number range to the remaining VoIP lines. For example, assign '330' to Line 2, '331' to line 3, and '332' to line 4. Ensure that a different number is used for each line.
3. Click on **Line 1** and then click on **Busy Forward Options...** to enable Busy Forwarding for VoIP Line 1.



4. Click on **Assign VoIP Lines...** and ensure that Lines 2, 3, and 4 are all selected as VoIP lines to Hunt if VoIP Line 1 is in use. Callers need only dial '251' to reach this location. If VoIP line 1 is busy, then the call will 'rollover' to line 2, 3, or 4 based on availability.

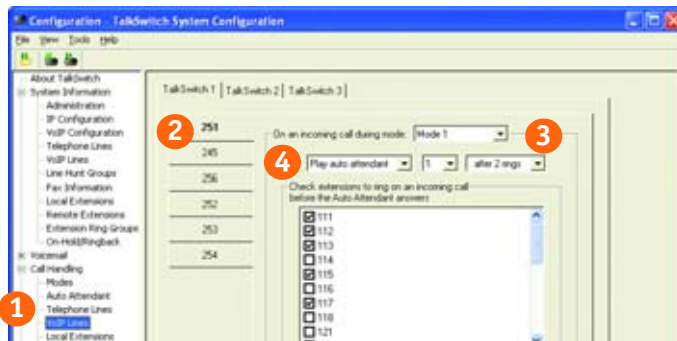


STEP 5 – CONFIGURE CALL HANDLING FOR VOIP LINES

Configuring call handling for VoIP lines customizes TalkSwitch to fit the unique needs of your business. For example, you can set all incoming VoIP calls to be answered by an Auto Attendant or sent directly to voicemail. These call handling options are similar to the options available for your regular phone lines.

Perform the following steps to configure VoIP call handling options for each TalkSwitch 48-CVA in a multi-location network.

1. In the **TalkSwitch System Configuration** window, select **Call Handling** and then **VoIP Lines** from the menu on the left.
2. Select your first VoIP line (It will be labeled with the number you assigned in *Step 4 — Assign Phone Numbers to VoIP Lines*).
3. From the drop-down list next to **On an incoming call during mode:**, select the mode to configure (for example, **Mode 1**). Additional modes can be configured later.
4. From the drop-down lists, select either: **Ring extensions only**, **Play auto attendant**, or **Go to voice mailbox**.
 - If you selected 'Ring extensions only', select the extensions from the list (by default they are all selected).
 - If you selected 'Play auto attendant', select which Auto Attendant to play (for example, **1**), the number of rings to take place before the Auto Attendant answers (for example, **after 2 rings**), and the extensions to ring before the Auto Attendant answers. If you want the Auto Attendant to answer all calls immediately without ringing extensions, select 'immediately' rather than the number of rings.
 - If you selected 'Go to voice mailbox', select the mailbox number (for example, **111**) and the number of rings to take place before the call goes to the specified voice mailbox.
5. For detailed information on these options, refer to the *TalkSwitch User Guide*.



6. Repeat this process to configure the second, third, and fourth VoIP line.
7. For systems with multiple units, repeat this process for each TalkSwitch 48-CVA unit in your system (click on the tabs labeled "TalkSwitch 1", "TalkSwitch 2", "TalkSwitch 3", and "TalkSwitch 4").

STEP 6 – SAVE SETTINGS TO TALKSWITCH

To transfer settings from your computer to the TalkSwitch system you have configured, save the settings as follows.

From the **File** menu in the **TalkSwitch System Configuration** window, select **Save to TalkSwitch...** A progress bar appears indicating that the configuration information is being sent to the TalkSwitch unit.



A backup copy of the configuration file is automatically saved to your computer when saving to TalkSwitch (this file can be found in the TalkSwitch folder that was automatically created when the TalkSwitch software was installed, usually in your Programs folder.)

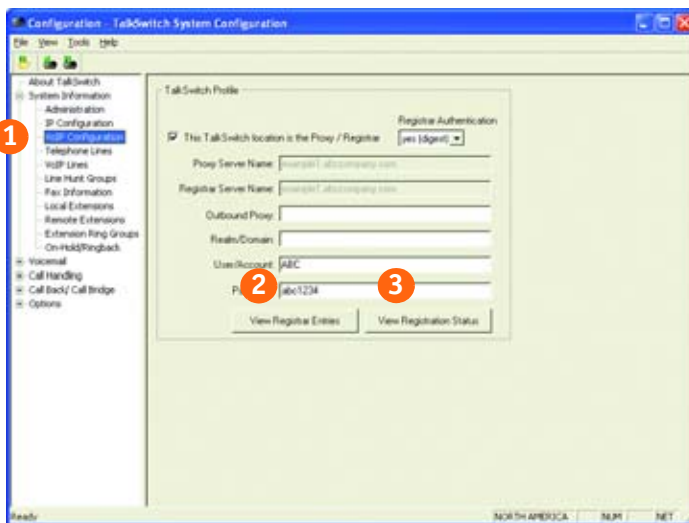
To save TalkSwitch configuration settings as a specific file on your PC, select **Save to File...** from the **File** menu. Enter the filename and location in the **Save As** dialog box.

Note: Saving to a specific file does not save the settings to TalkSwitch. You must use the **Save to TalkSwitch** option to ensure that the configuration settings are sent to the TalkSwitch system.

STEP 7 – VERIFY VoIP OPERATION

Once configured to make VoIP calls, you can verify the operation of your TalkSwitch system as follows:

1. In the **TalkSwitch System Configuration** window, select **System Information** and then **VoIP Configuration** from the menu on the left.
2. If this TalkSwitch system is the designated SIP server (Proxy/Registrar), click **View Registrar Entries** to view which systems are registered with the SIP server. A Registration Status window opens listing all TalkSwitch systems and SIP gateways registered with this system. See *Appendix B — View Registrar Entries Window Example*.
3. To confirm that this system is registered with the designated SIP server, click **View Registrations Status**. A Registration Status window opens listing all system VoIP phone numbers registered with the SIP server. See *Appendix C — View Registration Status Window Example*.



* Note: View Registrar Entries and View Registration Status functions are available beginning with Release 3.21.

APPENDIX A – TALKSWITCH VoIP NETWORK ADMINISTRATION FORM

Use the following form to keep track of phone numbers assigned to units that are part of a TalkSwitch telephone network.

SIP server location: _____

Equipment at SIP server location: _____

VoIP Phone Numbers assigned to SIP server location: _____

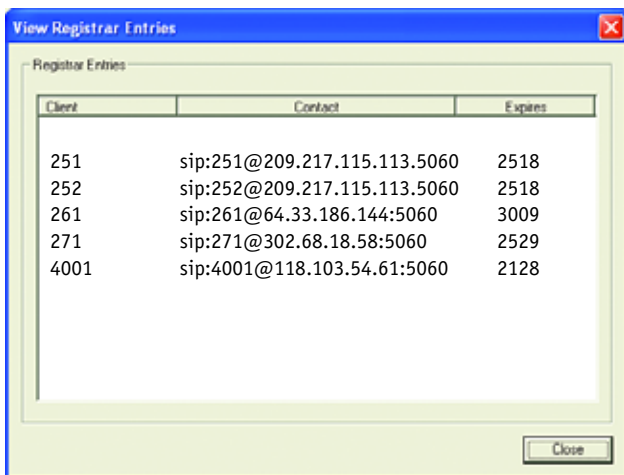
TABLE 2. Telephone Numbers assigned to units

Location Name	Equipment	VoIP Phone Number(s)

APPENDIX B – VIEW REGISTRAR ENTRIES WINDOW EXAMPLE

The following is an example of the **View Registrar Entries** window, where each entry is comprised of the following fields:

- **Client** is the phone number assigned to a VoIP line.
- **Contact** is the information used by the SIP server to contact the client. It is comprised of the phone number, public IP address, and signalling port.
- **Expires** is the amount of time, in seconds, that the client is registered with the SIP server before having to re-register.



The screenshot shows a window titled "View Registrar Entries" with a table of registration data. The table has three columns: "Client", "Contact", and "Expires". The data rows are as follows:


Client	Contact	Expires
251	sip:251@209.217.115.113:5060	2518
252	sip:252@209.217.115.113:5060	2518
261	sip:261@64.33.186.144:5060	3009
271	sip:271@302.68.18.58:5060	2529
4001	sip:4001@118.103.54.61:5060	2128

A "Close" button is located at the bottom right of the window.

APPENDIX C – VIEW REGISTRATION STATUS WINDOW EXAMPLE

The following is an example of the **View Registration Status** window, where each entry is comprised of the following fields:

- **Client** is the phone number assigned to a VoIP line.
- **Status** is the registration status: 'Registered' or 'Not Registered' to the SIP server.



The screenshot shows a window titled "Registration Status" with a standard Windows-style title bar (blue with a red close button). The window content is a table with two columns: "Client" and "Status". The table lists six entries, all with a status of "Not Registered". A "Close" button is located at the bottom right of the window.

Client	Status
245	Not Registered
251	Not Registered
252	Not Registered
253	Not Registered
254	Not Registered
256	Not Registered

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