

The 8180 requires the information below:

- SIP Domain/Proxy,
- Ring Detect Extension or/and Page Audio Extension, (The 8180 needs its own phone extension)
- Authentication ID,
- Password,

Get TalkSwitch SIP Domain/Proxy:

1) If your TalkSwitch act as a SIP Server (See page 9) then "SIP Domain/Proxy" = TalkSwitch local IP address (see page 7)

2) If your TakSwitch as to register with the TalkSwitch SIP Server (see page 10) then "SIP Domain/Proxy" = TalkSwitch SIP Server public IP address (see page 8)

Get Authentication ID and Password

1) If Authentication has been enabled you need the User/Account name and password for the TalkSwitch SIP Server (See page 9 point 5)

Get a Ring or/and Page extension (note that if you set Ring and Page on the 8180 the extension have to be different)

1) To assign a VoIP phone number see page 12.

SIP	
<input type="checkbox"/> SIP Domain/Proxy	192.168.1.202
Outbound Proxy	
STUN Server	
Registrar	
Register Period (seconds)	3600
Different Ports for Extensions	<input type="radio"/> On <input checked="" type="radio"/> Off
Keep-alive Method	<input type="radio"/> None <input checked="" type="radio"/> Double CRLF
Keep-alive Period (seconds)	30
<input type="checkbox"/> Ring Detect Extension	251
Auth ID	ABC
Password	●●●●●●
<input type="checkbox"/> Page Audio Extension	
Auth ID	
Password	

TALKSWITCH DOCUMENTATION

VoIPNETWORK CONFIGURATION GUIDE

RELEASE4.0

CT.TS005.00330 2.1A



ANSWERS WITH INTELLIGENCE[®]

INTRODUCTION

About this guide

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

Note: For TalkSwitch provisioning with a VoIP service provider, please refer to the documentation located on our website. Visit <http://global.talkswitch.com> and select country or region.

Who should use this guide

This guide is intended for customers who have:

- Purchased a TalkSwitch system equipped with VoIP lines.
- Upgraded their existing TalkSwitch phone system with a compatible VoIP module and wish to configure it to use VoIP.
- Subscribed to a VoIP service and wish to use it as part of their TalkSwitch phone system.

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. Visit <http://global.talkswitch.com> and select country or region.

- 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*.

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. Visit <http://global.talkswitch.com> and select country or region.

Contacting Technical Support

- Contact your authorized TalkSwitch reseller.
- Visit <http://global.talkswitch.com> and select country or region. Provide your company name and TalkSwitch product information.

TALKSWITCH VoIP

TalkSwitch uses the industry-standard Session Initiation Protocol (SIP) to make and manage VoIP calls. In this guide the term SIP is often used in place of VoIP.

To connect to a VoIP network or VoIP service to make and receive SIP calls, each location must be able to establish a high-speed connection to the Internet, or private network. The steps below should ensure success for typical network connection methods.

Connect to a LAN and IP network

Ensure the TalkSwitch or gateway is connected to a Local Area Network (LAN), equipped with an Ethernet switch and router, or a router with an integrated switch. A PC should be connected to the LAN, for configuring the TalkSwitch, as well as the router and/or switch.

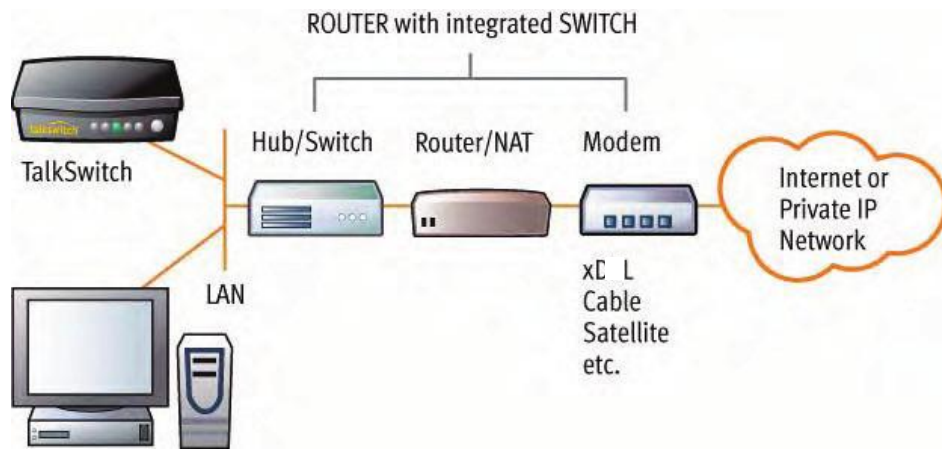


FIGURE 1. TalkSwitch network configuration

Confirm the network connection has sufficient capacity for VoIP

Ensure that the LAN in each location has a reliable high-speed broadband connection to the Internet. The quality of VoIP calls, especially on mixed voice and data networks, depends on high data-transfer rates across the network. 'Lite' broadband connections are not suitable for simultaneous voice and data traffic.

To test the VoIP call capacity based on your Internet connection speed, perform the test available on our website at: www.talkswitch.com/voip/voip_test.asp.

Ensure there is a path for voice data through your 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 automatically map firewall ports to forward VoIP data. 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 the router 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 TalkSwitch on a LAN. The *IP Configuration* window of the TalkSwitch software indicates with an asterix which TalkSwitch unit is the Local Proxy (see example on page 7). SIP Signaling uses UDP.

Example: in a 4-unit network, firewall port 5060 (UDP) would need to be mapped to the IP Address of the TalkSwitch unit ID #1.

- To allow voice traffic to be forwarded to TalkSwitch, corresponding ports must be mapped to each TalkSwitch unit. The following table shows the firewall ports to map to each TalkSwitch unit on a LAN.

Firewall ports† required open for VoIP communication with TalkSwitch

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

†Default port numbers. For instructions on re-assigning these numbers, contact TalkSwitch Technical Support.

- If you plan on configuring TalkSwitch remotely, you will need to map port 9393 (Type: TCP) to the TalkSwitch unit that is being configured. Configuration is done using TCP.

TIP: For information on configuring a router for port forwarding, visit http://www.portforward.com/english/routers/port_forwarding/routerindex.htm.

Prioritize your voice traffic

If your Internet connection is shared between computers and TalkSwitch (i.e. data and voice), it is critical that the voice traffic has priority over the data traffic. Many routers support Quality of Service (QoS) routing for this purpose. If your router supports QoS, ensure that it is enabled (see your router documentation for configuration details).

DDNS Support

If you are using Dynamic Domain Name Service (DDNS) to manage the IP address of the SIP server, ensure that your router(s) support the specific DDNS service that you are using.

If you are not setting up a VoIP network, but want to add VoIP service from a service provider to your TalkSwitch system, proceed to *Configure TalkSwitch to make and manage VoIP calls* on page 7.

SETTING UP A VoIP NETWORK

A TalkSwitch VoIP network can be set up in 4 steps:

1. Connect a TalkSwitch or SIP-compatible gateway at each location.
2. Select a TalkSwitch to use as the SIP network server.
3. Assign phone numbers to each VoIP location.
4. Configure TalkSwitch to make and manage VoIP calls.

Connect TalkSwitch or SIP-compatible gateways

In order to create a VoIP network, a VoIP-equipped TalkSwitch or a VoIP compatible gateway, such as a Mediatrix® 2102, is required at each location ¹. The instructions on page 2 describe how to do this.

Each location in a multi-location network can support up to 4 TalkSwitch units, with a maximum of 32 local extensions (analog and/or IP phones) and 16 VoIP lines² per location.

Example: The following is 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.

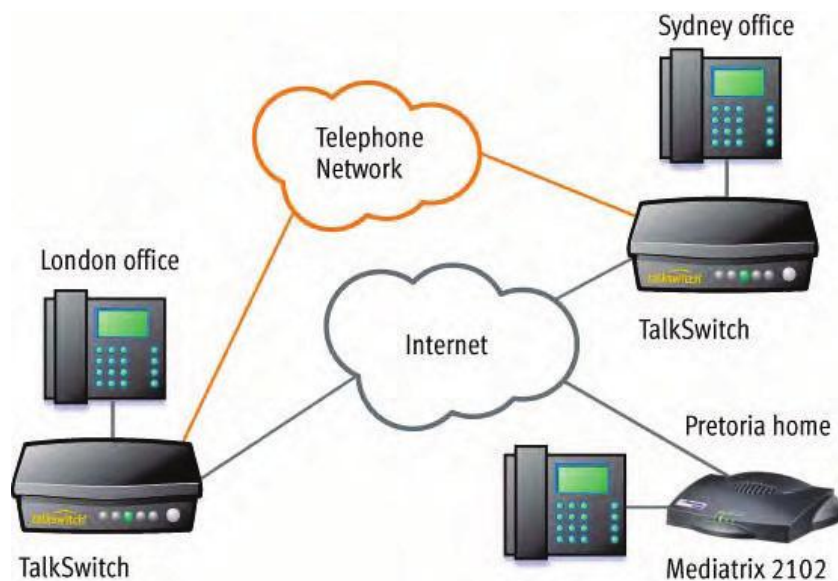


FIGURE 2. Multi-location network with VoIP calling

¹The Mediatrix 2102 is a TalkSwitch-certified third-party SIP gateway that is ideal for small branch offices or teleworkers.

²Up to 48 VoIP numbers can be assigned to the 16 VoIP lines.

Select which TalkSwitch to use as the SIP network server

A multi-location SIP network requires a SIP server to manage calls across the network. The SIP server contains registration information for all 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 destination.

TalkSwitch units have a built-in SIP server to manage a VoIP network. Though every networked TalkSwitch unit has the capability, you must designate only *one* TalkSwitch unit in a network as the SIP server.

Note: For VoIP calls made through a VoIP service provider, the service provider will act as the SIP Server.

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

Any TalkSwitch unit on the network can be designed to be the SIP server.

To simplify network operation and reliability, we recommend that you choose a location that has a *static*, rather than a *dynamic* public IP address. A static public IP address does not change, providing a consistent connection address for each location to communicate with the SIP server.

If you do not have a static IP address, you can obtain a Fully Qualified Domain Name (FQDN) from a Dynamic Domain Name Service (DDNS) provider. A DDNS matches your dynamic IP addresses to your FQDN, so that your TalkSwitch VoIP network operation is not affected by IP address changes.

One DDNS provider is DynDNS; visit: <http://www.dyndns.org/services/dns/dyndns> for more information.

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

Example: Building on the previous example, the TalkSwitch system in the London office is designated as the SIP server — see Figure 3. When a location initiates a call, the TalkSwitch in London will resolve the destination address for the call, and coordinate a connection to route the call to its destination.

Assign phone numbers to each VoIP location

You can now assign phone numbers to locations in your VoIP network.

Each TalkSwitch unit with VoIP lines can support up to 12 VoIP phone numbers. Unique call handling scenarios can be configured for each VoIP number. VoIP phone numbers 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 in the network. If needed, numbers outside of this range can also be added — see *TalkSwitch User Guide* for details. Each VoIP phone number assigned in a multi-location network must be unique (i.e. only used for one location).

For VoIP calls made through a VoIP service provider, the service provider will assign the VoIP numbers to be used for their service.

To assist in assigning VoIP numbers for your network, use *Appendix A — VoIP Network Administration Form* at the end of this guide.

Assigning VoIP phone numbers in your TalkSwitch configuration is described in STEP 4 of this guide — *Assign VoIP Phone Numbers*.

Example: Continuing with the previous example, five VoIP location numbers are assigned to VoIP lines in London (251 to 255), nine are assigned to VoIP lines in Sydney (261 to 269), and one is assigned to the teleworker office in Pretoria (271). These phone numbers can be dialed directly from an extension or from an Auto Attendant from any location within the network.

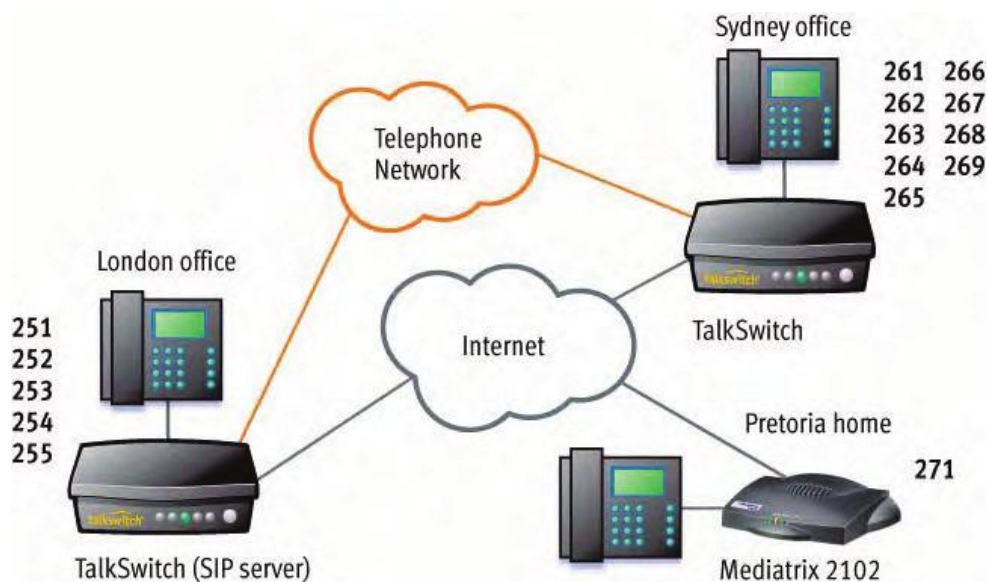


FIGURE 3. TalkSwitch network configuration

CONFIGURE TALKSWITCH TO MAKE AND MANAGE VoIP CALLS

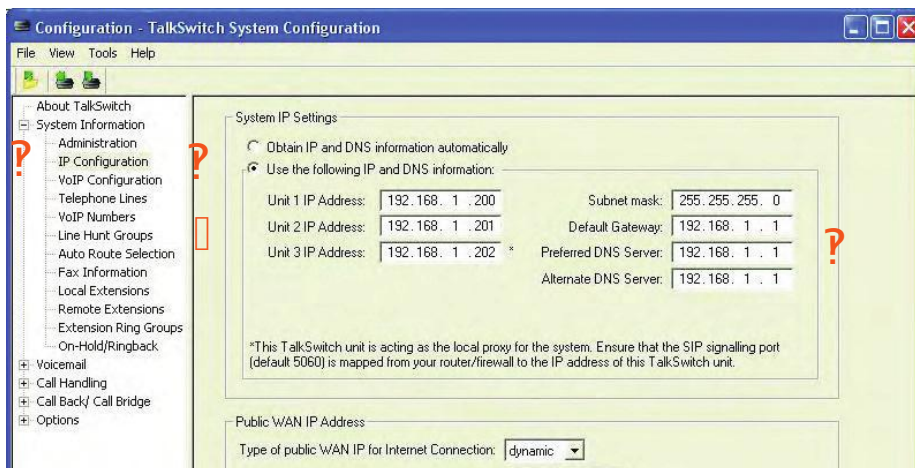
STEP 1 — CONFIGURE TALKSWITCH IP ADDRESSES

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, and share a common public 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** -> **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 User Guide*.
5. 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.

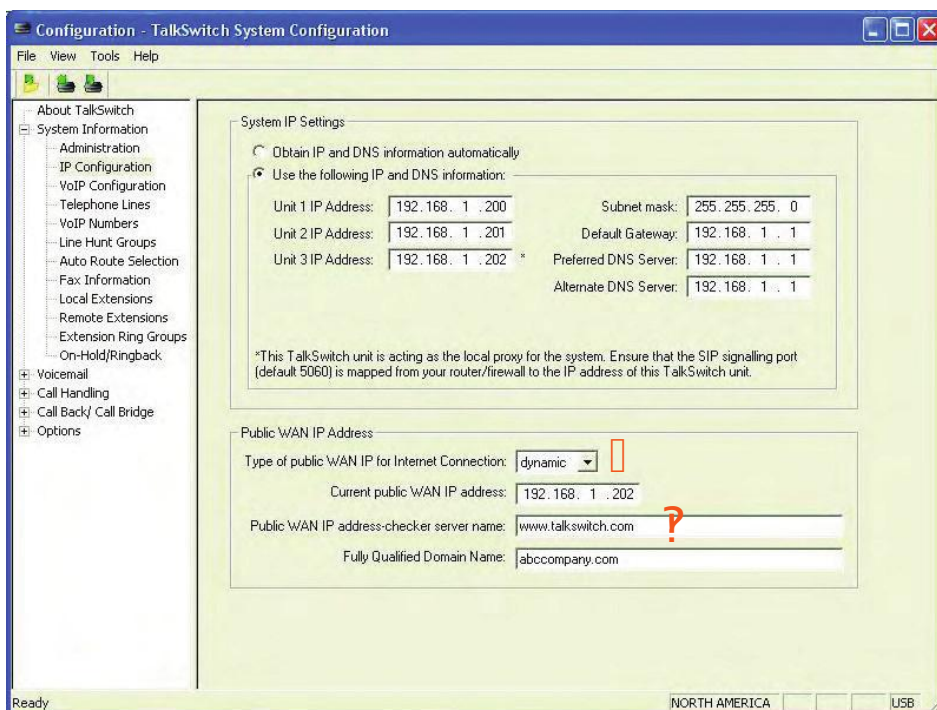
The following steps describe how to set the public IP address of a TalkSwitch system:

1. Specify the *Type of Public WAN IP for Internet Connection* by selecting either *dynamic* or *static* from the drop-down list. For private networks, please ensure the WAN IP address is the same as the Local IP address.

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 *Select which TalkSwitch to use as the SIP network server* on page 5.

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.com/services/dns/dyndns>). Check your router documentation to confirm which DDNS your router can support.



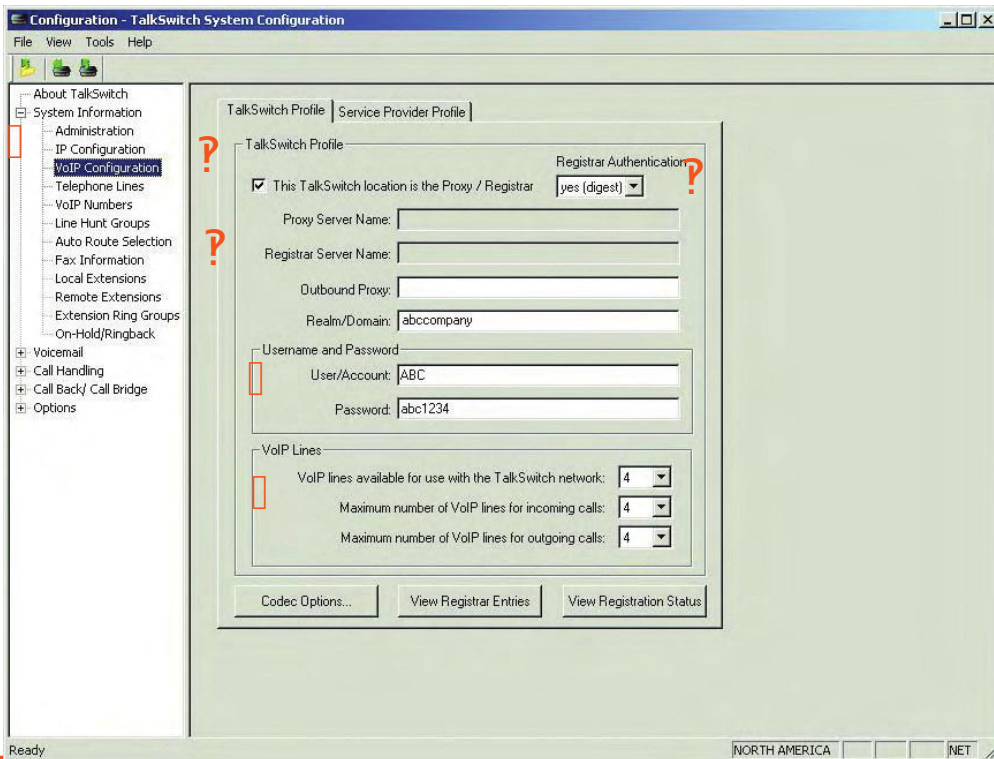
If using TalkSwitch with a VoIP Service Provider, proceed to STEP 6.

STEP 2 — CONFIGURE TALKSWITCH TO ACT AS THE SIP SERVER

A multi-location TalkSwitch system requires that one (and only one) unit be configured to act as the SIP server (Redirect Proxy/Registrar). For information on the SIP server, see *Select which TalkSwitch to use as the SIP network server* on page 5.

To configure TalkSwitch to act as the SIP server:

1. In the **TalkSwitch System Configuration** window, select **System Information -> 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. Leave the *Outbound Proxy* blank. In the *Realm/Domain* field, enter a name for your network (example: abccompany).
4. 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.
5. If Authentication has been enabled, enter a *User/Account* name (for example, ABC) and system *Password*.
6. Information on reserving VoIP lines for each TalkSwitch unit is detailed in STEP 3.

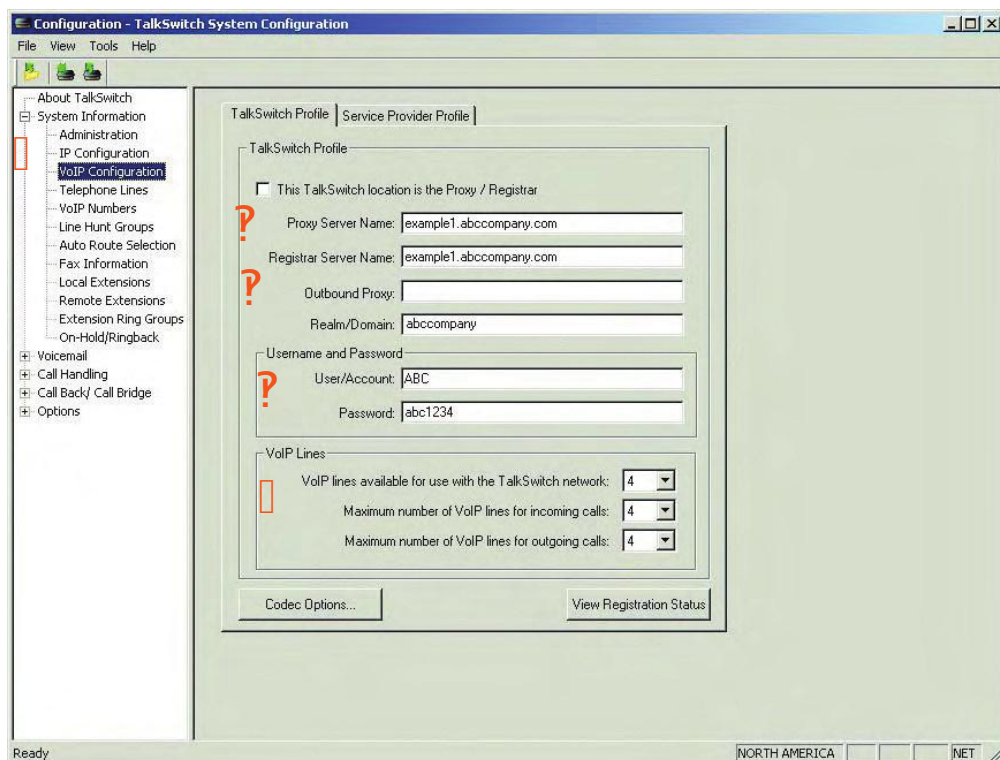


If using TalkSwitch with a VoIP Service Provider, proceed to STEP 6.

STEP 3 — CONFIGURE TALKSWITCH TO REGISTER WITH THE TALKSWITCH 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.

1. In the *TalkSwitch System Configuration* window, select **System Information -> 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. Leave the *Outbound Proxy* blank. In the *Realm/Domain* field, enter a name for your network (example: abccompany).
4. 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.
5. TalkSwitch provides options to reserve or limit the number of VoIP lines used for calls. For details, see next page.



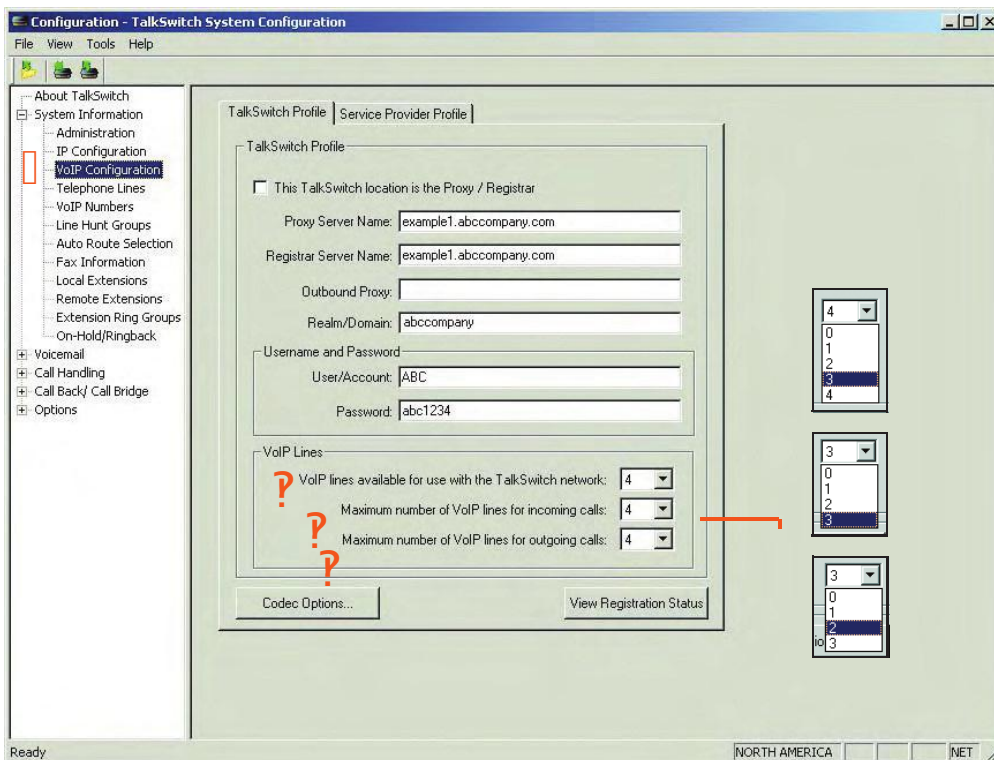
Allocating VoIP lines

TalkSwitch offers the flexibility to share VoIP lines with the TalkSwitch network and a Service Provider. One can reserve lines for use with one or the other depending on requirements.

To allocate VoIP resources, follow the following steps:

1. From the **System Information** menu, select **VoIP Configuration**, then select the profile you wish to configure (e.g. TalkSwitch Profile or Service Provider Profile).
2. In the *VoIP Lines* area, select an upper limit (if required) for lines to use for this profile. In the example below, 3 of the 4 available VoIP lines are allocated to a TalkSwitch SIP network.
3. If required, select the maximum number of VoIP lines to be used for incoming calls. In the example below, all 3 of the permitted VoIP lines are selected.
4. If required, select the maximum number of VoIP lines to be used for outgoing calls. In the example below, 2 of the 3 permitted VoIP lines are selected. Each VoIP phone number can route over any of the permitted VoIP lines.

Note: If no restrictions are selected, all VoIP lines will be permitted for incoming and outgoing calls.



If using TalkSwitch with a VoIP Service Provider, proceed to STEP 6.

STEP 4 — ASSIGN VoIP PHONE NUMBERS

The following steps describe how to assign VoIP phone numbers. For information on assigning VoIP phone numbers for a TalkSwitch VoIP network, see *Assign phone numbers to each VoIP location* on page 6.

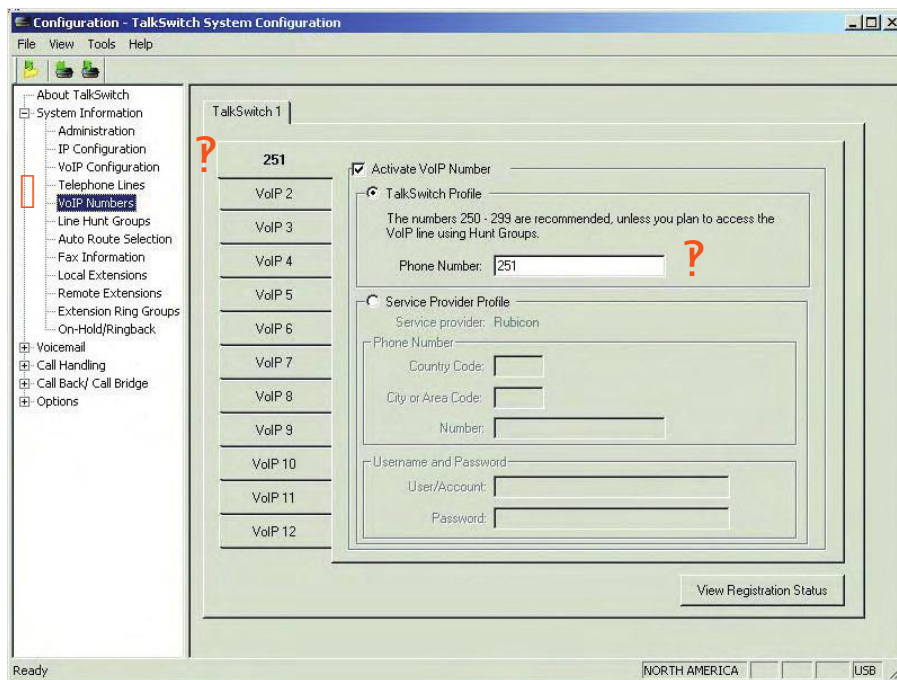
To assign VoIP phone numbers:

1. In the *TalkSwitch System Configuration* window, select **System Information -> VoIP Numbers** from the menu on the left.
2. Select **VoIP 1** and check the *Activate VoIP Number* box, the default profile for entering a phone number is the TalkSwitch Profile.
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 unit with VoIP lines on the network, you can assign up to 12 VoIP phone numbers, increasing the number of inbound VoIP call handling options available.

4. Repeat the above steps for VoIP numbers 2-12.
5. For systems with multiple units, repeat the process for each TalkSwitch unit (click on the tabs labeled *TalkSwitch 1*, *TalkSwitch 2*, *TalkSwitch 3*, and *TalkSwitch 4*).

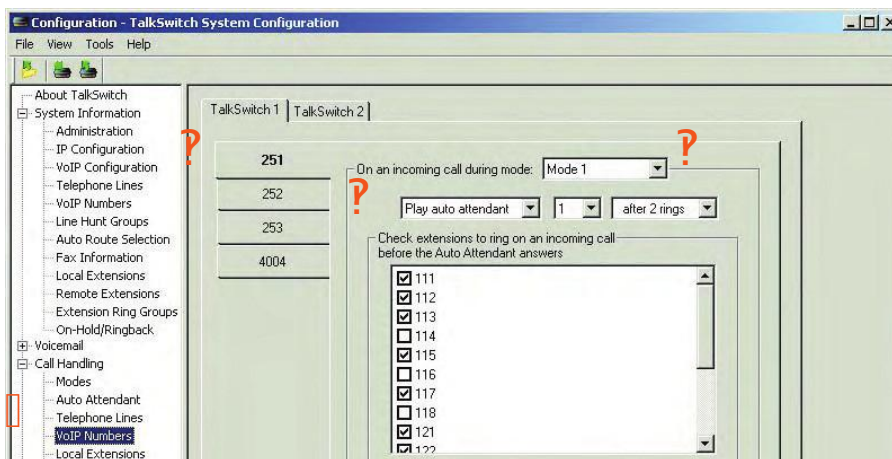


STEP 5 — CONFIGURE CALL HANDLING FOR VoIP NUMBERS

Configuring call handling for VoIP numbers customizes TalkSwitch to fit the unique needs of your business. 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 networked TalkSwitch unit with VoIP lines.

1. In the *TalkSwitch System Configuration* window, select **Call Handling -> VoIP Numbers** from the menu on the left.
2. Select your first VoIP number (It will be labeled with the number you assigned in *STEP 4 — Assign VoIP Phone Numbers*).
3. From the drop-down list next to *On an incoming call during mode:*, select the mode to configure (for example, *Mode 1*).
4. From the drop-down lists, select either: *Ring extensions only*, *Play auto attendant*, or *Go to voice mailbox*. For details, refer to the *TalkSwitch User Guide*.
 - For *Ring extensions only*, select the extensions from the list (by default they are all selected).
 - For *Play auto attendant*, select which Auto Attendant to play (for example, *I*), 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.
 - For *Go to voice mailbox*, select the mailbox number (for example, *111*) and the number of rings to occur before the call goes to the voice mailbox.
5. Repeat Steps 3 and 4 for each mode.



6. Repeat this process to configure the VoIP numbers 2-12.
7. For systems with multiple units, repeat the process for each TalkSwitch unit (click on the tabs labeled *TalkSwitch 1*, *TalkSwitch 2*, *TalkSwitch 3*, and *TalkSwitch 4*).

If you will not be using TalkSwitch with a VoIP Service Provider, proceed to STEP 8.

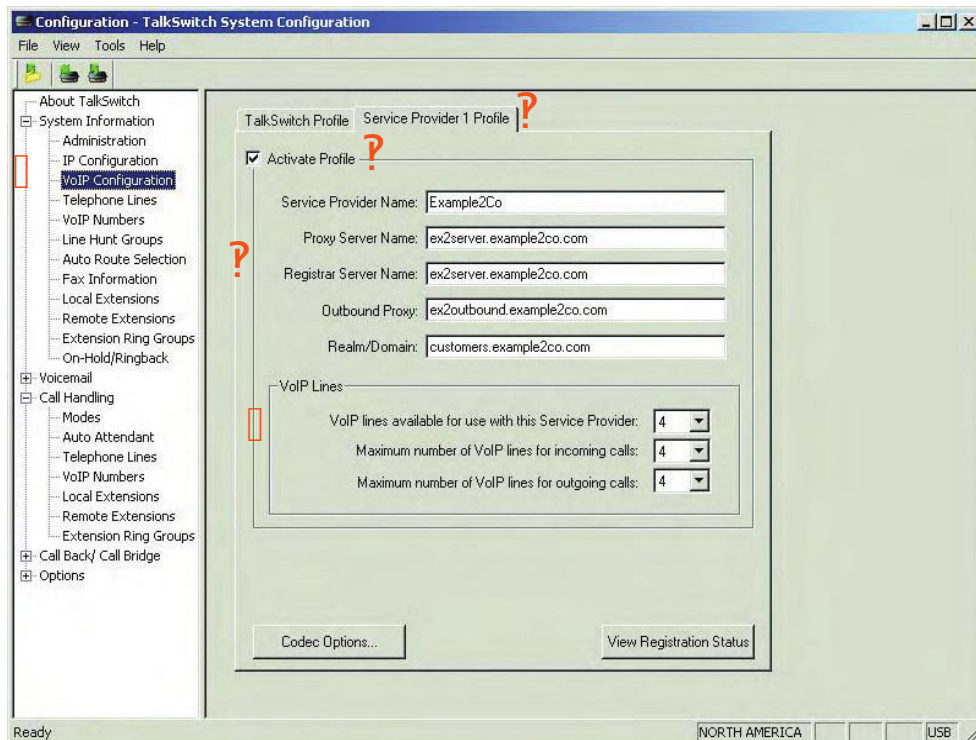
STEP 6 — CONFIGURE TALKSWITCH FOR USE WITH A SERVICE PROVIDER

Visit www.talkswitch.com/support to access the Service Configuration Guide for your VoIP Service Provider. The guide provides all necessary information to complete your configuration and should be used instead of this guide.

If there is no configuration guide for your Service Provider, follow the instructions in this guide.

To make and receive VoIP calls with a Service Provider, TalkSwitch must connect and register with the Service Provider's SIP Server(s). You need to configure TalkSwitch to do this, using information provided by the Service Provider.

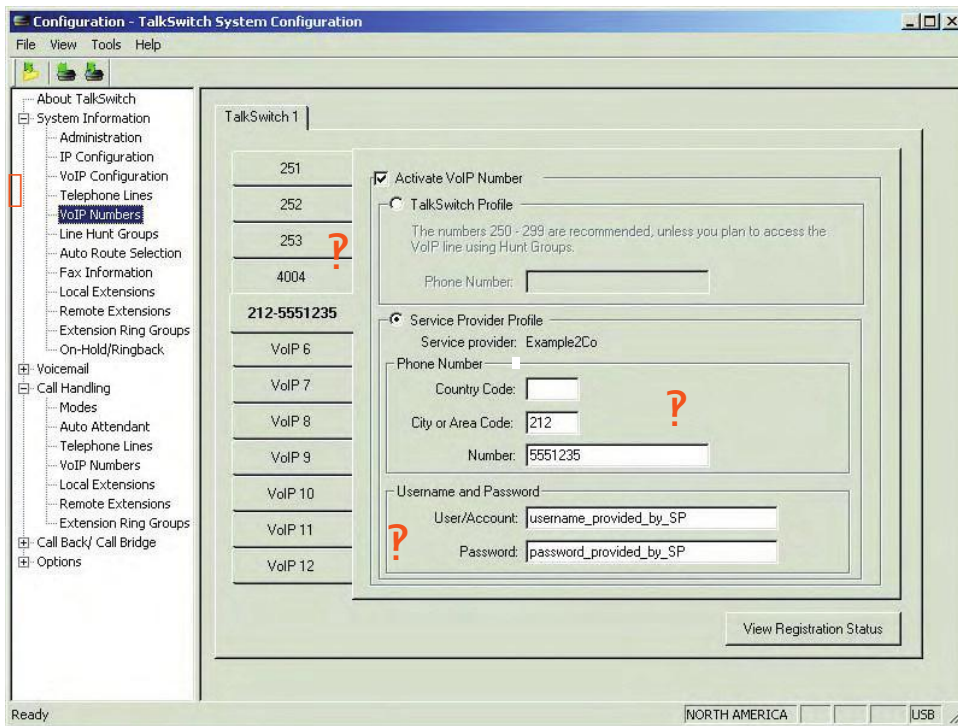
1. In the *TalkSwitch System Configuration* window, select **System Information -> VoIP Configuration** from the menu on the left.
2. Select the *Service Provider Profile* tab. Enable the *Activate Profile* checkbox.
3. Enter the name of the Service Provider.
4. Enter the **Proxy Server Name, Registrar Server Name, Outbound Proxy** and **Realm/Domain** information provided by your Service Provider, in the appropriate fields.
5. If desired, edit the number of VoIP lines reserved for use with this Service Provider. Follow the same process described in STEP 3.



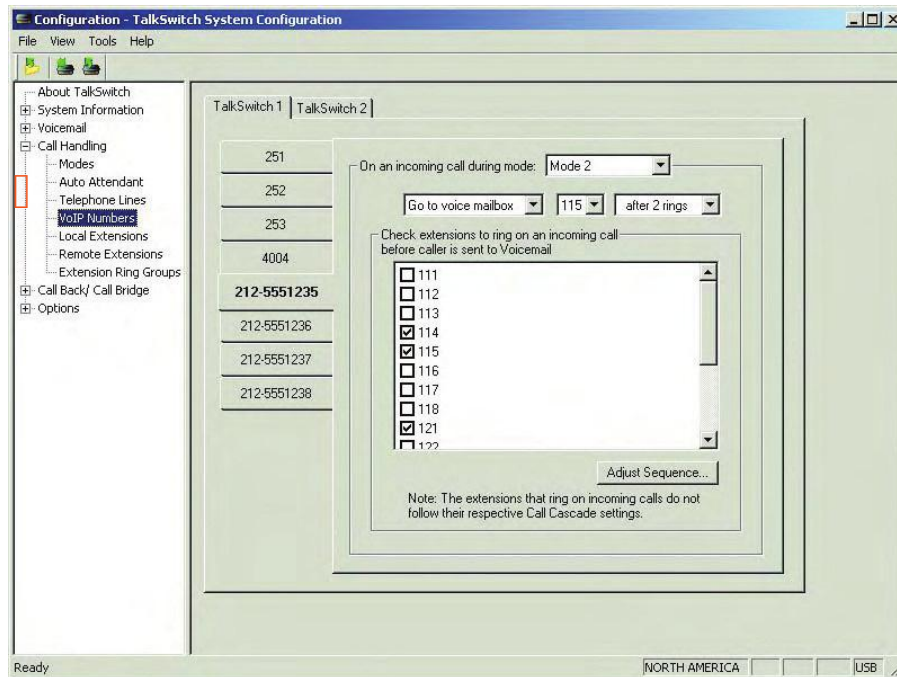
STEP 7 — ASSIGN SERVICE PROVIDER PHONE NUMBERS

1. In the *TalkSwitch System Configuration* window, select **System Information -> VoIP Numbers** from the menu on the left.
2. Select **1** and check the *Activate VoIP Number* box, the default profile for entering a phone number is the *TalkSwitch Profile*. Select *Service Provider Profile*.
3. Enter one of the phone numbers, exactly as provided by your VoIP Service Provider. If your Service Provider requires a country code, enter it, otherwise leave it blank.
4. Enter the *User/Account* information and *Password* for this particular phone number.
5. Repeat the above steps for VoIP numbers 2-12 as is required.

For systems with multiple units, repeat the process for each TalkSwitch unit (click on the tabs labeled *TalkSwitch 1*, *TalkSwitch 2*, *TalkSwitch 3*, and *TalkSwitch 4*).



6. Configure Call Handling for these VoIP numbers as you would for any of your regular phone lines. (See STEP 5 for further information.)



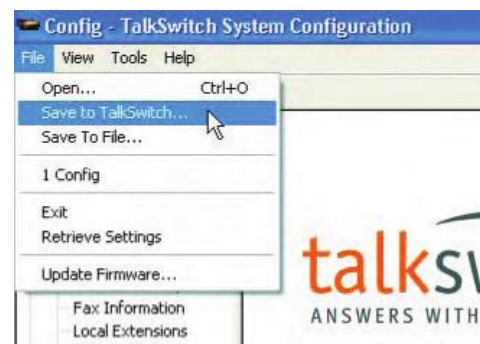
STEP 8 — SAVE SETTINGS TO TALKSWITCH

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 on your PC.

(e.g. \Program Files\TalkSwitch\TalkSwitch Configuration 4.00\LastSavedConfig4.00.tsd)



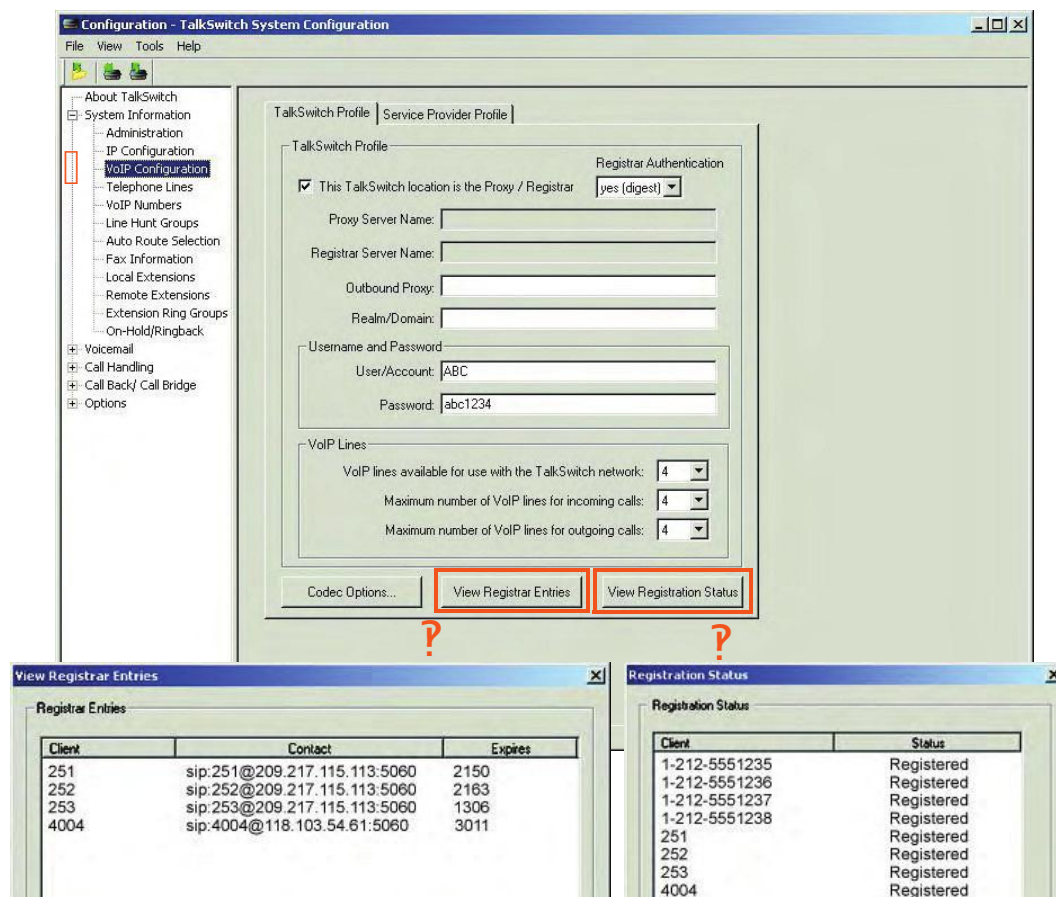
To save TalkSwitch configuration settings as a specific file on your PC, select **Save to File...** from the *File* menu. Enter the filename and location in the **Save As** dialog box.

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 9 — 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 -> 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 *View Registrar Entries* window opens listing all TalkSwitch systems and SIP gateways registered with this system.
3. To confirm that this system is registered with the designated SIP server, click **View Registration Status**. A *Registration Status* window opens listing all system VoIP phone numbers registered with the SIP server.



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.

Client is the phone number assigned to a VoIP line.

Status is the registration status: 'Registered' or 'Not Registered' to the SIP server.

APPENDIX A VoIP NETWORK ADMINISTRATION FORM

Use this form to keep track of phone numbers assigned to units that are part of a TalkSwitch VoIP network.

SIP server location: _____

Equipment at SIP server location: _____

VoIP phone numbers assigned to SIP server location: _____

TABLE 1. **Telephone numbers assigned to units**

Location name	Equipment	VoIP phone number(s)

We trust that your **TalkSwitch phone system** will provide exceptional features, performance and value to your business. Should you have any further questions, please contact your authorized TalkSwitch reseller. If you purchased directly from TalkSwitch, visit <http://global.talkswitch.com> and select country or region.

We welcome your feedback, comments and suggestions. Please e-mail us at info@talkswitch.com or write us at **TalkSwitch, 1545 Carling Avenue, Suite 510, Ottawa, ON Canada K1Z 8P9.**

Thank you for choosing TalkSwitch.

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CT.TS005.003302.IA (December 2006)
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