PureConnect®

2023 R2

Generated:

24-July-2023

Content last updated:

21-July-2023

See Change Log for summary of changes.

GENESYS™

CIC Managed IP Phones

Administrator's Guide

Abstract

CIC systems can reduce initial IP phone configuration time and ongoing maintenance with "managed IP phones". A provisioning subsystem manages all IP phone configuration in Interaction Administrator, updates manufacturer-specific firmware, and manages resetting phones as needed. This document describes all aspects of Polycom phone, Interaction SIP Station I & II, SIP Soft Phone, and AudioCodes phone implementation including configuring the network for managed IP phones, creating multiple managed IP phones and associated SIP stations, advanced configuration, boot and provision sequences, and troubleshooting.

For the latest version of this document, see the PureConnect Documentation Library at: http://help.genesys.com/pureconnect.

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CIC Provisioning Subsystem

The CIC provisioning subsystem on the CIC server manages the configuration of Polycom phones, Interaction SIP Station I and II phones, SIP Soft Phones, AudioCodes phones, and Genesys phones for the purpose of reducing initial IP phone configuration time and ongoing maintenance. The CIC provisioning subsystem manages the CIC features available on each phone, and also updates the firmware and manages resetting the phones as needed.

The Managed IP Phones container in Interaction Administrator contains all Polycom phones, Interaction SIP Station phones, SIP Soft Phones, AudioCodes phones, and Genesys phones that the CIC provisioning subsystem manages. The Managed IP Phones container also contains templates to create managed IP phones and the SIP stations associated to each, and registration groups for organizing phones according to the sources of registration data.

The CIC provisioning subsystem allows administrators to complete the following tasks in Interaction Administrator:

- Set Polycom, Interaction SIP Station I and II, SIP Soft Phone, AudioCodes, and Genesys model templates with feature sets.
- Create individual managed Polycom phones, Interaction SIP Station I and II phone, SIP Soft phones, AudioCodes phones, and Genesys phones.
- Create multiple managed IP phones for Polycom phones; Interaction SIP Station I and II phones; SIP Soft Phones; AudioCodes phones; and Genesys phones from CSV files, and migrate existing Polycom phones to managed IP phones.
- Alter feature sets on Polycom, Interaction SIP Station I and II, SIP Soft Phones, AudioCodes phones, and Genesys phones. On the next reboot, the phones pull the updated configuration files automatically.
- Schedule an *after hours* reboot for one or multiple Polycom phones, Interaction SIP Station phones I and II, SIP Soft Phones, AudioCodes phones, and Genesys phones.
- Manage and distribute appropriate firmware versions to Polycom phones, Interaction SIP Station I and II Phones, AudioCodes phones, and Genesys phones.

Supported Managed IP Phones

About Polycom Phones

Managed IP phones support Polycom phones in new installations and migrations.

Supported models and firmware

For the latest supported Polycom phone models and firmware, see Polycom Firmware for Supported and EOL Phones.

For the latest Polycom phone models and firmware that Genesys tested, see the Testlab site.

New installations

Genesys recommends that you create managed IP phones for Polycom phones using a CSV list as part of your new CIC installation to manage those phones more easily and efficiently. For more information, see Create Multiple Managed IP Phones Procedure Summary.

> **Note:**
> *Non-managed* Polycom phones are also supported in CIC 2015 R1 or later. However, you won't gain any of the advantages that managed Polycom phones offer.

About Interaction SIP Station I and II Phones

Managed IP phones support Interactive SIP Station I and II phones in new installations. The Interaction SIP Station I and II are SIP-based devices designed for the contact center and enterprise environment that use power over Ethernet with physical controls for volume, mute, on-hook/off-hook, and emergency/urgent speed autodial.

The Interaction SIP Station I and II offer a low-cost alternative to basic IP phones, soft phones with USB headsets, and high-priced high-end multimedia phone devices.

For contact center and enterprise users, Interaction SIP Station I and II with Interaction Desktop and other CIC clients offer full-featured call control.

The major differences between Interaction SIP Station I and II are:

- Interaction SIP Station I, formerly "Interaction SIP Station", has Fast Ethernet ports, an emergency speed dial button, and requires Power over Ethernet.
- Interaction SIP Station II, available in CIC 2015 R2 or later, has Gigabit Ethernet ports, a full dialpad, and the option of using a power adapter or Power over Ethernet.

Interaction SIP Station I and II work with the CIC provisioning subsystem and you configure them in Interaction Administrator in the same way as Polycom phones, SIP Soft Phones, AudioCodes phones, and Genesys phones. You *must* implement each Interaction SIP Station I and II phone as a managed IP phone.

Supported models and firmware

For the latest Interaction SIP Station I and II phone models and firmware that Genesys supports, see Interaction SIP Station I and II Firmware for Supported Phones.

For the latest Interaction SIP Station I and II phone models and firmware that Genesys tested, see the Testlab site.

Physical description

To manage incoming and outgoing calls, the Interaction SIP Station includes call control buttons and an LED status indicator.

Interaction SIP Station I

4.5" (11.43 cm) x 4.5" (11.43cm) c 1.5" (3.81cm)

Interaction SIP Station II

4.5" (11.43 cm) x 4.7" (11.93cm) c 1.5" (3.81cm)

For more information, see:

- [Interaction SIP Station and Interaction SIP Station II Product Information Sheet](#)
- *[Interaction SIP Station Quick Reference](#)*
- *[Interaction SIP Station II Quick Reference](#)*

New installations

Genesys recommends that you create managed IP phones for Interaction SIP Station I and II phones using a CSV list as part of your new CIC installation to more easily and efficiently manage those phones. For more information, see [Create Multiple Managed IP Phones Procedure Summary](#).

SIP compliancy statements

Interaction SIP Station I and II are fully compliant with the SIP communications standard and work in most global deployments, including EU countries.

The use of this equipment is subject to local rules and regulations. The following rules and regulations are relevant in some or all areas:

- Federal Communications (FCC Statement)
- CE Notice (European Union)
- WEEE EU Directive

Federal Communications (FCC statement)

This device complies with FCC Rules Part 15. Operation is subject to the following two conditions: (1) this device cannot cause harmful interference and (2) this device must accept any interference received, including interference that can cause undesirable operation.

This test equipment complies within the limit of a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits provide reasonable protection against harmful interference in a residential installation.

However, there is no guarantee that interference won't occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which you can determine by switching the equipment on and off, the user can try to correct the interference using one or more of the following measures:

1. Reorient or relocate the interference receiving antenna.
2. Increase the distance of separation between the equipment and interference receiver.
3. Connect the equipment to a power outlet on a circuit different from that to which the interference receiver connects.
4. Consult the dealer or an experienced radio/TV technician for assistance.
5. Changes or modifications that the party responsible for compliance doesn't expressly approve can void the user's authority to operate the equipment.

CE Notice (European Union)

The symbol indicates compliance of this equipment to the EMC Directive and the Low Voltage Directive of the European Union. These markings indicate that this system meets the following technical standards:

1. EN 55022 - *Limits and Methods of Measurement of Radio Interference Characteristics of Information Technology Equipment*

   > **Note:**
   > EN 55022 emissions requirements provide for two classifications:
   > - Class A is for typical commercial areas.
   > - Class B is for typical domestic areas.

2. EN 55024 - *Information technology equipment - Immunity characteristics - Limits and methods of measurement*
3. EN 61000-3-2 - *Electromagnetic compatibility (EMC) - Part 3: Limits - Section 2: Limits for harmonic current emissions (Equipment input current up to and including 16 A per phase)*
4. EN 61000-3-3 - *Electromagnetic compatibility (EMC) - Part 3: Limits - Section 3: Limitation of voltage fluctuations and flicker in low-voltage supply systems for equipment with rated current up to and including 16 A.*
5. EN 60950 - *Safety of Information Technology Equipment.*

   To determine which classification applies to your device, examine the FCC registration label on the device. If the label indicates a Class A rating, the following warning applies to your computer:

   This device is classified for use in a typical Class B domestic environment.

WEEE EU Directive

Pursuant to the WEEE EU Directive, don't dispose of electronic and electrical waste with unsorted waste. For disposal of this product, contact your local recycling authority.

SIP Station I and II phone specifications

The following table summarizes Interaction SIP Station I and II phone specifications.

Feature	Details
VoIP Signaling Protocols	SIP: RFC 3261, RFC 2327 (SDP)
Data Protocols	IPv4, TCP, UDP, ICMP, ARP, DNS 802.1p/Q for Traffic Priority and QoS ToS (Type of Service) field, indicating desired QoS DHCP Client NTP Client

Media Processing	Voice Coders: G.711, G.723.1, G.729A/B, G.722. Additionally, Interaction SIP Station II supports Opus.
	Acoustic Echo Cancelation: G.168-2004 compliant, 64-msec tail length
	Adaptive Jitter Buffer 300 ms
	Voice Activity Detection
	Comfort Noise Generation
	Packet Lost Concealment
	RTP/RTCP Packetization (RFC 3550, RFC 3551)
	DTMF Relay (RFC 2833)
Telephony Features	Speed Dial (Interaction SIP Station I) Dialpad (Interaction SIP Station II), pickup, disconnect, switchover/failover support
Configuration/Management	Automatic provisioning for firmware and configuration file upgrade
	DHCP options for automatic provisioning
Port Usage	Default port 4000 for RTP traffic, port 4001 for RTCP traffic. Depends on the value of **Media Port Start Range** in the Managed IP Phone Configuration Options in Interaction Administrator.
Power	Class 1 PoE
	Optional DC to USB power cord (Interaction SIP Station II only)
Hardware	Connectors interfaces:
	2 x RJ-45 ports (10/100BaseT Ethernet) for WAN and LAN (Gigabit support on Interaction SIP Station II)
	PoE: IEEE802.3af
	RJ-9 port (jack) for Handset
	Mounting:
	Wall mounting
	Power:
	Class 1 PoE
	Keys:
	Emergency Speed Dial (Interaction SIP Station I)
	Dialpad (Interaction SIP Station II)
	Pickup Disconnect
	Mute
	Volume Up
	Volume Down
	Multi-function status LED
	Idle
	Call alerting
	On mute
	Volume up/down

About SIP Soft Phones

PureConnect's SIP Soft Phone is a standalone application that places and controls calls, providing SIP endpoint functionality. The SIP Soft Phone application requires a USB headset to deliver audio to the user. You can use the SIP Soft Phone with Interaction Desktop and other CIC clients.

The SIP Soft Phone application works with the CIC provisioning subsystem and you configure it in Interaction Administrator in the same way as Polycom phones, Interaction SIP Station phones, AudioCodes phones, and Genesys phones. You *must* implement each SIP Soft Phone as a managed IP phone.

New installations

Create a managed IP phone for each SIP Soft Phone as part of your new CIC installation using a comma-separated values (CSV) list. For more information, see Create Multiple Managed IP Phones Procedure Summary.

About AudioCodes and Genesys Phones

Supported AudioCodes and Genesys models and firmware

For the latest AudioCodes and Genesys phones models and firmware that Genesys supports, see AudioCodes and Genesys Firmware for Supported Phones.

For the latest AudioCodes and Genesys phone models and firmware that Genesys tested, see the Testlab site.

For more information about the AudioCodes and Genesys phones, see the AudioCodes website.

New installations

Genesys recommends that you create managed IP phones for AudioCodes and Genesys phones using a comma-separated values (CSV) list as part of your new CIC installation to manage those phones more easily and efficiently. For more information, see Create Multiple Managed IP Phones Procedure Summary.

Managed IP Phone Network Provisioning

Standard procedure: Automated provisioning

CIC provisioning of managed IP phones connects the managed IP phones and downloads their configurations from the CIC server. The goal for CIC Managed IP Phone provisioning is to *automate* it as much as possible so that it doesn't need user participation.

The two factors necessary to the success of automated provisioning are:

- **Use a DHCP server (and DNS server for a Switchover pair)**

  The administrator sets the appropriate DHCP server records to configure the managed Polycom, Interaction SIP Station I and II, SIP Soft Phone, AudioCodes, and Genesys network. If the CIC system uses a Switchover pair, the administrator must also set two types of DNS records so that the phones switch to the active CIC server automatically when a Switchover event occurs. For more information, see Create DNS Host (A) Records for Switchover.

- **Provide the MAC Address (Polycom, Interaction SIP Station I and II, AudioCodes, and Genesys) or the Full Computer Name (SIP Soft Phone) for each managed IP phone**

  For an initial, new managed IP phone deployment, the administrator provides the MAC Addresses for Polycom phones, Interaction SIP Station I and II phones, AudioCodes phones, and Genesys phones, and the Full Computer Names for SIP Soft Phones in a CSV list that imports to Interaction Administrator. (For more information, see Create Multiple Managed IP Phones Procedure Summary.) When a managed IP phone contacts the provisioning subsystem to request configuration, the provisioning subsystem matches the phone's MAC address or computer name with an existing managed IP phone configuration, and it serves the configuration to the device without any additional steps.

Manual provisioning (managed IP phones)

Manual provisioning of some or all managed IP phones are necessary in certain circumstances, for example:

- You didn't specify the MAC address or full computer name in Interaction Administrator configuration for some or all Polycom, Interaction SIP Station I and II, and/or SIP Soft Phones, AudioCodes phones, or Genesys phones.

  > **Note:**
  > In an implementation of new Polycom, Interaction SIP Station I and II, SIP Soft Phones, AudioCodes phones, and/or Genesys phones, the MAC address or full computer name may not be known at the same time the comma-separated values (CSV) list is created.

- The CIC system has no DHCP server or it cannot access the DHCP server.
- Some Polycom phones, Interaction SIP Station I and II phones, SIP Soft Phones, AudioCodes phones, and/or Genesys phones are in remote locations.

Typically, an administrator with required permissions does the manual provisioning through the phone's local user interface. In some cases (for example, remote locations), users must do the manual provisioning. You do manual provisioning on each managed phone's configuration.

For more information, see Complete the Provisioning Process (Multiple Managed IP Phones).

Managed IP Phone Creation Methods

Create individual managed IP phones

Genesys recommends that you create one or more managed IP phones individually in the Interaction Administrator **Managed IP Phones** container under the following circumstances only:

- **For test purposes**

  For more information, see [Create Individual Managed IP Phones for Test Purposes](#). Walk through the creation of an individual managed IP phone so that you can explore the **Managed IP Phone** container and learn about managed IP phone configuration before deploying multiple managed IP phones in new or upgrade installations.

- **Post-implementation**

  Create managed IP phones individually after the initial managed IP phone implementation as you add more managed IP phones to the network.

Use Managed IP Phone Assistant to create multiple managed IP phones

The standard procedure for the initial managed IP phone implementation for a CIC installation is to create new managed IP phones (Polycom, Interaction SIP Station I and II, SIP Soft Phone, AudioCodes, and Genesys) using **Managed IP Phone Assistant** in Interaction Administrator. Managed IP Phone Assistant is a wizard that simplifies the creation of multiple managed IP phones and associated SIP stations.

To use the Managed IP Phone Assistant Import option to create multiple managed IP phones and associated SIP stations

1. Create one or more **managed IP phone templates**.
2. Create a **CSV Managed IP Phone List** based on 1) Template, or 2) Type, Manufacturer, and Model.
3. Run the **Managed IP Phone Assistant**, choosing the **Import** option to **import the CSV Managed IP Phone List**.

For more information, see [Create Multiple Managed IP Phones Procedure Summary](#).

Managed IP Phones and SIP Security

Polycom, SIP Soft Phone, Interaction SIP Station I and II, AudioCodes phones, and Genesys phones support SIP line security (TLS/SRTP) for SIP station-to-station calls and managed IP phone registration groups. TLS/STRP support for AudioCodes phones, and Genesys phones is available in CIC 2016 R4 and later. TLS/STRP support for Interaction SIP Station I and II phones is not available.

For more information about SIP security features and configuration, see:

- *PureConnect Security Features Technical Reference* (requires logon credentials)
- *Interaction Administrator Help*

Basic Managed IP Phones Configuration

IP Phone Network Requirements

CIC Managed IP Phones support the following network protocols:

- **Domain Name System (DNS)** protocol to specify a DNS domain for the voice VLAN, and for Switchover server pairs to resolve DNS names for the two servers so that phones switch to the active CIC server automatically when a Switchover event occurs.
- **Dynamic Host Configuration Protocol (DHCP)** to reduce system administration workload, allowing the addition of voice VLAN devices to the network with little or no manual configuration.

IP Phone Network Architecture

The CIC network architecture is built around the concept of VLANs. A VLAN (Virtual-LAN) is a network within a network which you access by tagging network traffic for that VLAN. Any network can have many VLANs and can allow for routing traffic between the VLANs as needed. VLANs serve as a way to isolate certain devices from the general network (typically called the data VLAN). They also serve to isolate the traffic from those devices from general view, whether by devices not on that specific VLAN or by malicious parties attempting to penetrate the network.

It is important to isolate all the voice traffic in a CIC system onto a separate VLAN, typically called the voice VLAN. This means that all devices that generate/pass on voice traffic need to exist on the voice VLAN. The devices includes the CIC Servers, phones, Media Servers, Gateways, and Interaction SIP Proxies.

Accessing CIC Servers, Media Servers, and Proxies from the data VLAN allows for easy remote access. Giving servers access to both VLANs complicates configuration, though, as they must have multiple NICs (Network Interface Controllers) and proper DNS/DHCP configuration on both VLANs. The configuration can become more complicated when including remote sites.

The following diagrams show the ideal configuration for the network over multiple offices, but your actual configuration can differ. For instance, although the Managed Proxy and Media Server for a remote office are shown as two distinct servers, in some cases it is appropriate to have them share the same physical device.

The following diagram shows a wide view of a typical network configuration for managed IP phones.
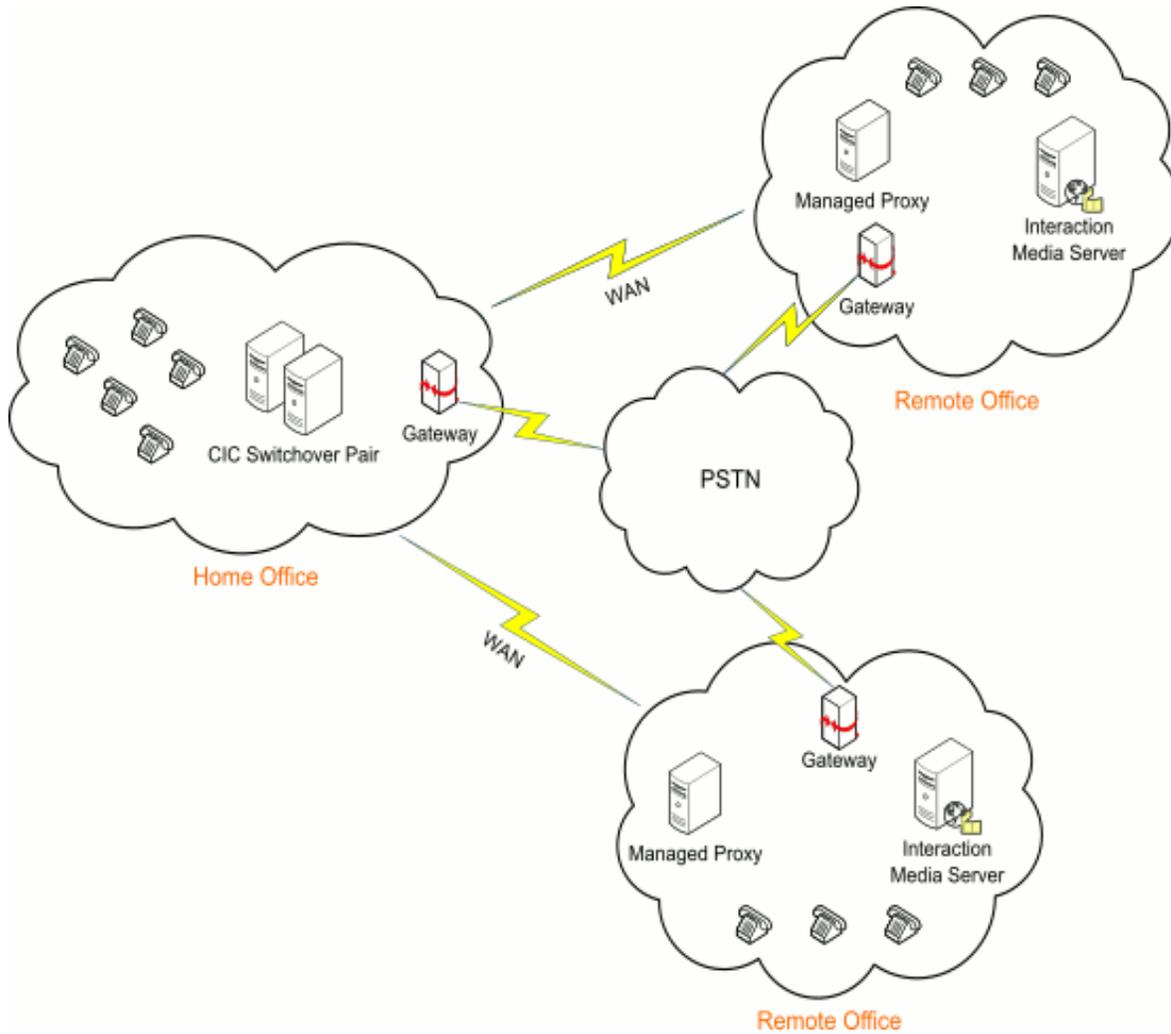
Each site, including the home office, needs access to the voice VLAN over the WAN, and each site needs its own DHCP and DNS servers. An exception is very small offices that *can* use the DNS and DHCP servers of the home office over the WAN. However, it increases WAN traffic so, if possible, you should avoid. Also, any sites that have Interaction SIP Station I and II deployed must have a DHCP server on both the voice VLAN and the data VLAN.

The following diagrams show basic home office and remote office network architecture.

Data VLAN

PUBLIC INTERNET

Switch
(no routing
between VLANs)

Interaction
Media Server

Managed Proxy

DHCP / DNS Server

Gateway

PSTN

Voice VLAN

WAN
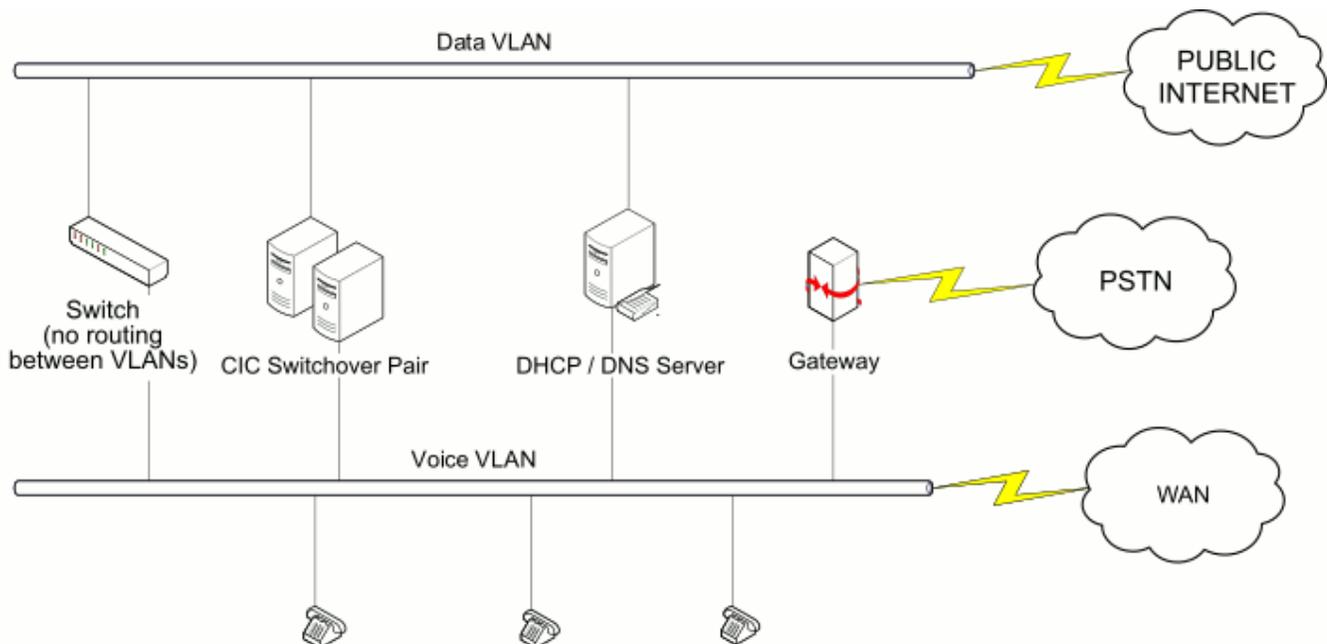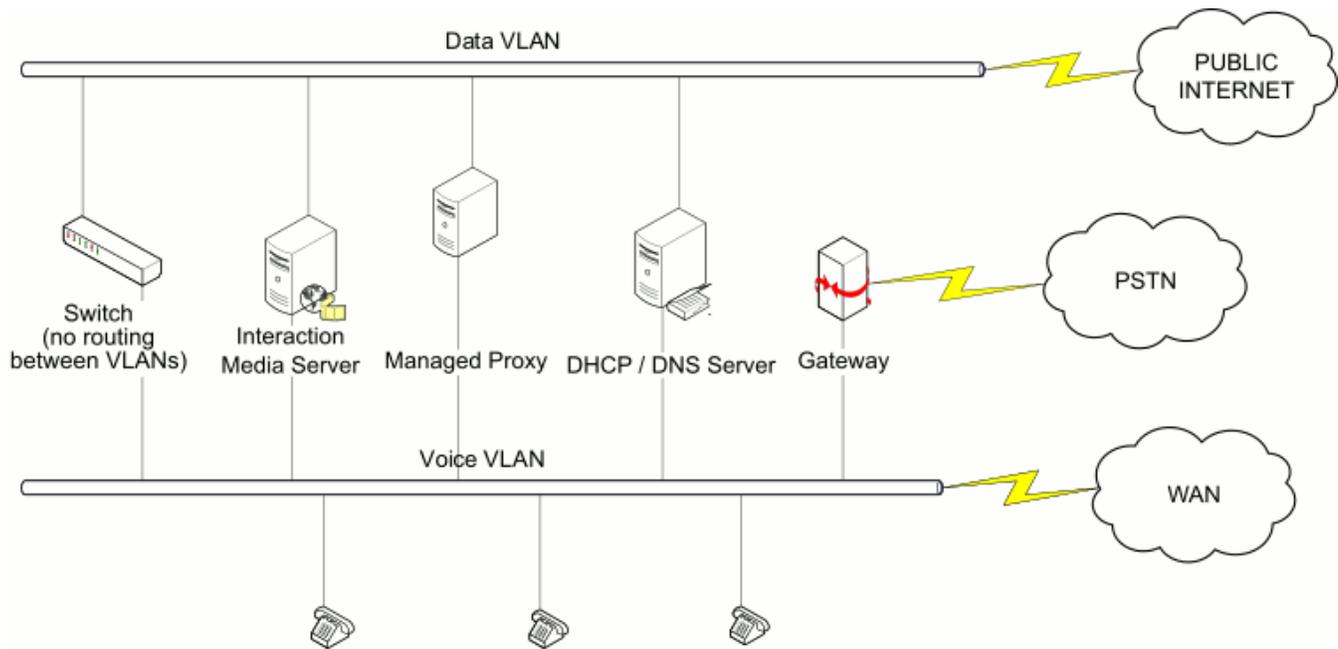
DNS and DHCP servers are separate, non-CIC Servers in the CIC system network. It allows for DHCP and DNS maintenance without putting the CIC server at risk. Use a static IP address for each of the non-phone devices (CIC server, Media Server, Proxy, Gateway) so that the system can contact them in the event of a reboot without the querying device needing to refresh its data.

For phones, use dynamic IP addresses that the DHCP server assigns to them for maintenance and configuration considerations.

IP Phone Network Configuration Task List

Determine the IP phone network settings you need for your implementation ahead of time so that managed IP phone process can run as smoothly as possible.

> **Note:**
> Interaction SIP Station is abbreviated as ISS.

Task	Polycom	ISS I and II, AudioCodes, and Genesys	SIP Soft Phone
Create a DNS domain for the voice VLAN	X	X	
Create DNS Host (A) records for Switchover	X	X	X
Create DNS SRV records for Switchover	X	X	X
Configure the TFTP server		X	
Create DHCP provisioning records	X	X	X
Configure time server	X	X	X
Implement QoS in your environment	X	X	X

Configure the Network for Managed IP Phones

Use the SIP Soft Phone on the Voice VLAN

It is possible to use the SIP Soft Phone on a different VLAN (your voice VLAN). The support for this method is not within the application layer, so the SIP Soft Phone itself cannot accomplish this without help at lower layers. Third-party drivers for many Network Interface Cards (NICs) allow the support for multiple VLANs through a single NIC port by creating virtual adapters. When adding each VLAN to the configuration, it creates a new *virtual* NIC, which looks like a second physical NIC to the Operating System. The SIP Soft Phone configuration allows you to select which NIC to use for SIP and RTP. Selecting the virtual NIC that communicates on the voice VLAN binds that data to that NIC, which the NIC drivers then assign to the voice VLAN. For more information about whether your workstation's NIC supports this feature, consult the manufacturer.

DNS Configuration

Create a Voice DNS Forward Lookup Zone

Genesys recommends that you create a "voice" DNS forward lookup zone to create a separate DNS domain for the voice VLAN (managed IP phones and SIP lines) for the following phones:
- Polycom
- Interaction SIP Station I and II
- AudioCodes
- Genesys

For example, you can create a DNS domain for voice, `yourvoicelan.voip`, in addition to the existing DNS domain for data, `yourdatalan.local` or `yourdatalan.com`, and set each NIC to register with the appropriate DNS server. (The examples used here are `lab.voip` and `lab.local`) This action ensures that when the DHCP scope provides the managed IP phones with the primary DNS lookup zone, the phones register correctly with the DNS voice domain.

You must be familiar with DNS server configuration and have required domain administrator permissions to complete the network configuration procedures.

To create a voice DNS forward lookup zone
1. On the computer that functions as the DNS server, click **Start->Programs->Administrative Tools->DNS**.
2. Right-click on **Forward Lookup Zones** and then click **New Zone…**
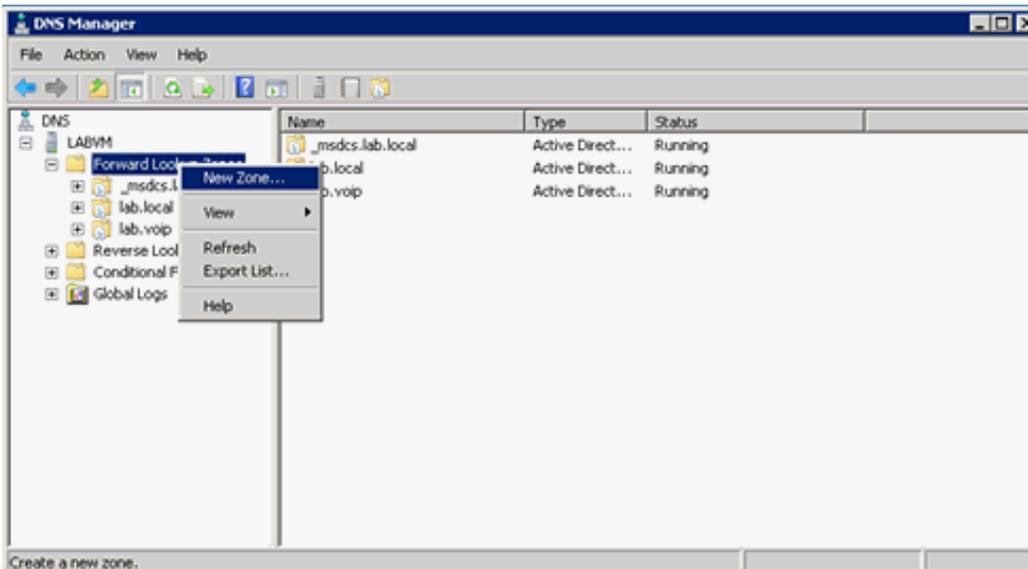
The **New Zone Wizard** launches.

3. On the **Zone Type** page, click **Primary Zone** (default).
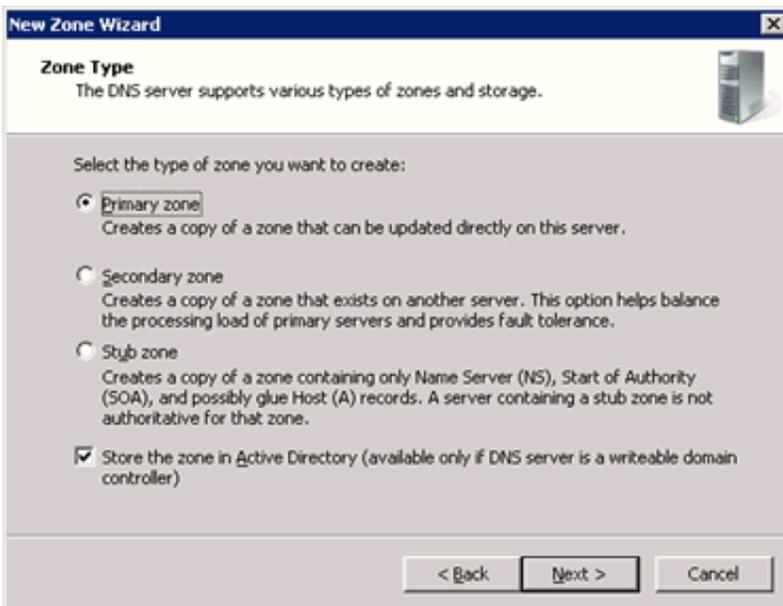
Leave **Store the zone in Active directory** selected (default).

4. On the **Active Directory Zone Replication Scope** page, click **To all DNS servers in the Active Directory domain** (default).
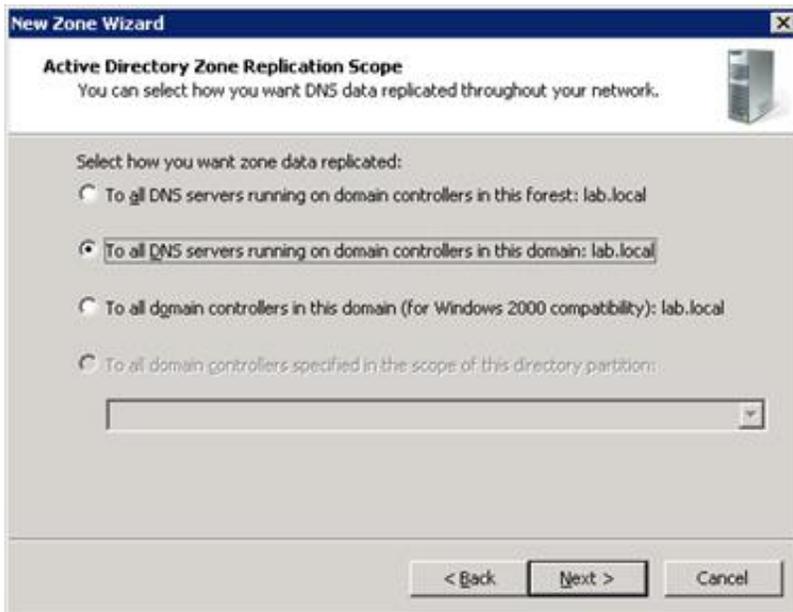
5. On the **Zone Name** page, type the name to use for the DNS voice domain (in this example, `lab.voip`).
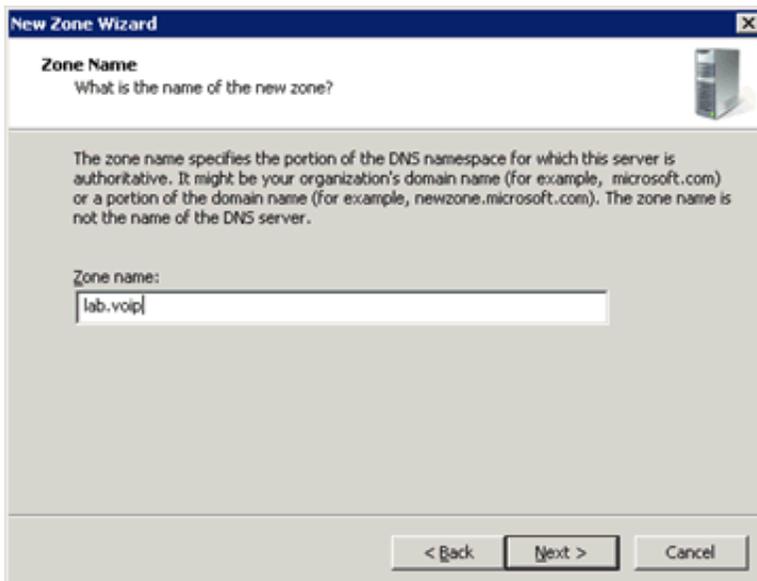
6. On the **Dynamic Update** page, click **Do not allow dynamic updates**.
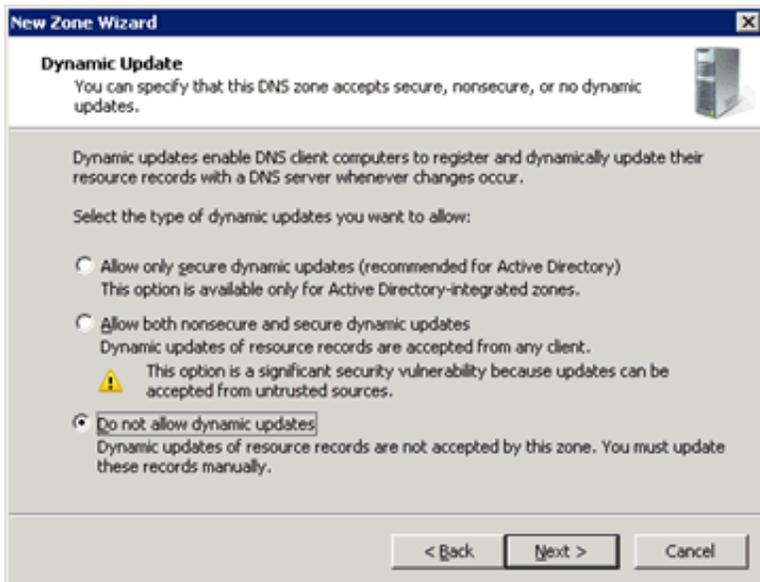
For a more secure environment, the server "voice" NIC IP addresses need to register in the voice DNS domain only. Genesys doesn't allow the server to register them dynamically.

7. Once the **New Zone** wizard completes, click **Finish**.

The new **lab.voip** forward lookup zone for the DNS voice domain is now available in the **Forward Lookup Zones** container.
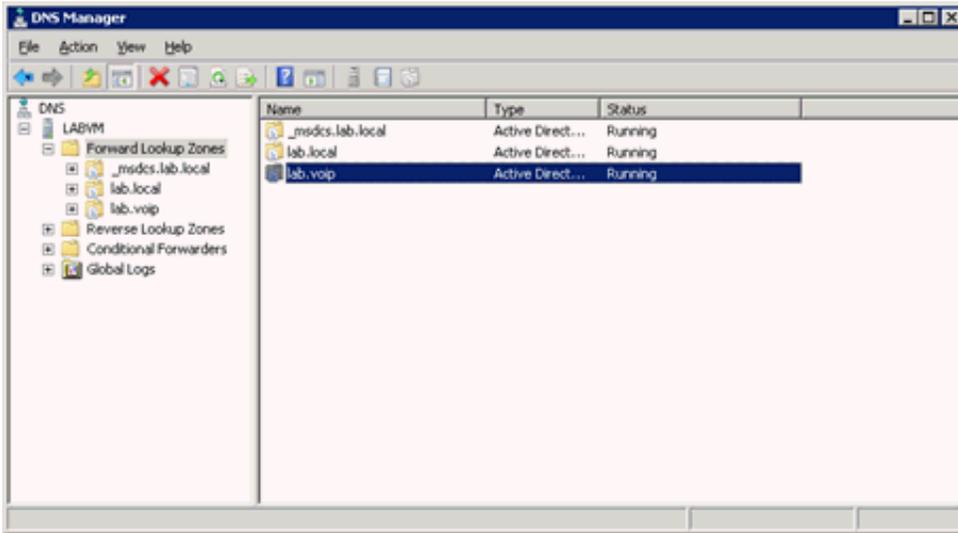
8. Complete the steps to Create DNS Host (A) Records for Servers in the DNS Voice Domain.

Create DNS Host (A) Records for Servers in the DNS Voice Domain

Create a DNS Host (A) record for each server in the DNS voice domain (for example, CIC server(s), media server, proxy server). This procedure is necessary because Genesys is administering the DNS records manually instead of allowing dynamic registrations. You must be familiar with DNS server configuration and have required domain administrator permissions to complete the network configuration procedures.

To create a DNS Host (A) record

1. Right-click on the new DNS Forward Lookup zone for voice (in this example, `lab.voip`) and then click **New Host (A or AAAA)**
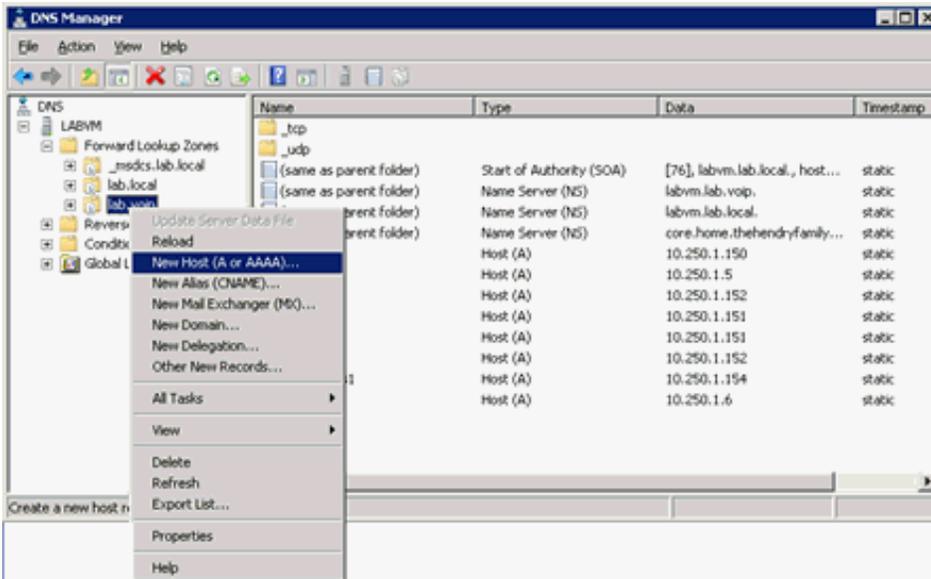
2. In **New Host** dialog box, complete the following information.
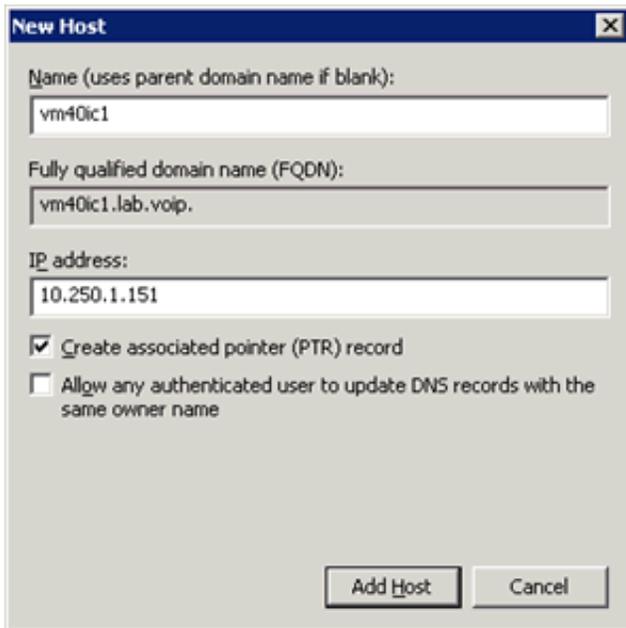
Name: Name of a server in the DNS voice domain (in this example, the CIC server `vm40ic1`).

IP address: IP address of this server (in this example, `10.250.1.151`). You can verify the IP address value for the selected server on the voice VLAN by typing **ipconfig** in the command-line on the server.

Create associated pointer (PTR) record: If selected, the system creates an associated pointer record for the new host. Ensure that you select this check box.

3. Click **Add Host**. A message appears, indicating that the host record (in this example, `vm40ic1.lab.voip`) created successfully.

4. Click **OK**.

5. Repeat the previous steps to create records for the second CIC server, Proxy Server, Media Server and other servers to include in the DNS voice domain. When finished, the records you created are available in the voice VLAN forward look up zone (in this example, `lab.voip`) container.
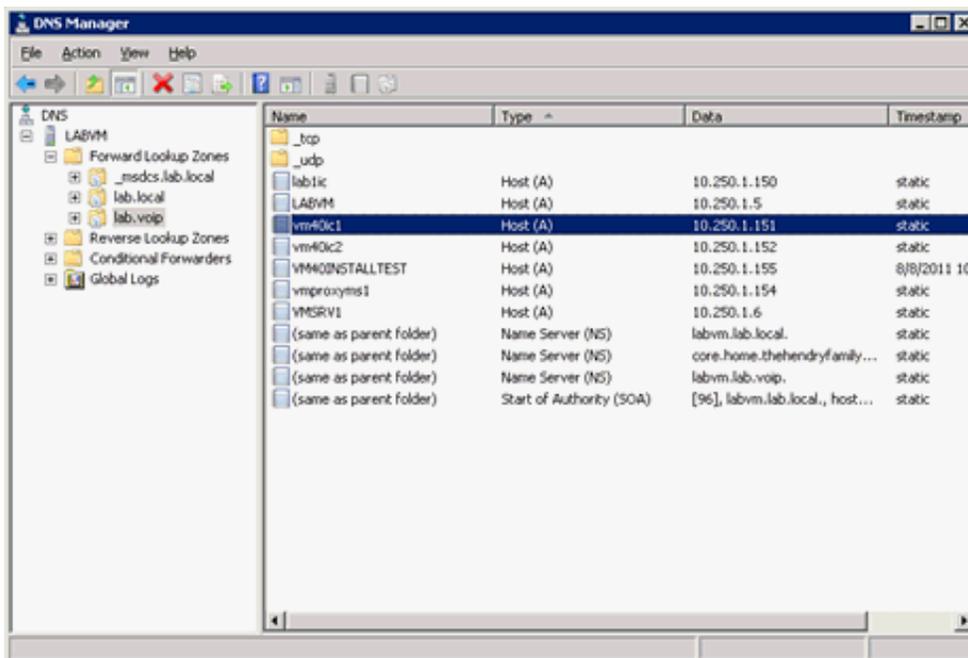
Create DNS Host (A) Records for Switchover

If your implementation includes a Switchover pair, complete this procedure for the following phones:
- Polycom
- Interaction SIP Station 1 and II

- Audiocodes
- Genesys

This procedure creates the DNS Host (A) records required for provisioning for the two CIC servers so that the phones can find the active server when booting up (provisioning) and completing SIP operations. You must be familiar with DNS server configuration and have required domain administrator permissions to complete the network configuration procedures.

Create two "provisioning" DNS Host (A) records for the Switchover pair with the same name, then use the different CIC server voice VLAN IP addresses, as shown in the following table containing sample entries.

Host (A) record name	Type	Address
provision.lab.voip	A	10.250.1.151
provision.lab.voip	A	10.250.1.152

> **Note:**
> - Genesys recommends that you use "provision" in the Host (A) record name to identify clearly the purpose of these records.
> - The Host (A) record name cannot match any existing record name.
> - Do not use the name of either CIC server.

To create the DNS Host (A) records for the Switchover pair

1. Right-click on the new DNS forward lookup zone for voice (in this example, `lab.voip`) and then click **New Host (A or AAA)** …
2. In the **New Host** dialog box, complete the following information.

   Name: Name of the provisioning Host (A) record for the Switchover pair (in this example, `provision`).
   IP address: IP address of the active CIC server (in this example, `10.250.1.151`).
   Create associated pointer (PTR) record: If selected, the system doesn't create an associated pointer record for the new host. Ensure that you clear this check box.

3. Click **OK**. A message appears, indicating that the host record (in this example, `provision.lab.voip`) created successfully.
4. Repeat the first two steps to create the second Host (A) record and then do the following:
   a. In the **Name** box, type the exact same name that you used for the provisioning Host (A) (in this example, `provision`).
   b. In the **IP address** field, specify the IP address of the backup CIC server (in this example, `10.250.1.152`).
   c. All other fields should be identical to the first Host (A) record created.

   When finished, the records that you created in the voice VLAN forward look up zone (in this example, `lab.voip`) appear in the container.
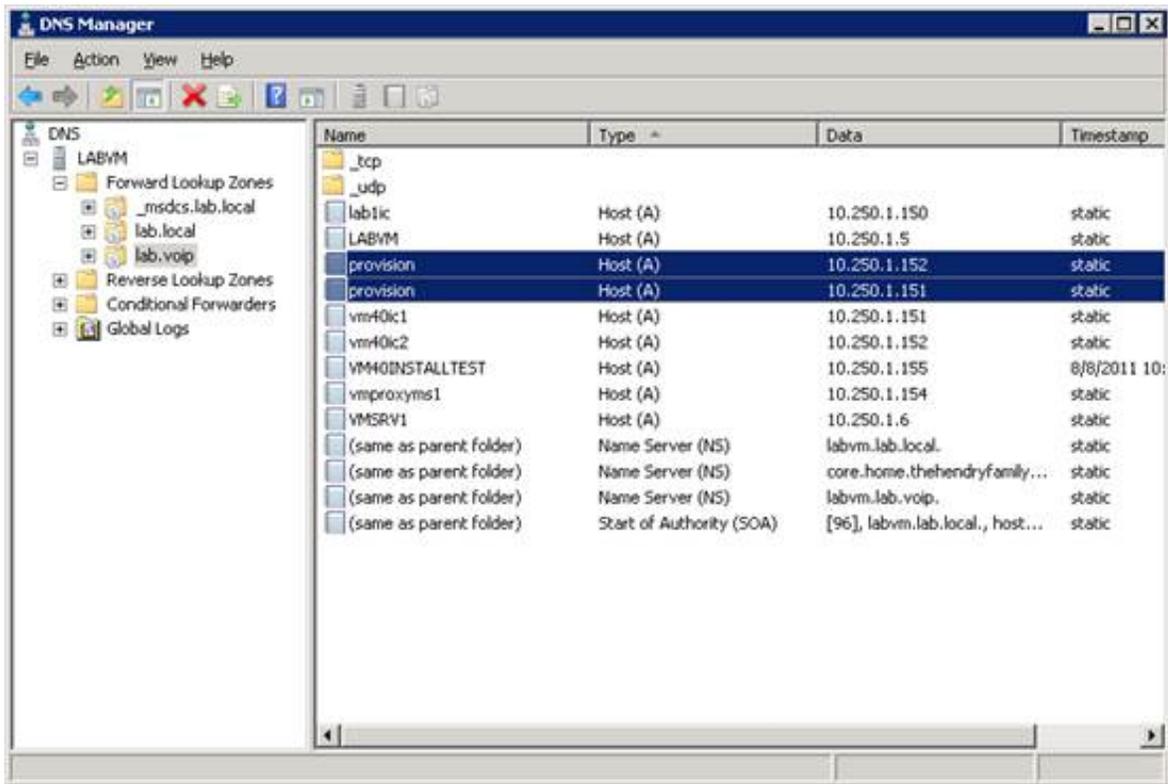
5. To verify the records that you created, run an **nslookup** on the command-line for this record, for example, `provision.lab.voip.` The server returns the IP addresses in a random order.
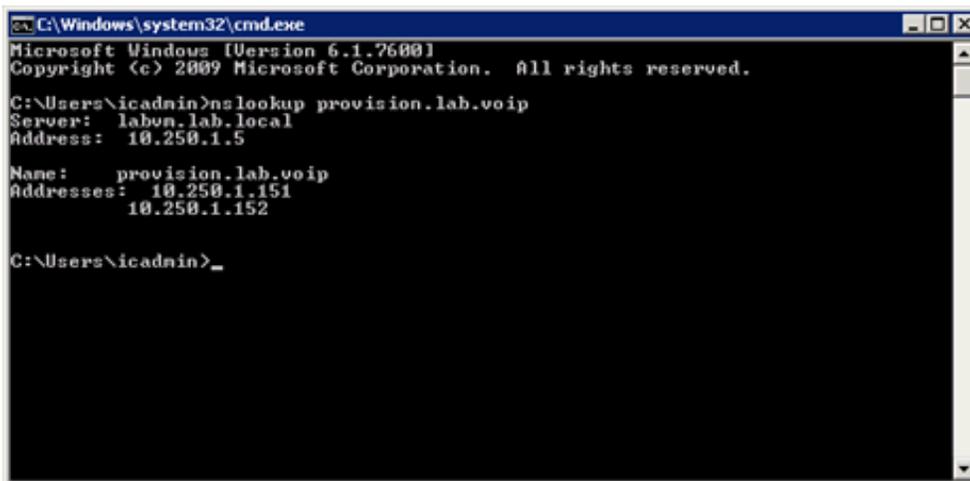
Create DNS Service Location (SRV) Records for Switchover

If your implementation includes a Switchover pair, complete the following procedures to create the DNS SRV records required for provisioning for the two CIC servers so that the phones can find the active server when booting up (provisioning) and completing SIP operations.

You need to create a DNS SRV record for each of the supported SIP line transport protocols (UDP, TCP, and TLS over TCP (SIP Secure or _sips)) that you plan to use for the Switchover pair. For example, if you plan to use the TCP protocol, create two DNS SRV records for TCP, one for the active CIC server and one for the backup CIC server. You must be familiar with DNS server configuration and have required domain administrator permissions to complete the network configuration procedures.

If you plan to use all three protocols, you need to create six DNS SRV records, as shown in the following table containing sample entries:

Service	Priority	Weight	Port	Hostname
`_sip._tcp`	0	0	8060	`vm40ic1.lab.voip`
`_sip._tcp`	0	0	8060	`vm40ic2.lab.voip`
`_sips._tcp`	0	0	8061	`vm40ic1.lab.voip`
`_sips._tcp`	0	0	8061	`vm40ic2.lab.voip`
`_sip._udp`	0	0	8060	`vm40ic1.lab.voip`
`_sip._udp`	0	0	8060	`vm40ic2.lab.voip`

Note:
- Port configuration may vary. The ports specified in the table are the default port configurations in CIC that match the default registration group and SIP line configurations.
- Interaction SIP Station I supports DNS SRV in CIC 4.0 SU 4 to SU 6 and CIC 2015 R1 and later, which contains v.1.2.2._p10_build_17 or later firmware.
- Interaction SIP Station I and II support DNS SRV in CIC 2015 R2 and later which contains v.2.0.4.15.7 or later firmware.
- AudioCodes and Genesys phones support DNS SRV in CIC 4.0 SU 5 to SU 6 and CIC 2015 R1 and later, which contains v2.0.0.18 or later firmware.

To create the DNS SRV records

1. Right-click on `lab.voip` to create two DNS domains for SRV (one for `_tcp` and one for `_udp`) and then click **New Domain**.
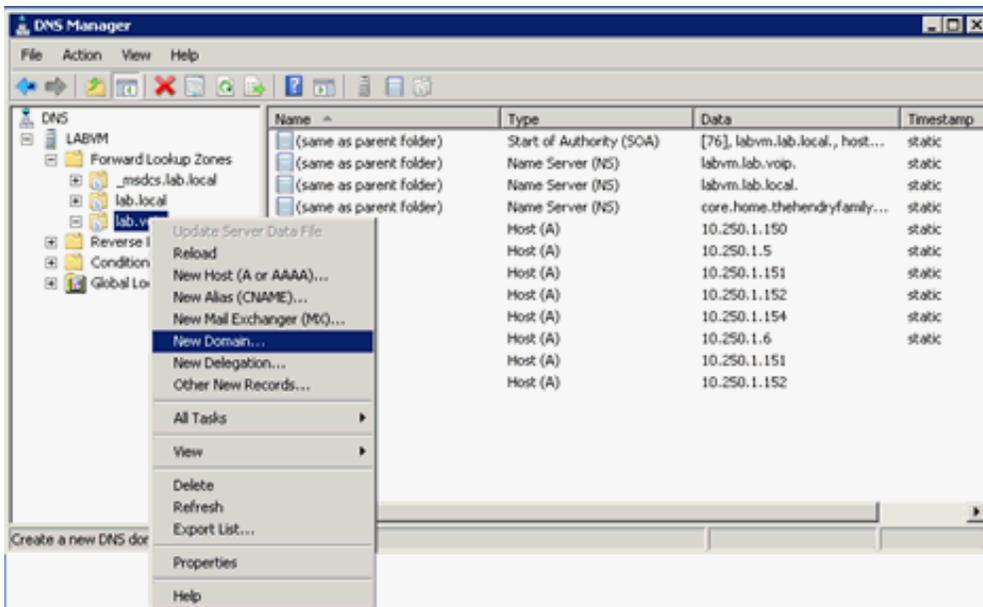
2. In the **New DNS Domain** dialog box, type **_tcp**.

3. Repeat steps 1 and 2 to create the _udp domain.
4. Right-click on the new DNS forward lookup zone for voice (in this example, `lab.voip`) and then click **Other New Records**
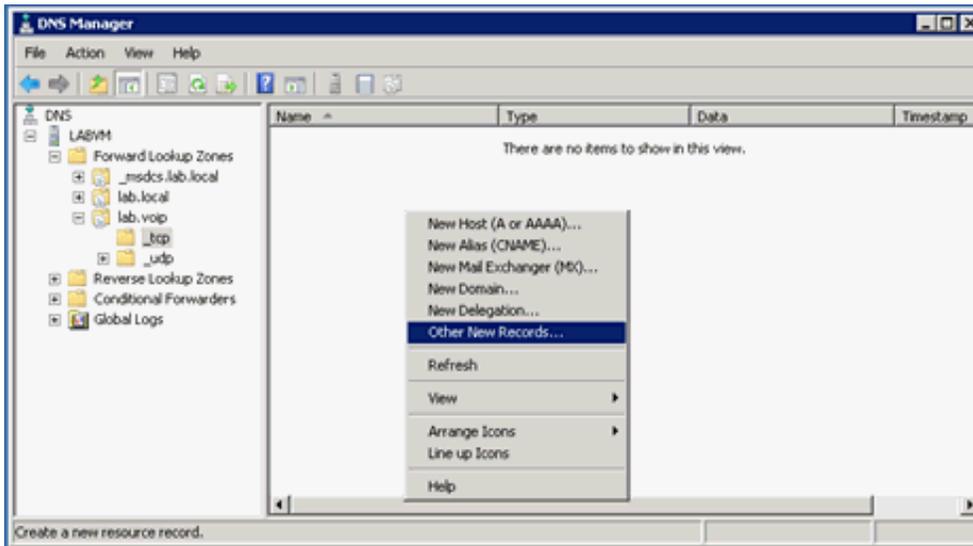
5. In the **Resource Record Type** dialog box, click **Service Location (SRV)** and then click **Create Record**.
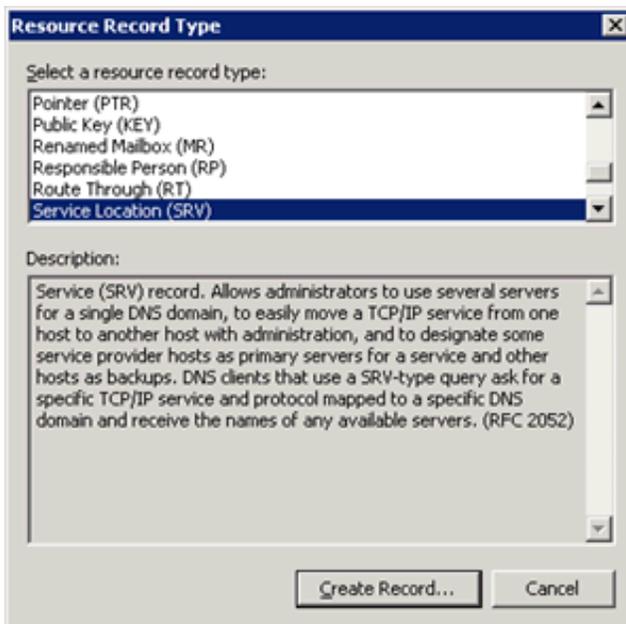
6. In the **New Resource Record** dialog box, complete the information for the first DNS SRV record for the active CIC server.

Service: Name of the service for this DNS SRV record (for example, `_sip`).

Protocol: Protocol for this DNS SRV record (for example, `_tcp`). The container for the protocol will be created if it does not already exist.

Port number: Port number for this DNS SRV record (for example, `8060`).

Priority: Leave the default setting.

Weight: Leave the default setting.

Host offering this service: FQDN name of the active CIC server (for example, `vm40ic1.lab.voip`).

7. Repeat steps 4 through 6 to create the DNS SRV record for the same protocol for the backup CIC server. Ensure that all the boxes in the **New Resource Record** identical are identical to the first DNS SRV record except for **Host offering this service.** Use the name of the backup server (in this example, `vm40ic2.lab.voip`).

8. Repeat steps 4 through 7 to create the rest of the DNS SRV records for the Switchover pair. Following is an example of the DNS SRV records created in the **_tcp** container for TCP and TLS over TCP.

9. To verify the records that you created in the **_tcp** container, run an `nslookup` on the command-line for this record, for example, `_sip._tcp.lab.voip`. The server returns the IP addresses in a random order.

Following is an example of the DNS SRV records created in the **_udp** container.

10. To verify the records that you created in the **_udp** container, do an **nslookup** on the command-line for this record, for example, `_sip_udp.lab.voip`. The server returns the IP addresses in a random order.

In Create Individual Managed IP Phones for Test Purposes, you configure the default Registration group to obtain registration

settings from the DNS SRV records that you created.

Enable the TFTP Server on the CIC Server

Interaction SIP Station I and II, AudioCodes, and Genesys phones obtain firmware and configuration using HTTP instead of TFTP, allowing for a faster upgrade process. However, they still use the TFTP server when an error occurs during a firmware upgrade process (recovery mode). Therefore, all sites with Interaction SIP Station I and II, AudioCodes, and Genesys phones need to enable the TFTP server on the CIC serverthe TFTP server.

You must enable the TFTP server on the IC Server to open up the TFTP server to accept TFTP requests, and then configure it to talk to the Provision server on the CIC server.

> **Note:**
> If your implementation includes a **Switchover pair,** enable the TFTP server **on only one of the servers** in the Switchover pair. The IP address of that server is the value used for DHCP Option 66.

To enable the TFTP server on the CIC server

1. Click **Start->Programs->PureConnect...IC Setup Assistant.** The **IC Setup Assistant** page appears.
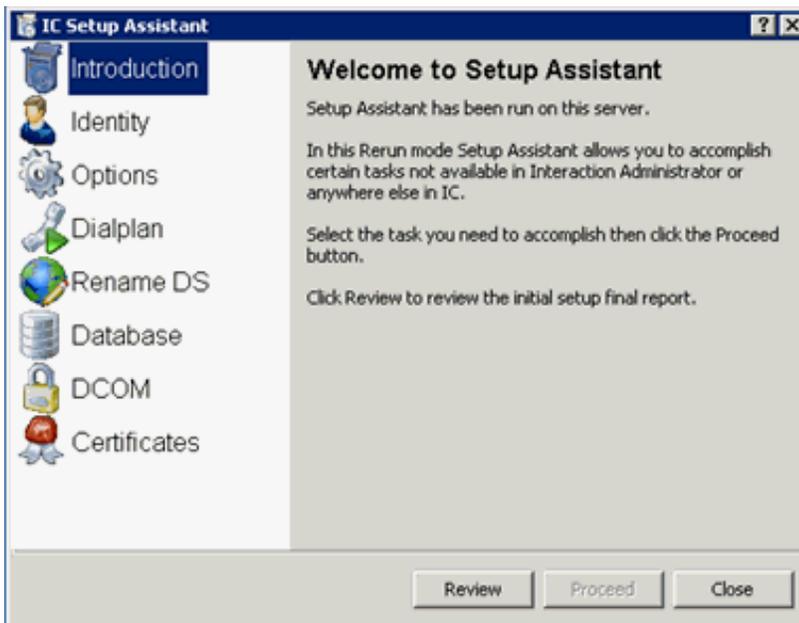
2. Click **Options**.
3. On the **Select IC Optional Components** page, if not selected, select **TFTP Server**.
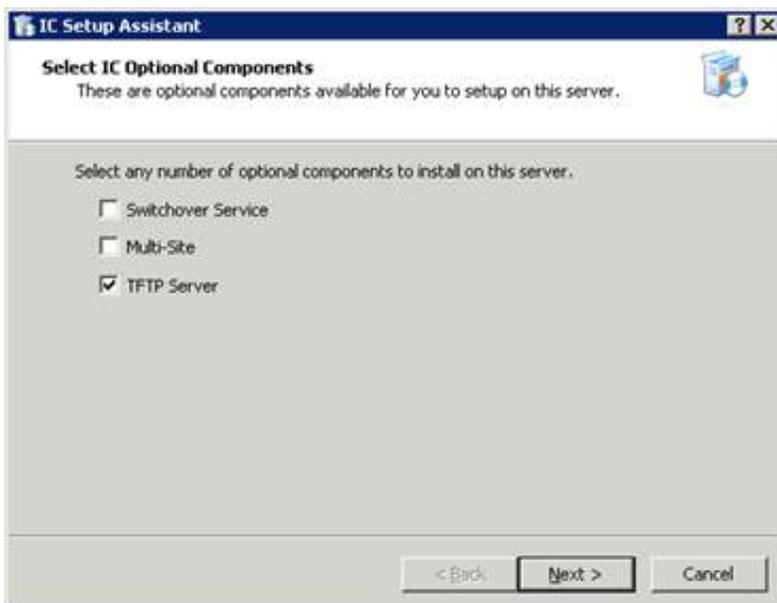
> **Note:**
> If your implementation includes a **Switchover pair, Switchover Service** is selected also.

Create DHCP Provisioning Records

Required DHCP Option Provisioning Records

The following table describes the required DHCP server provisioning records configuration for each managed IP phone type. For notes about each managed IP phone type, see the following:

- Polycom DHCP Record Notes
- Interaction SIP Station I and II, AudioCodes, and Genesys DHCP Record Notes
- SIP Soft Phone DHCP Record Notes

> **Note:**
> ISS is the abbreviation for Interaction SIP Station.

Option record	Purpose	Example value	Type	Polycom	ISS I and II, AudioCodes, and Genesys Phones	SIP Soft Phone
002	Time Offset	FFFF.B9B0 (hexadecimal input value) -18000 (converted value)	Long	X	X	
004	Time Server	mytimeserver.lab.voip	Array	X	X	
006	DNS Server	10.250.1.5 10.250.0.2	Array	X	X	X
015	DNS Domain Name	lab.voip	String	X	X	
042	NTP Server	mytimeserver.lab.voip	Array	X	X	
066	TFTP Server IP Address	10.0.0.11	String		X	
067	Firmware Filename	sip100.img Interaction SIP Station I and II 405.img AudioCodes 405, Genesys 405 405hd.img AudioCodes 405HD, Genesys 405HD 420hd.img AudioCodes 420HD, Genesys 420HD 430hd.img AudioCodes 430HD, Genesys 430HD 440hd.img AudioCodes 440HD, Genesys 440HD	String		X	
132	VLAN ID	100	String		X	
160	Provisioning URL	http://provision.lab.voip:8088	String	X	X	X

Polycom DHCP Record Notes

Time server settings

Options 002, 004, and 042 configure the time server on the DHCP server. Alternatively, you can choose to configure the time server in Interaction Administrator. For more information, see Configure the Time Server.

- **Option 002**: The values for Option 002 require hexadecimal numbers. To convert values to hexadecimal numbers, see How to Calculate the Hexadecimal Value for DHCP Option 2 (time offset).
- **Options 004 or 042**: Set *either* option 004 *or* 042, not both. Genesys recommends setting multiple values in the Option 004 or 042 record, particularly for CIC systems using TLS, to provide redundancy. If you do not specify values for one of these options, the provisioning server uses the default location and SNTP server settings provided in the phone configuration files, based on the region of the phone.

Option 015

Use the DNS domain for the voice VLAN for the Option 015 record.

Option 160

- Option 160 is the primary default record that Polycom phones use to find the provisioning server.
- Option 160 takes precedence over Option 066 (the secondary default record that Polycom phones use to find the provisioning server). Genesys recommends that you do *not* use Option 066 for Polycom phones.
- **Switchover environments**: Ensure that you created the **DNS Host (A) records** as described in Create DNS Host (A) Records for Switchover and **DNS Service Location (SRV) records** as described in Create DNS Service Location (SRV) Records for Switchover. For DHCP Option 160, use the DNS Host (A) provisioning record in the URL.

Interaction SIP Station I and II, AudioCodes, and Genesys DHCP Record Notes

Time server settings

Options 002, 004, and 042 configure the time server on the DHCP server.

- **Option 002:** The values for Option 002 require hexadecimal numbers. To convert values to hexadecimal numbers, see How to Calculate the Hexadecimal Value for DHCP Option 2s(time offset).
- **Options 004 or 042**: Set *either* option 004 or 042, not both. If you do not specify values for one of these options, the provisioning server uses the default location and SNTP server settings provided in the phone configuration files, based on the region of the phone.

All DHCP Option records

- Set all DHCP Options on the DHCP server in both the data and voice VLANs.

Option 066

- Option 066 is the IP address of the TFTP server used in recovery mode.
- Ensure that you **enabled and configured the TFTP server** as described in Enable the TFTP Server on the CIC Server.
- **Switchover environments**: Ensure that you enabled and configured the TFTP server on the initial active server only, as described in Enable the TFTP Server on the CIC Server. Set DHCP Option 066 on the **initial active server only**.

Option 067

- Option 067 is for implementations using different managed IP phone types and models, requiring recovery mode. To handle the conflict, set up Option 067 according to Vendor Class. The value for Option 067 is one of the following:
  - `sip100.img` (Interaction SIP Station I firmware filename)
  - `sip200.img` (Interaction SIP Station II firmware filename)
  - `405.img` (AudioCodes 405 or Genesys 405 firmware filename)
  - `405hd.img` (AudioCodes 405HD or Genesys 405HD firmware filename)
  - `420hd.img` (AudioCodes 420HD or Genesys 420HD firmware filename).
  - `430hd.img` (AudioCodes 430HD or Genesys 430HD firmware filename).
  - `440hd.img` (AudioCodes 440HD or Genesys 440HD firmware filename).

    See Configure DHCP Records for Multiple Phone Model Recovery.

Option 132

- If your Interaction SIP Station I, Interaction SIP Station II, AudioCodes phones, or Genesys phones communicate on any VLAN other than the native VLAN (use a voice VLAN), use Option 132. Interaction SIP Station I and II, AudioCodes, and Genesys phones have a special "recovery mode" that it falls back to under certain conditions, such as a firmware update failure. This fail-safe mode brings the device up in the native VLAN and ignores any CDP/LLDP VLAN assignment. Since CDP/LLDP is ignored, the only way to push the device to the voice VLAN is to set option 132 in the native VLAN DHCP scope. You do not need to set this option in the voice VLAN DHCP scope. Failure to set this option causes any device that goes into "recovery mode" for any reason to stay in that mode until it either finds provisioning information on native VLAN, or obtains an option 132 message in the native VLAN DHCP assignment. **Option 132 has no effect in normal (non-recovery) mode**.

> **Note:**
> 2.0.4+ firmware caches the VLAN tag acquired from CDP/LLDP during normal operations. This caching negates the need for Option 132 unless the phone factory reset or otherwise lost the cache. For this reason, configure Option 132 in the native VLAN.

Option 160

- Option 160 is the default record that Interaction SIP Station I and II, AudioCodes phones, and Genesys phones use to find the provisioning server.
- **Switchover environments**: Ensure that you created the DNS Host (A) records as described in "Create DNS Host (A) records for Switchover". For DHCP Option 160, use the DNS Host (A) provisioning record in the URL.

SIP Soft Phone DHCP Record Notes

Option 006

Option 006 is the default DHCP record used for DNS server discovery. Use the SIP Soft Phone with DNS to allow referencing the CIC server by name. You can also set up DNS discovery manually in the client workstation's network configuration.

Option 160

Option 160 is the default record that the SIP Soft Phone uses to find the provisioning server. It is the only DHCP record required for the SIP Soft Phone.

Configure DHCP Option Records

Use this procedure to configure DHCP option records for each of the managed IP phone types. You must be familiar with DHCP server configuration and have required domain administrator permissions to complete the network configuration procedures.

The following example creates DHCP Option 160.

To configure DHCP option records

1. On the computer that functions as the DHCP server, click **Start->Programs->Administrative Tools->DHCP**, or use **Server Manager** to open DHCP.
2. Right-click the DHCP server (if using a Windows 2008 server, right-click **IPv4**) and then click **Set Predefined Options**
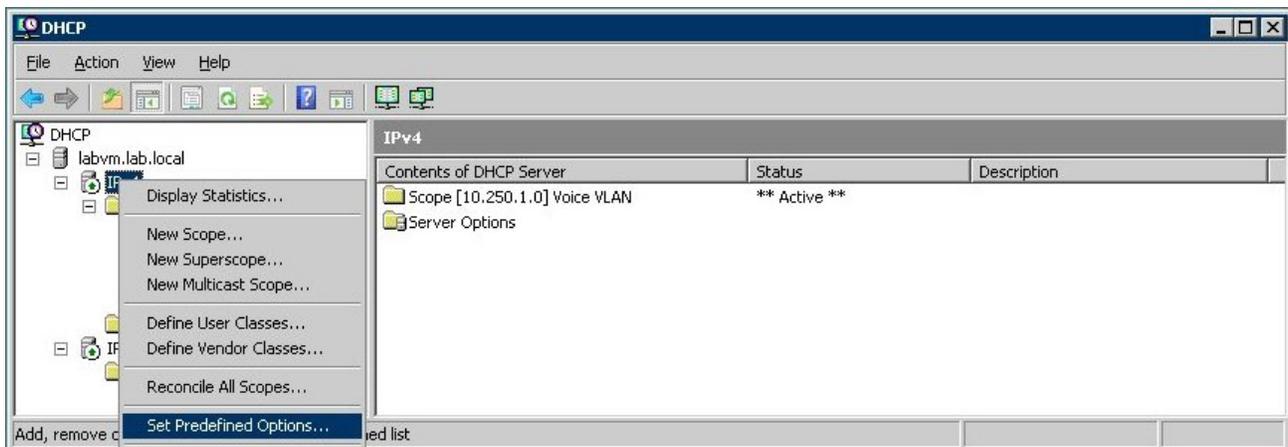
3. In the **Predefined Options and Values** dialog box, click **Add**.
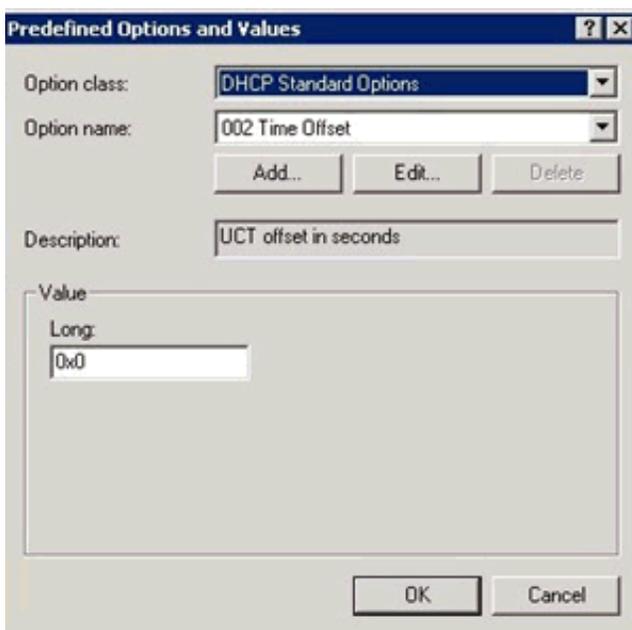
4. In the **Option Type** dialog box, complete the information.

Name: Name for this record, related to IP phone provisioning.

Data type: Data type (in this example, **String**).

Code: Option record code (in this example, **160**).

Description: Description for this record, related to IP phone provisioning.

5. Click **OK**.
6. In the **Predefined Options and Values** dialog box, in the String **Value** box, type the URL for the IP phones to use to contact the provisioning server (for a Switchover pair, use the `provisioning` DNS Host (A) record).
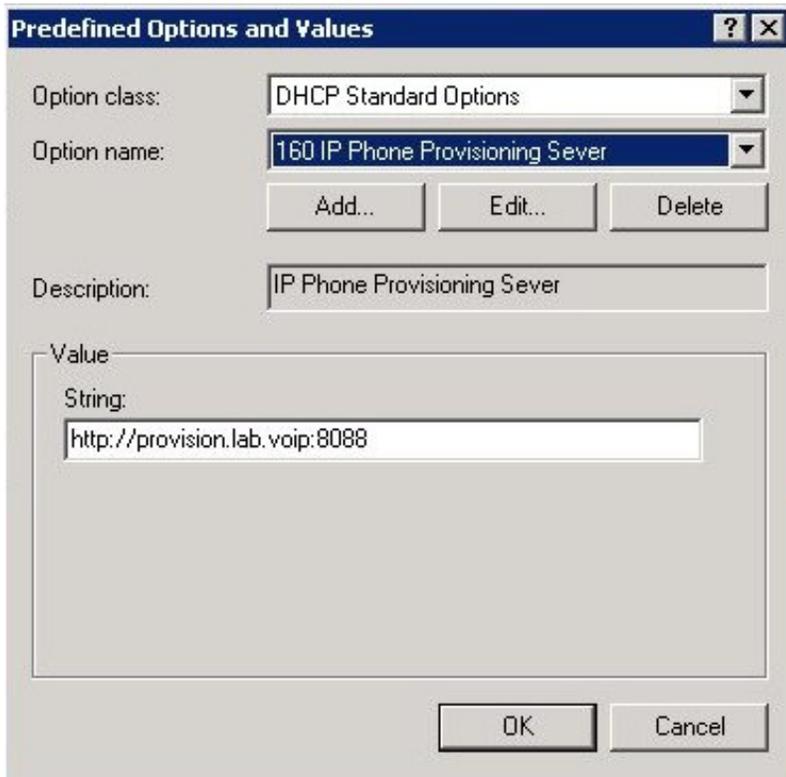
7. Click **OK**.
8. Right-click the **Scope Options** (or **Server Options**) container and then click **Configure Options**.
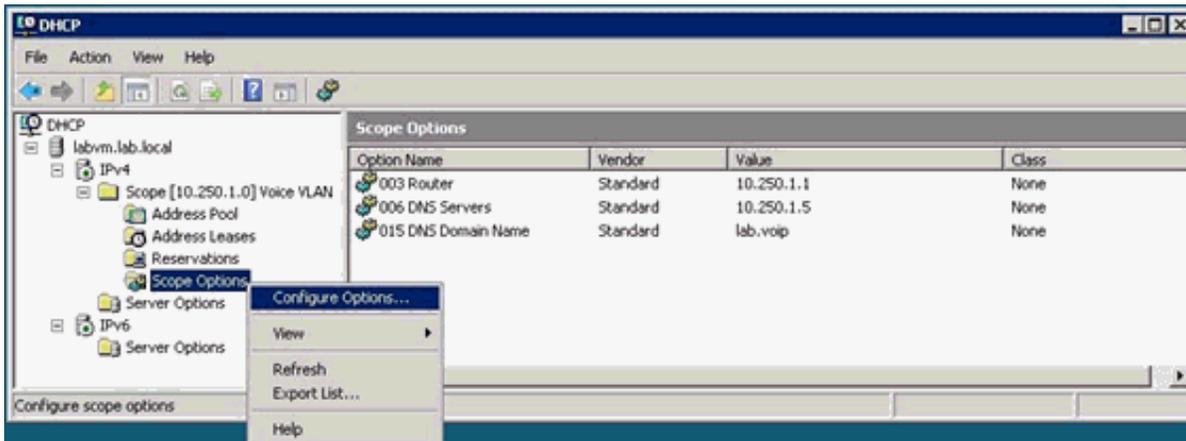
9. In the **Scope Options** dialog box, select the check box for the option record that you created (in this example, **160 IP Phone Provisioning Server**).
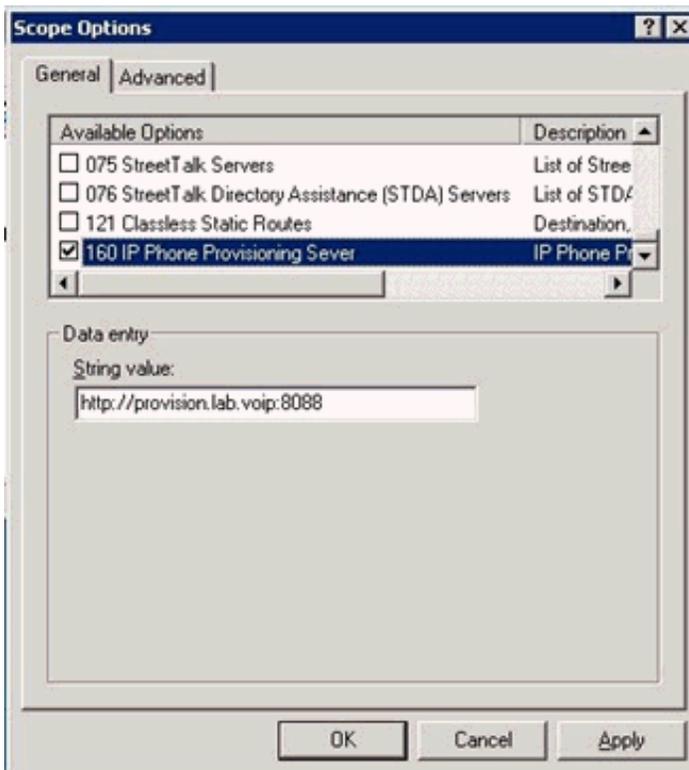
10. Repeat these steps to create the other DHCP option records required for the appropriate managed IP phone type.

Configure DHCP Records for Multiple Phone Model Recovery

Some environments use multiple models requiring recovery mode. For example, some environments use AudioCodes, Genesys, Interaction SIP Station I, and Interaction SIP Station II phones. Each requires a different Option 67 value. To handle the conflict, environments using multiple models need to set up Option 67 according to Vendor Class.

You must be familiar with DHCP server configuration and have required domain administrator permissions to complete the network configuration procedures.

To configure DHCP records for multiple phone model recovery
1. Right-click the **IPv4** container on the DHCP server and then click **Define Vendor Classes**.
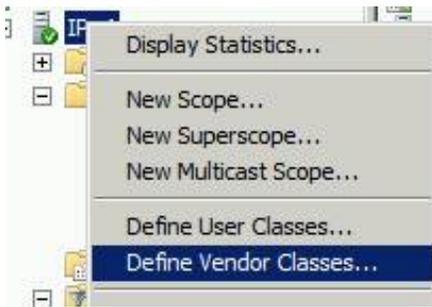
2. Click **Add...**
3. Type a **Display name**.
4. Click under **ASCII**.
5. Type the **Vendor Class** (e.g. SIP100, SIP200, or 420HD).
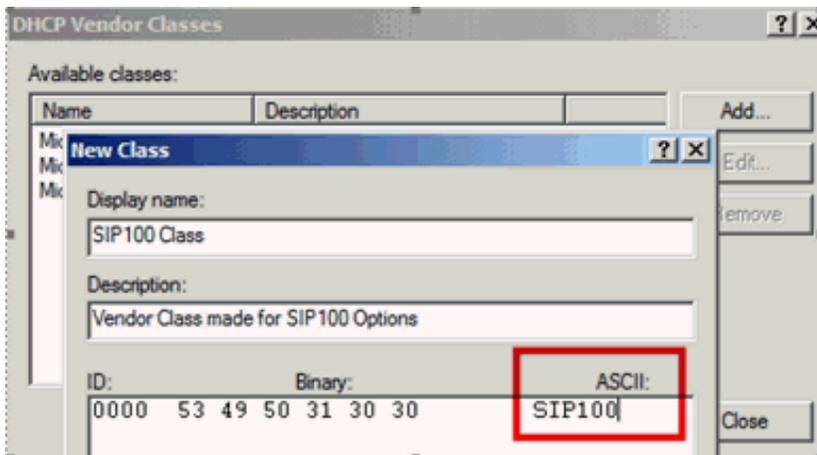
6. Click **OK**.
7. Click **Close**.
8. Right-click the **IPv4** container.
9. Click **Set Predefined Options...**
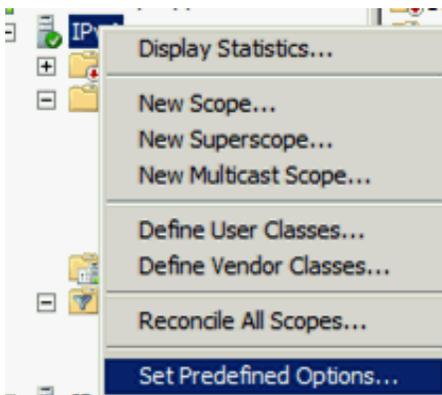
10. Set the **Option class** to the **Vendor Class** that you previously created in this procedure.
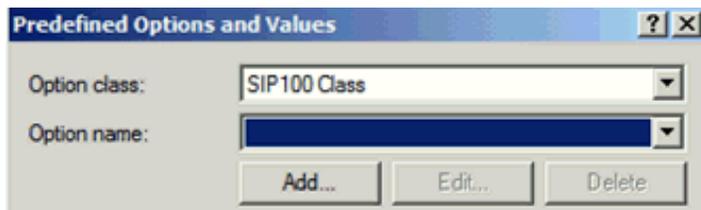
11. Click **Add...**
12. Type a descriptive name for Option 67.
13. Set the **Data type** to **String**.

14. Type `Code 067`.

15. Click **OK**.
16. Click the firmware file name.
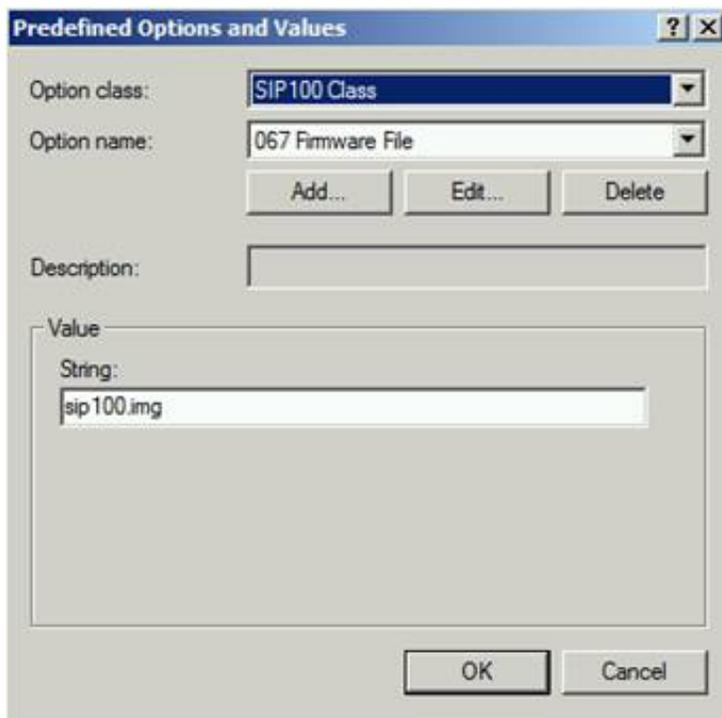
When the phone boots up, it identifies its Vendor Class in DHCP Option 60 of the Discover. The DHCP server uses this value to match the appropriate Option 67.

Configure the Time Server

In Polycom, Interaction SIP Station I and II, AudioCodes, and Genesys managed IP phone implementations, each managed IP phone requests the time from the time server (SNTP provider) in the region where the phone is. You can configure the time server for these three managed IP phone types either on the DHCP server or in Interaction Administrator.

> **Important!**
> Whether configured on the DHCP server or in Interaction Administrator, **the time server configuration is crucial in CIC systems using TLS**. If a managed IP phone cannot connect to a time server to determine the time, it cannot do certificate expiration validation needed to authenticate a secure SIP connection. The phone cannot register with the CIC system. Further, if the managed IP phone cannot connect to a time server when it boots up, the phone fails. **It is important that the specified time server is available when the phones boot up**.

Configure the time server on the DHCP server

The DHCP Option records for the time server are Options 002, 004, and 042. See Polycom DHCP Record Notes and Interaction SIP Station I and II, AudioCodes, and Genesys DHCP Record Notes.

Configure the time server in Interaction Administrator

You can configure a Simple Network Time Protocol (SNTP) server in Interaction Administrator.

> **Note:**
> - When you run IC Setup Assistant in a new or upgrade installation, Setup Assistant sets the CIC server for use as the SNTP server.
> - Post-installation, either before or after you created managed IP phones in Interaction Administrator, you must configure the SNTP server in the Interaction Administrator **Regionalization** container so that it is available before rebooting the phones. The default is to use the CIC server as the SNTP server.
> - If the DHCP server provides the time server settings, they override the location and SNTP server settings by default. You can change it so that the location and SNTP server settings provided in the provisioning server's phone configuration files override the DHCP server settings. To do, change the following *for each managed IP phone* in the Interaction Administrator **Managed IP Phones** container: Right-click on the phone, click **Properties > Options > Advanced Options** and then change **Configuration Time Zone Overrides DHCP and Configuration NTP Server Overrides DHCP** to **Yes**.

To configure the time server in Interaction Administrator

Complete this procedure for the Default Location and all other locations configured in the **Regionalization** container.

1. In the Interaction Administrator **Regionalization** container, select **Locations**.
2. Right-click **<Default Location>** and then click **Properties**.
3. On the **Configuration** tab, in the **SNTP Server** box, do one of the following:
   - Keep the default selection **Use CIC server**. Interaction Administrator uses the Windows Time service on the CIC server to synchronize the time on managed IP phones.
   - Type the name or IP address of a valid time server in the **other** box, for example, `176.10.10.199`. If you don't know the time server's IP address, check with IT. Note that Interaction Administrator does not validate the time server that you specify in this box.
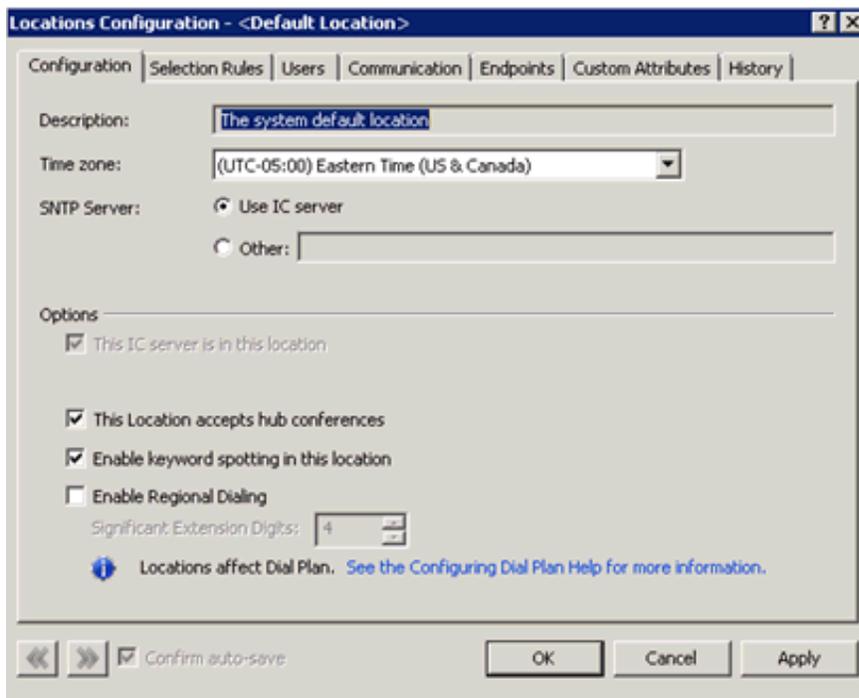
4. Repeat steps 1 through 3 for the other locations configured in the **Regionalization** container.

Implement QoS in Your Environment

Following are the PureConnect QoS driver, SIP Soft Phones and QoS considerations, and recommended QoS configuration settings for RTP and SIP for all managed IP phone types. For general information about implementing QoS on CIC systems, see the *PureConnect Quality of Service Technical Reference*.

PureConnect QoS driver

In CIC, Genesys developed a new QoS driver to more closely integrate with Customer Interaction Center. It is the default QoS driver for CIC. It operates independently of the Windows QoS Packet Scheduler, which was the recommended QoS driver for CIC 3.0.

Installs that use the QoS feature include the following:

Application name	Install name	OS
CIC Server	`ICServer.msi`	64-bit
IC User Applications 32-bit	`ICUserApps_32bit.msi` with one or both of these features selected: • **SIP Soft Phone** • Interaction Screen Recorder Capture Client	32-bit
IC User Applications 64-bit	`ICUserApps_64bit.msi` with one or both of these features selected: • **SIP Soft Phone** • Interaction Screen Recorder Capture Client	64-bit

For a complete list of installs that use the QoS feature, see the [PureConnect KB article](#).

The default behavior for most of these installs is to install the PureConnect QoS driver silently and add the certificate to the Trusted Publishers list. The IC User Applications installs are the exception. If you prefer to modify this default behavior, see the [PureConnect KB article](#).

SIP Soft Phone and QoS considerations

Among managed IP phones, SIP Soft Phones have special considerations because audio originates on the computer instead of a Polycom phone or Interaction SIP Station I and II phones. The IC User Applications install (32-bit and 64-bit) with the SIP Soft Phone feature selected is one of the installs that installs the PureConnect QoS driver.

- If the IC User Applications install is run in "Full UI mode" (through `Setup.exe` or double-clicking the `.msi`), the QoS Requirement page appears, asking the user to choose the QoS driver installation and recommending installation of the PureConnect QoS driver.
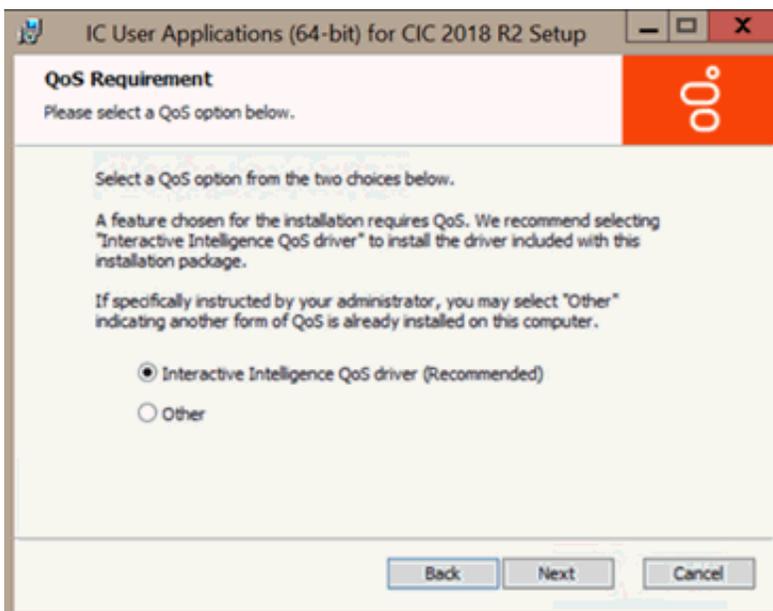
    PureConnect QoS driver: If selected, the install adds the PureConnect certificate to the Trusted Publishers list, and then installs the driver.

    Other: If selected, the install doesn't add the PureConnect certificate to the Trusted Publishers list and doesn't install the driver. Genesys recommends that you don't select this option unless your administrator instructs you to. Selecting this option implies use of another form of QoS.

    For more information, see the [PureConnect KB article](#).

- If you run the IC User Applications install using Group Policy; startup or logon script; or command-line, the default is to install the PureConnect QoS driver silently and add the certificate to the Trusted Publishers list silently. To modify the default PureConnect QoS driver installation, use one of these methods to modify the QoS feature properties and run the install, as described in the [PureConnect KB article](#).

Configure QoS Settings for RTP and SIP

The QoS value passes to manage IP phones during provisioning using the phone configuration file. The value stores as a Byte value in Interaction Administrator, and then converts to a DSCP decimal value. The value can range from 0 to 63 (decimal).

Genesys recommends that you modify the QoS values for all managed IP phone types to make client workstations compatible with other devices and systems on the network.

Configure the QoS setting for RTP (SIP station)

To configure the QoS setting for RTP packets for managed IP phones and associated SIP station audio, create one or more individual managed IP phones first.

To configure the QoS setting for RTP

1. Create Individual Managed IP Phones for Test Purposes.
2. In Interaction Administrator, open a managed IP phone in the **IP Managed Phones** container.
3. In the **Managed IP Phone Configuration - Options** tab, click **Advanced Options**. The **Advanced Options - SIP Options** dialog box appears, with the **Audio** page selected.
4. In the **RTP DSCP Value** box, specify the QoS setting and then click **OK**. The optimal setting for VoIP packets is `B8` (hex), which is equivalent to a DSCP value of `46`. To disable DSCP tagging, set the value to `0`.

> **Note:**
> The provisioning server reads this value and includes it in the phone configuration file.
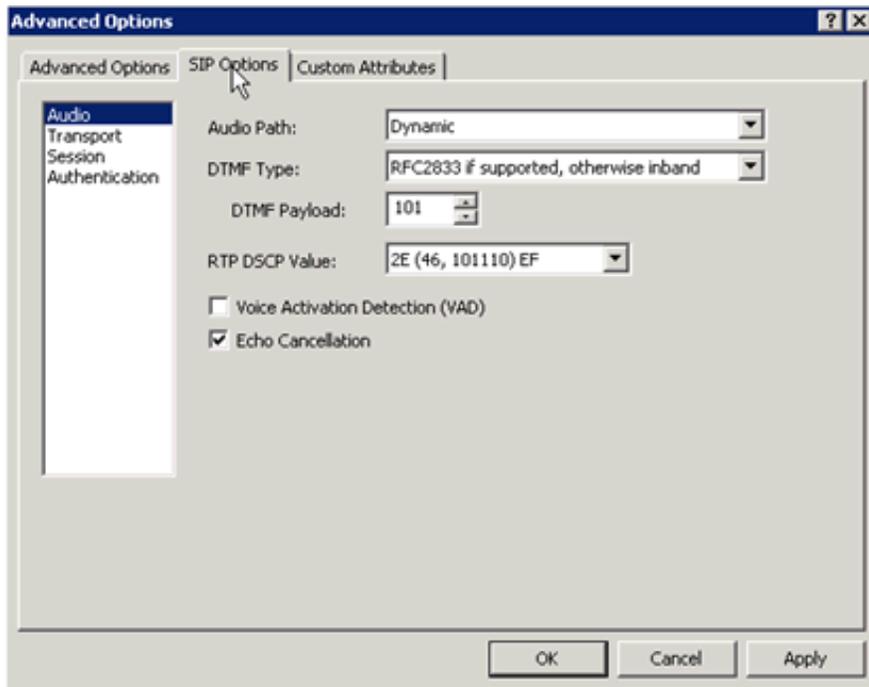
Configure the QoS setting for SIP (SIP lines)

You can configure the QoS value for SIP **globally or by individual managed IP phone**.

Configure the QoS Setting for SIP (global)

To configure the QoS setting for SIP (global)

1. In Interaction Administrator, open **Default Station Configuration** in the **Stations** container. The **Default Station Global SIP Station** dialog box appears, with the **Audio** page selected.
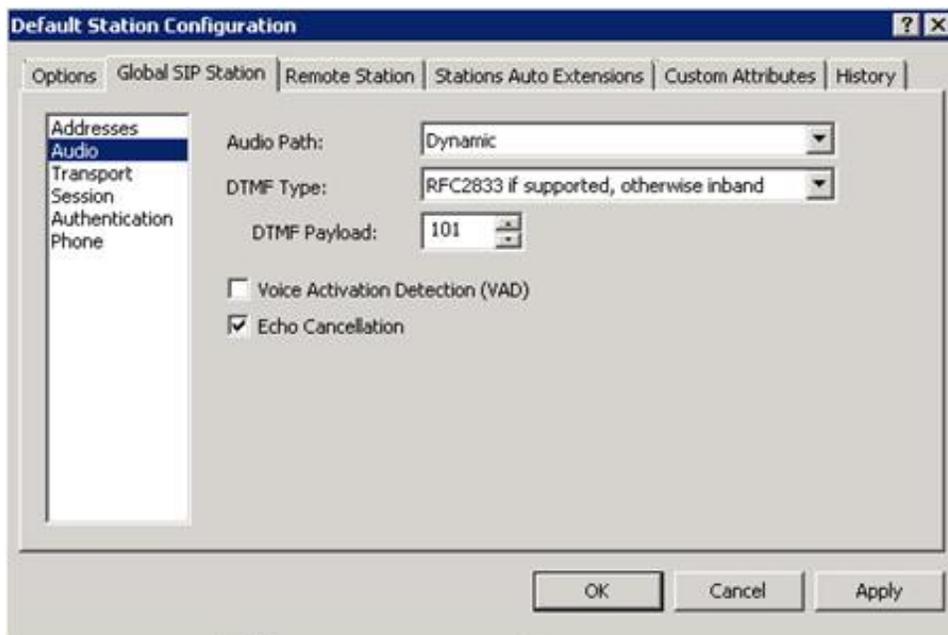
2. Click **Transport** in the left column. The **Transport** dialog box appears.

3. In the **SIP DSCP Value** box, specify the QoS setting. The optimal setting for VoIP packets is `60` (hex), which is equivalent to a DSCP value of `24`. To disable DSCP tagging, set the value to `0`.

> **Note:**
> The provisioning server reads this value and includes it in the phone configuration file.
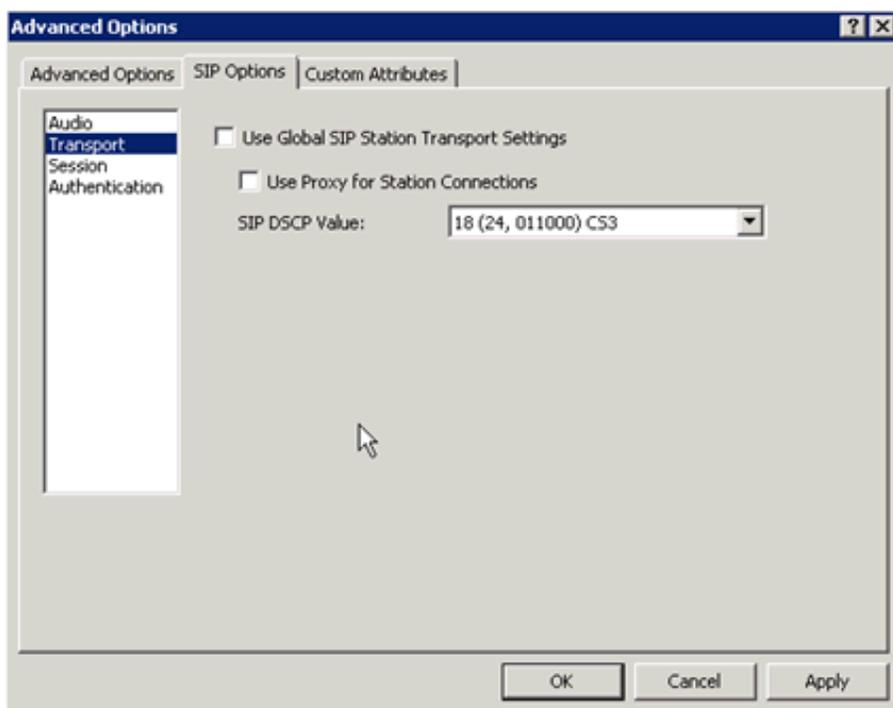
Configure the QoS setting for SIP (individual managed IP phone)

To configure the QoS setting for SIP (individual managed IP phone)
1. Create Individual Managed IP Phones for Test Purposes.
2. In Interaction Administrator, open a managed IP phone in the **IP Managed Phones** container.
3. In **Managed IP Phone Configuration,** click the **Options** tab and then click **Advanced Options**. The **Advanced Options - SIP Options** dialog box appears, with the **Audio** page selected.
4. Click **Transport** in the left column. The **Transport** dialog box appears.

5. Clear the **Use Global SIP Station Transport Settings** check box. The **SIP DSCP Value** box is available.
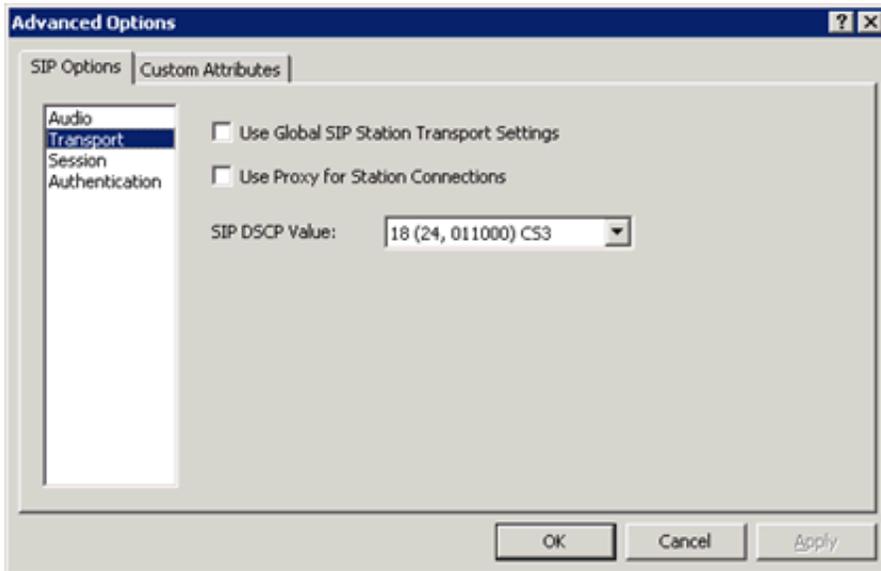
6. In the **SIP QOS Byte** box, specify the QoS setting. The optimal setting for VoIP packets is `60` (hex), which is equivalent to a DSCP value of `24`. To disable DSCP tagging, set the value to `0`.

> **Note:**
> The provisioning server reads this value and includes it in the phone configuration file.

Individual Managed IP Phones for Test Purposes

Create Individual Managed IP Phones for Test Purposes

Genesys recommends creating individual managed IP phones for the appropriate managed IP phone types in your CIC system for test purposes, before proceeding with a full managed IP phone implementation. Create at least one individual managed IP phone for each of the managed IP phone types that you plan to implement and configure for your test purposes.

To create individual managed IP phones

1. Open Interaction Administrator, right-click in the **Managed IP Phones** container, and then click **New**
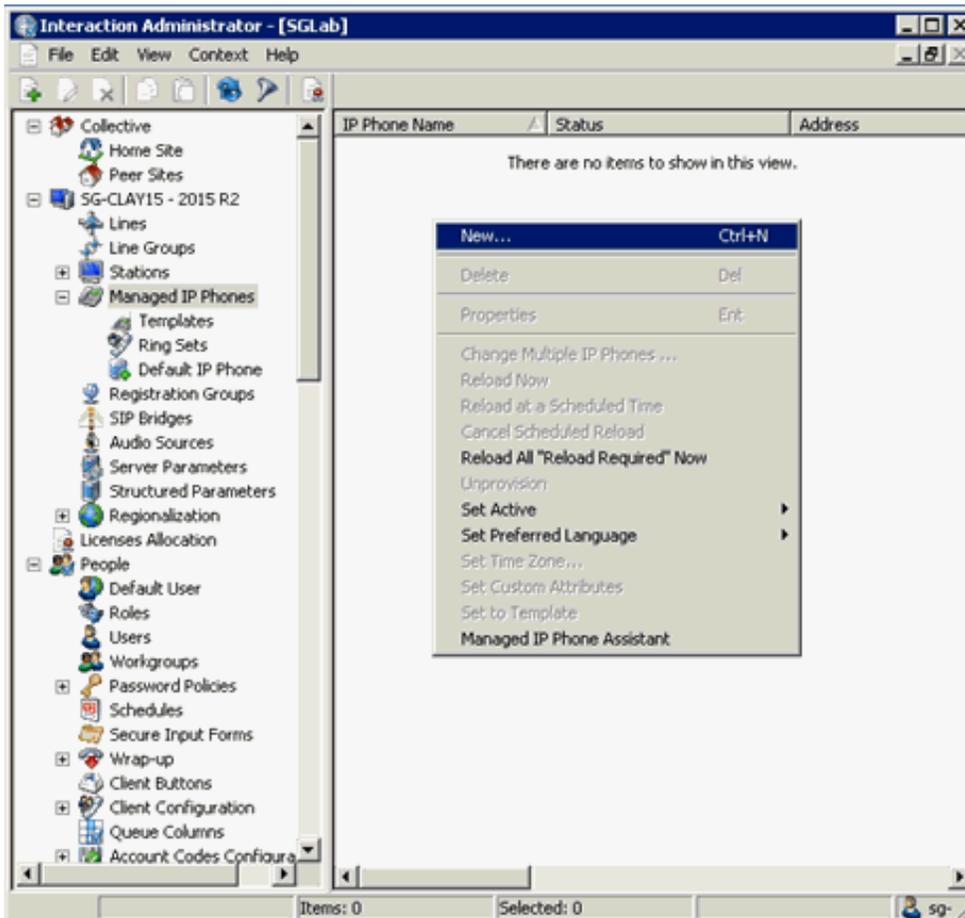
2. In the **New Managed IP Phone** dialog box, type the information for the first managed IP phone type to create and then click **OK**.
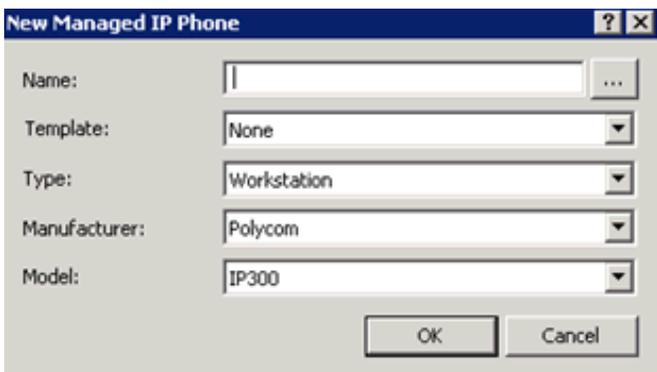
Name: Name of the new managed IP phone.

Template: Keep the default setting of **none**. (It is not necessary to create a managed IP phone template until you Create Multiple Managed IP Phones Procedure Summary.)

Type: Type of IP phone: **Workstation** or **Stand-alone phone**.

> **Note:**
> If you are creating a managed IP phone for Interaction SIP Station, do not select **Stand-alone phone**. The system doesn't support it.

Manufacturer: IP Phone manufacturer. The supported manufacturers are **AudioCodes**, **Genesys**, **ININ**, and **Polycom**. Select ININ for Interaction SIP Station I, Interaction SIP Station II, and SIP Soft Phone.

Model: Phone model based on the manufacturer. If the manufacturer is AudioCodes, Genesys, or Polycom, choose from a list of AudioCodes, Genesys, or Polycom phone models. If the manufacturer is ININ, choose Interaction SIP Station (for Interaction SIP Station I), Interaction SIP Station II or Soft Phone.

3. In the **Managed IP Phone Configuration** dialog box, type the information on the **General** tab to define for this managed IP phone type for test purposes.
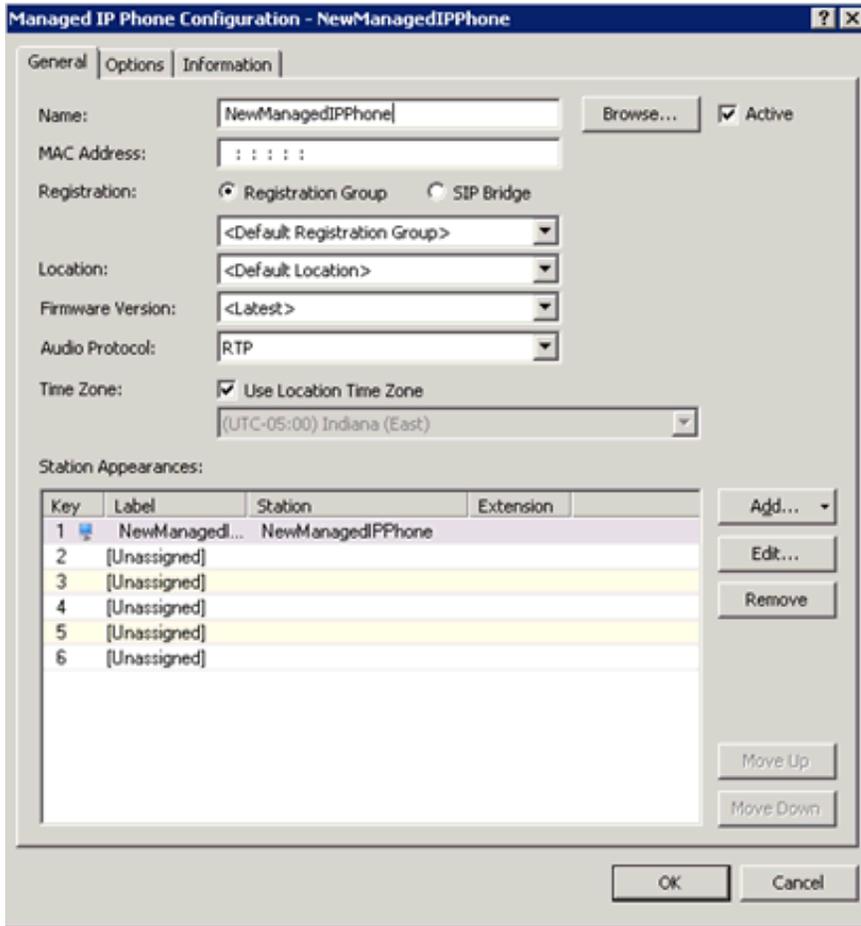
Name: Unique name of this IP phone.

Active: Keep the default setting of **Active** for this managed IP phone.

MAC Address (Polycom, Interaction SIP Station I and II, AudioCodes, Genesys): MAC Address in the following format: *xx.xx.xx.xx.xx.xx*. Polycom addresses start with `00:04:f2`. Interaction SIP Station MAC addresses start with `00:26:fd`. AudioCodes and Genesys addresses start with `00:90:8f`.

Full Computer Name: (SIP Soft Phone) Type the full computer name of the managed IP phone. The correct full computer name is on the SIP Soft Phone user's computer. To see the full computer name, click **My Computer** -> **Properties** -> **Computer Name**. For example: **PattyJ.acme.com** or **PattyJ**.

Registration: Select **Registration Group**. Depending on your implementation and what you plan to test, it may be necessary to modify the default registration group. For more information, see Registration Groups.

Location: Keep the default location unless you plan to test multiple locations. Besides reading SNTP settings from the phone, the CIC server uses the location settings to determine the codecs to use for the phone's audio during the call.

Firmware Version: Firmware version for this managed IP phone. By default, the recommended option of <Latest> is selected. Older firmware versions installed previously on this system can also appear in the list box. In certain scenarios, you can select older approved firmware for this phone model. For example, to control the rollout of new firmware to a managed IP phone or group of managed IP phones during a release update.

> **Note:**
> If the selected managed IP phone model does not support the selectable firmware feature, this option isn't available.

Audio Protocol (Polycom, SIP Soft Phone, Interaction SIP Station I and II, AudioCodes, Genesys): The system can unencrypt the audio stream on this IP phone using RTP (Real Time Protocol) or encrypt it using Secure RTP (SRTP). If you plan to use TLS/STRP, select **STRP**. Otherwise, leave the default setting of **RTP**. (Certain Polycom IP phone models do not support this audio protocol option; therefore, this option doesn't appear.)

Time Zone (Polycom, Interaction SIP Station I and II, AudioCodes, Genesys): The time zone that set here passes to the phone during configuration. This option overrides the setting that is in the phone's configuration.

Station Appearances: As part of creating managed IP phones, you create associated SIP station appearances on those IP phones. Proceed to step 4 to configure the station appearance for this managed IP phone.

4. Under **Station Appearances** on the **Managed IP Phones Configuration General** tab, click **Edit** to open the **Station Configuration** for the selected managed IP phone.

5. On the **Station Configuration-Configuration** tab, type the information to define for the station appearance for test purposes.

Extension: Extension number for this station appearance.

Changes to other **Configuration** tab settings are optional. For more information, see the *Interaction Administrator Help*.

6. Click the **Station Configuration-Licensing** tab.
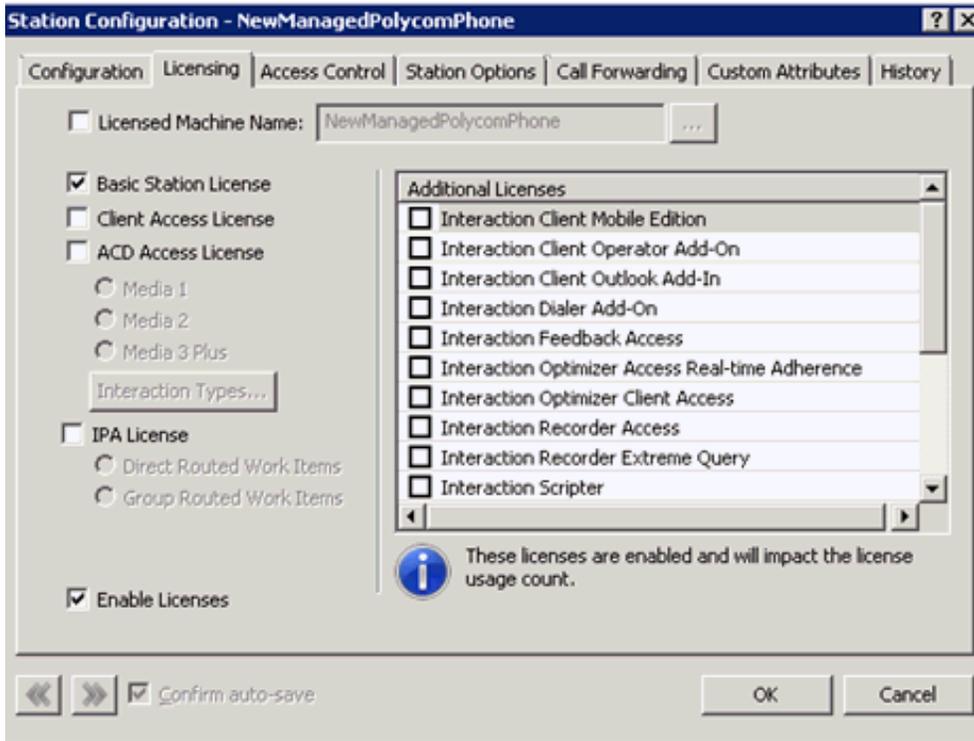
The system assigns a Basic Station License by default to the SIP station appearance associated to the managed IP phone. Assign other licenses, such as Client Access and ACD Access, as needed for test purposes.

7. Click **OK** to return to the **Managed IP Phone Configuration** dialog box.

8. To configure additional options for the managed IP phone for test purposes, see the **Managed IP Phone Configuration-Options** tab. The available options vary, depending on the managed IP phone type.
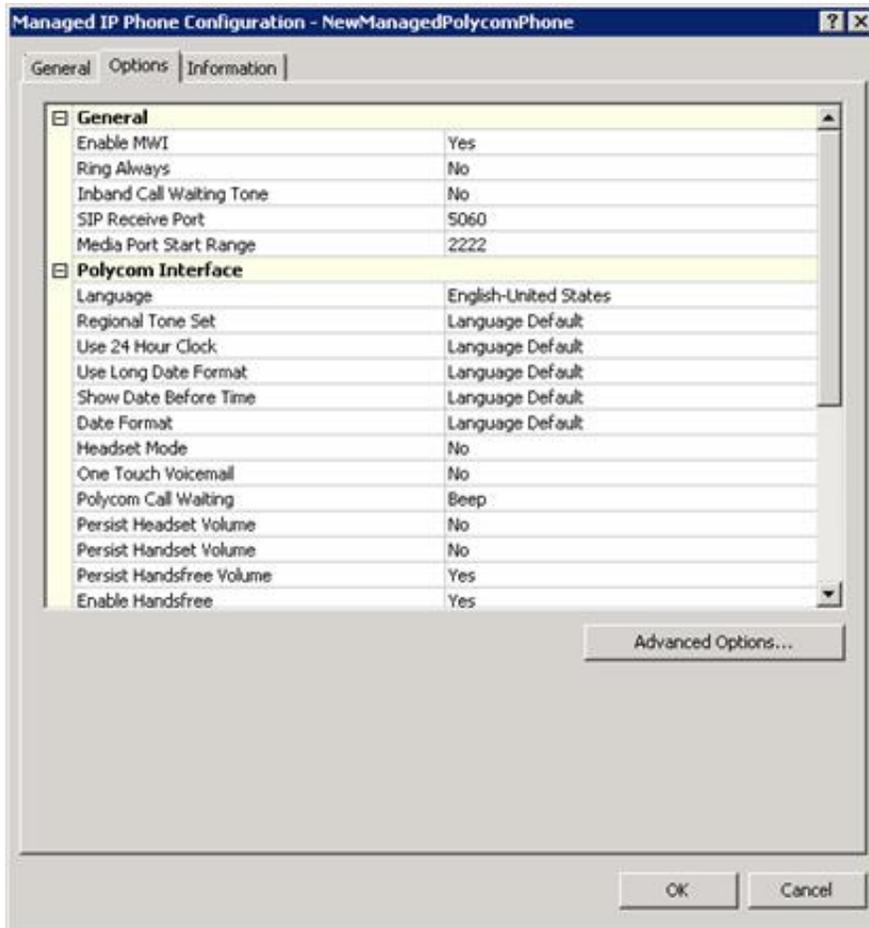
For more information about these Polycom configuration options, see the *Interaction Administrator Help*.

> **Note:**
> If you have not configured QoS for RTP and SIP as described in Implement QoS in Your Environment, do so now.

9. Repeat the previous steps to create a managed IP phone for the other managed IP phone types you plan to implement.

Registration Groups

Every managed IP phone requires a registration group. The registration group controls who the phone registers and communicates with. Each registration group consists of an ordered list of registrations. Each registration either points to a line, is specified manually, or is obtained from a SIP proxy or DNS SRV. A managed IP phone attempts to use the first registration. If it fails, it uses the second one. If it fails again, it attempts to use the subsequent registrations.

Depending on your implementation and what you plan to test, you might have to modify the registration group configuration for the individual managed IP phones that you create for test purposes.

- View Default Registration Group Settings
- Registration Group Configuration for TLS/STRP and/or Switchover

View Default Registration Group Settings

As part of the CIC installation, IC Setup Assistant created two permanent default registration groups:

- **<Default Registration Group>** for the default <Stations-UDP> line
- **<Default Secure Registration Group>** for the default <Stations-TLS> line

To view the <Default Registration Group> settings

1. In the Interaction Administrator **Managed IP Phones** container, click **Registration Groups**.
2. Right-click **<Default Registration Group>** and then click **Properties**.
3. In the **Registration Group Configuration** dialog box, note that only one registration appears under **Registrations**. You can add more registrations for redundancy.
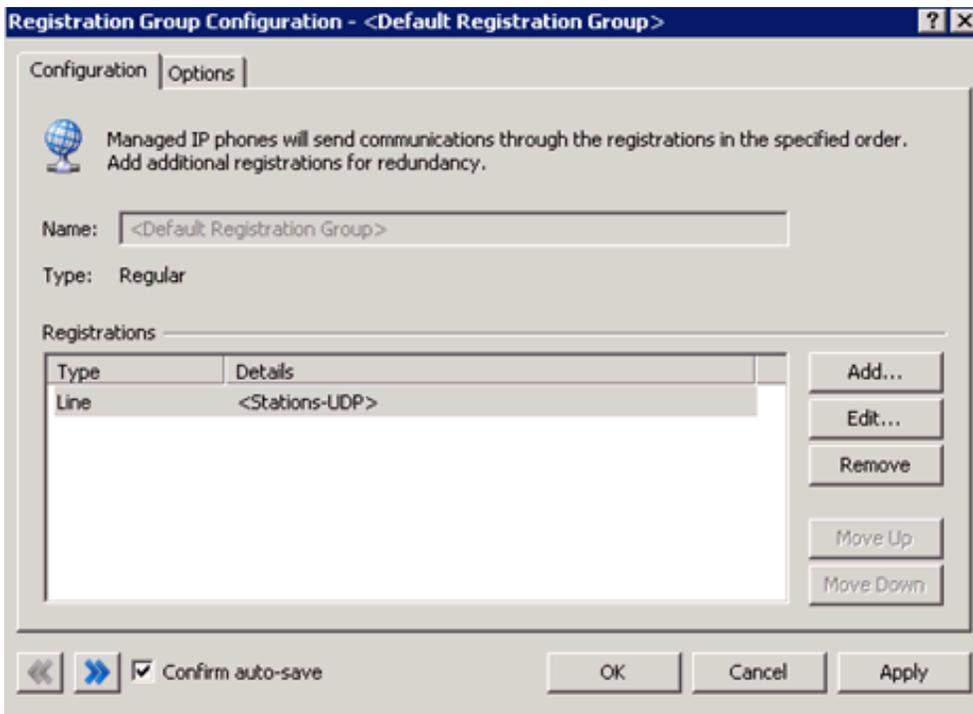
> **Note:**
> ○ A registration group cannot have more than one **SIP Line** registration. If a registration group does have a SIP Line registration, it must be the first entry in the list.
> ○ A registration group cannot have more than one **DNS SRV** registration. If a registration group does have a DNS SRV registration, it must be the first entry in the list.

4. Click the listed registration and then click **Edit** to view the registration settings.
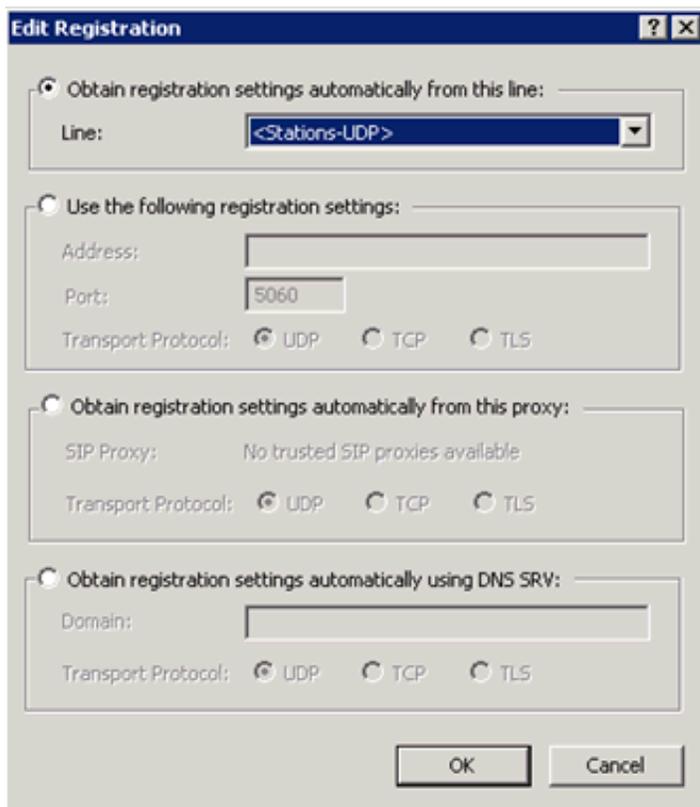
This registration is set to obtain registrations automatically from the default <Stations-UDP> line.

Depending on your implementation and what you plan to test, you may need to create additional registrations or modify the existing registration for the <Default Registration Group>.

Registration Group Configuration for TLS/STRP and/or Switchover

The following describes registration group configuration for Polycom, SIP Soft phones, Interaction SIP Station, AudioCodes phones, and Genesys phones in environments implementing TLS/STRTP and/or Switchover. TLS/STRTP support for Interaction SIP Station I and II, AudioCodes phones, and Genesys phones is available in CIC 2016 R4 and later.

Security settings

If you plan to use TLS/STRP:
- Ensure that the **Audio Protocol** is set to **SRTP** for the managed IP phone.
- Use the **<Default Secure Registration Group>** instead of the <Default Registration Group>.

Switchover settings (TLS/SRTP)

If you are configuring a Switchover environment and are using TLS/SRTP, modify the following for the **<Default Registration Group>** or **<Default Secure Registration Group>**:
- In the **Registration Group Configuration** dialog box, click **Edit** and then click **Obtain registration settings automatically using DNS SRV**. This option obtains registration settings from the DNS SRV records that you created when you configured the network the phone. Ensure that you select the appropriate **Transport Protocol** for this registration group.
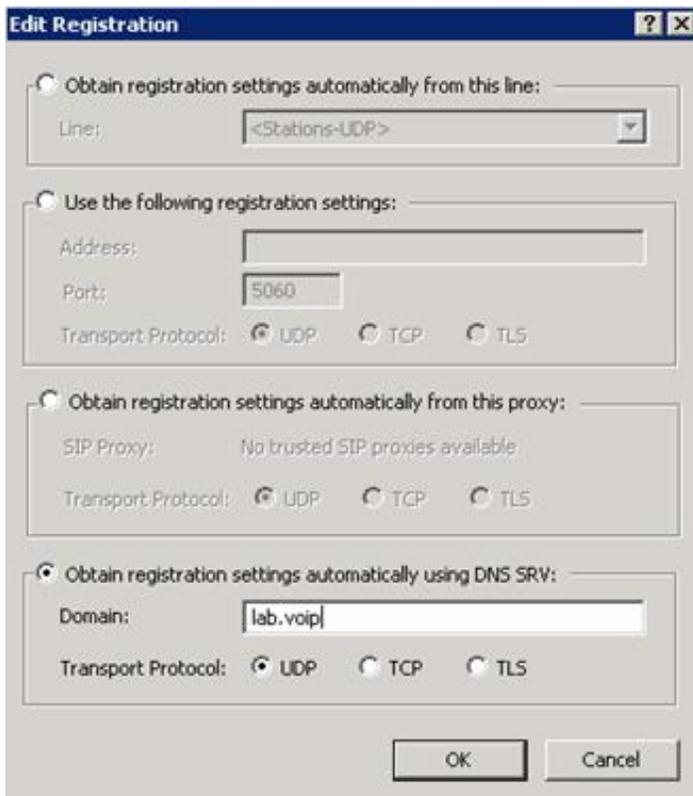
- If your Switchover environment requires remote survivability (for example, a remote office using a SIP Proxy), click **Add…** in the **Registration Group Configuration** dialog box to add another registration to the <Default Registration Group>. In the **Add Registration** dialog box, click **Obtain registrations settings automatically from this proxy**. Ensure that you select the appropriate **Transport Protocol** for this registration group.

Interaction SIP Station I and II, AudioCodes, and Genesys phone support for DNS SRV

- Interaction SIP Station I supports DNS SRV in CIC 4.0 SU 4 to SU 6 and CIC 2015 R1 and later, which contains v.1.2.2._p10_build_17 or later firmware.
- Interaction SIP Station I and II support DNS SRV in CIC 2015 R2 and later which contains v.2.0.4.15.7 or later firmware.
- AudioCodes and Genesys phones support DNS SRV in CIC 4.0 SU 6 and CIC 2015 R1 and later, which contains v2.2.277.2.3 or later firmware.

> **Note:**
> Interaction SIP Station I and II, AudioCodes, and Genesys phones support only a single registration entry: either DNS SRV or a stations line.

Limitations

Interaction SIP Station I and II, AudioCodes, and Genesys phones have the following registration group configuration limitations:
- Interaction SIP Station I and II, AudioCodes, and Genesys phones do *not* support multiple entries (registrations) in a registration group.

> **Note:**
> Starting with CIC 2016 R1, firmware updates with a version of 2.2.2.77 or higher for Interaction SIP Station I and II remove the constraint of registering the phone with a single server.

Switchover settings (SIP Station I and II, AudioCodes, or Genesys phones)

Switchover environments that use only Interaction SIP Station I and II, AudioCodes, or Genesys phones do not need to modify the <Default Registration Group>.

However, **if the Switchover environment includes Interaction SIP Station I and II, AudioCodes, Genesys, Polycom, or**

SIP Soft phones, follow this example to **create separate registration groups specifically for the Interaction SIP Station I and II, AudioCodes, and/or Genesys phones**.

To create separate registration groups

1. In **Managed IP Phones > Registration Groups**, create a new Regular registration group for the Interaction SIP Stations.

2. In the **Registration Group Configuration** dialog box, click **Add**

3. In the **Add Registration** dialog box, click either **obtain registration settings automatically from this line** to obtain registration settings from the <Stations-UDP> line or **obtain registration settings automatically using DNS SRV**.
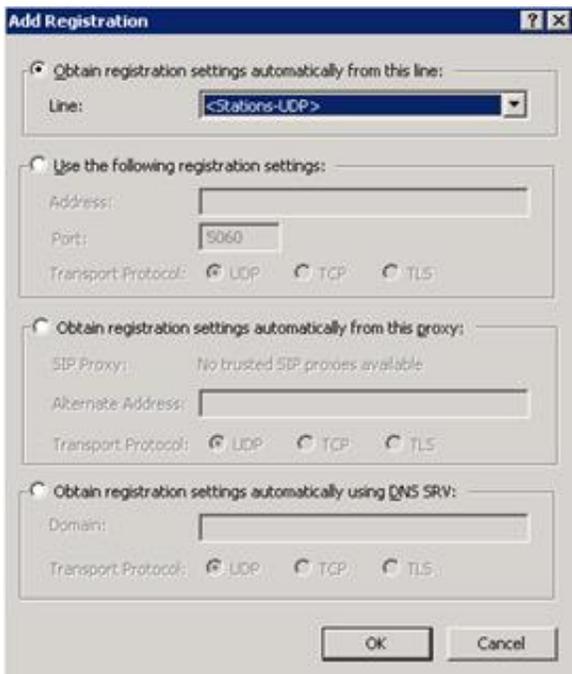
Create Multiple Managed IP Phones Procedure Summary

After you configure the network for managed IP phones, create individual managed IP phones for test purposes, and are satisfied with your testing, you can proceed with the full managed IP phone implementation. The implementation includes creating multiple managed IP phones for all IP phone types (Polycom, Interaction SIP Station I and II, SIP Soft Phone, AudioCodes, and Genesys phones) at your site.

> **Note:**
> You can migrate or recreate existing unmanaged Polycom phones and SIP Soft Phones as managed IP phones.

The recommended procedure for creating new managed IP phones is to use the **Managed IP Phone Assistant** in Interaction Administrator to create multiple managed IP phones and associated SIP stations. The procedure consists of the following steps:

1. Create one or more **managed IP phone templates** based on managed IP phone type, manufacturer, model, location, language, audio protocol, station appearance, and more. For more information, see Create Managed IP Phone Templates.
2. Create one or more **CSV Managed IP Phone Lists based on 1) Template** containing name, template, proxy group, extension, identification address, label, and address information for the appropriate IP phones in your CIC system, **and/or 2) Type, Manufacturer, and Model** containing name, type, model, manufacturer, proxy group, extension, identification address, label, and address information for the appropriate IP phones in your CIC system. For more information, see Create CSV Managed IP Phone Lists.
3. Run the **Managed IP Phone Assistant** in Interaction Administrator to create new managed IP phones and associated SIP stations by importing the CSV Managed IP Phone list(s). For more information, see Create Managed IP Phones with Managed IP Phone Assistant.

Create Managed IP Phone Templates

Genesys recommends that you plan the IP phone configuration that you want for your CIC system.

If it is appropriate for your site, create one or more managed IP phone templates based on your planning decisions. For example, you might want to create separate templates for the following situations:

- The CIC system has Polycom phones (perhaps a variety of Polycom models), SIP Soft Phones, Interaction SIP Station I and II phones, AudioCodes, and Genesys phones.
- The CIC system has IP phones in a variety of locations.
- The audio stream on certain IP phones unencrypts using RTP, and others encrypt using SRTP.
- Certain IP phones have regular station appearances, and others have shared station appearances.

To create managed IP phone templates

1. In the Interaction Administrator **Managed IP Phones** container, click **Templates**, right-click in the right pane, and then click **New**.
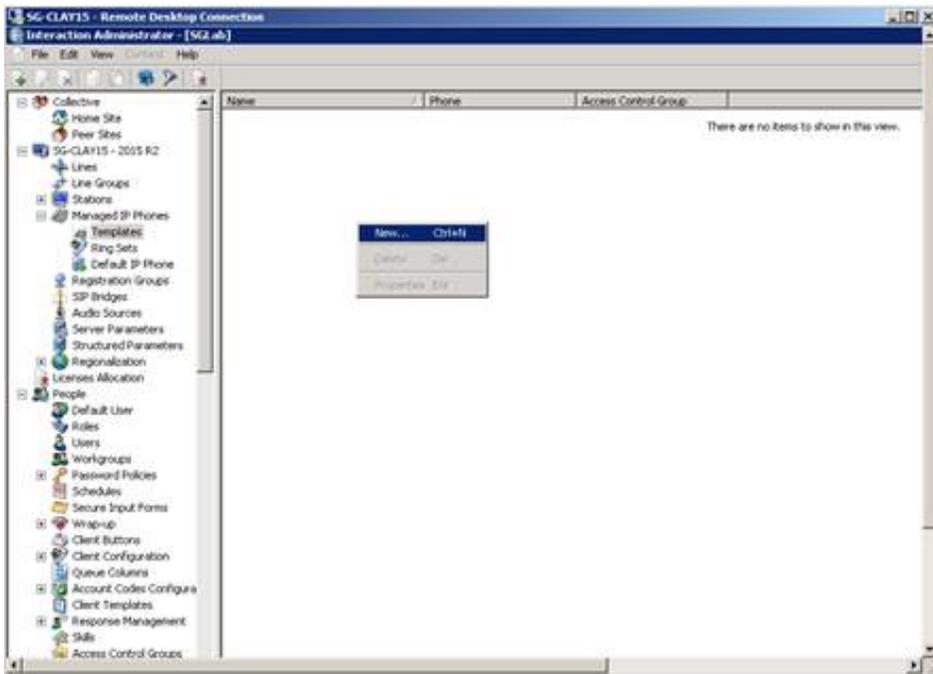
The **New Managed IP Phone Template** dialog box appears.

Name: Name of the new managed IP phone template.

Type: Type of IP phone for this template **Workstation** or **Stand-alone Phone**. The default is **Workstation**.

Manufacturer: IP Phone manufacturer. The supported manufacturers are **AudioCodes**, **Genesys**, **ININ**, and **Polycom**. Click **ININ** for Interaction SIP Station I and II and SIP Soft Phone.

Model: Phone model based on the manufacturer. If the manufacturer is AudioCodes, Genesys, or Polycom, choose from a list of AudioCodes, Genesys, or Polycom phone models. If the manufacturer is ININ, choose Interaction SIP Station (for Interaction SIP Station I), Interaction SIP Station II, or Soft Phone.

2. Complete the information for the first managed IP phone template to create and then click **OK**. The **Managed IP Phone Template Configuration** dialog box appears.
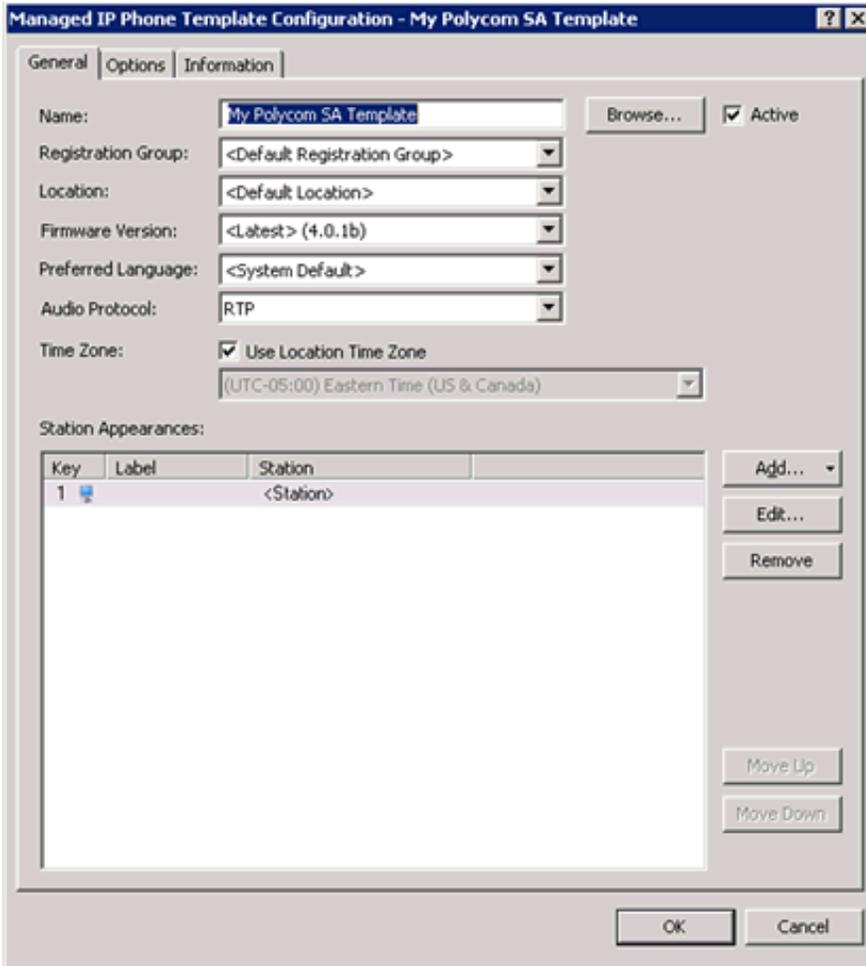
3. Select or add the appropriate configuration items on the **General** and **Options** (including **Advanced Options** for Polycom, Interaction SIP Station, AudioCodes,and Genesys phones) tabs *for this managed IP phone template*.

4. When completed, click **OK**.

    For a summary of advanced configuration features for each IP phone type, see the following:

- Managed IP Phone (Polycom) Configuration Options
- Managed IP Phone (Interaction SIP Station I and II) Configuration Options
- Managed IP Phone (AudioCodes and Genesys Phones) Configuration Options

    You can also click Help (?) for details on individual configuration items.

    Genesys recommends that you specify as many configuration items as possible, instead of using the default settings. Defining the template now saves you time later.

    The managed IP phone template appears in **Managed IP Phones…Templates**.
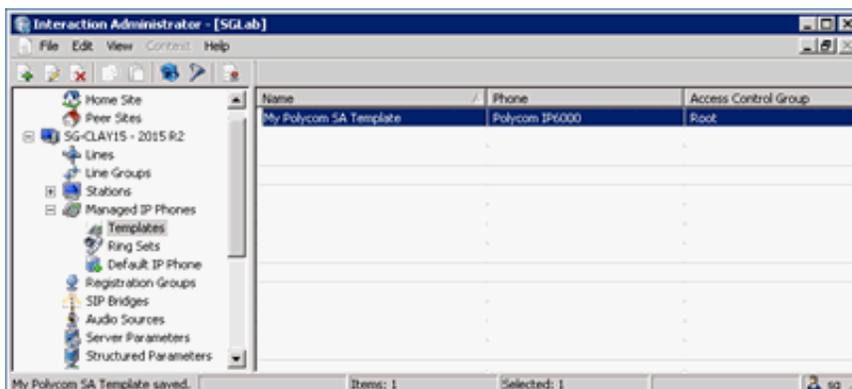
5. Repeat the previous steps for the other managed IP phone templates you want to create. For example:
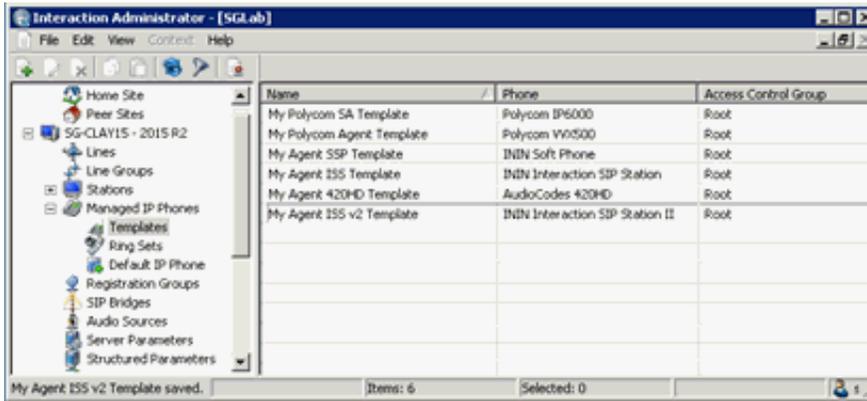
Create CSV Managed IP Phone Lists

Once you create a managed IP phone template, create one or more **CSV Managed IP Phone lists** based on one or both of the following:

- **Template:** Containing name, template, proxy group, extension, identification address, label, and address information for the appropriate IP phones in your CIC system. Each IP phone must reference one of the managed IP phone templates you created by name. For more information, see Create a CSV Managed IP Phone Template List.
- **Type, Manufacturer, and Model** containing name, type, manufacturer, model, proxy group, extension, identification address, label, and address information for the appropriate IP phones in your CIC system. For more information, see Create a CSV Managed IP Phone TMM List.

Two sample CSV Managed IP Phone lists corresponding to the two types of CSV lists are available to download from the Utilities and Downloads page:

- `CSV Managed IP Phone List-Template.csv` and `CSV Managed IP Phone List-Template.xlsx`
- `CSV Managed IP Phone List-TMM.csv` and `CSV Managed IP Phone List-TMM.xlsx`

When you run Managed IP Phone Assistant, you import a completed CSV Managed IP Phone list.

Create a CSV Managed IP Phone Template List

Follow this procedure to create a CSV managed IP phone list based on a template.

To create a CSV Managed IP Phone-Template list

1. **Download the most recent sample CSV Managed IP Phone Lists** from the Utilities and Downloads page to the CIC server or location accessible by the CIC server.
2. Open a **copy of** `CSV Managed IP Phone List-Template.xlsx` and specify the attributes in the appropriate columns for each managed IP phone and associated SIP station to create.

The following table provides descriptions of each attribute.

Attribute	Description
Name	(Required) Name of the IP phone.
Template	(Required) Managed IP phone template for this IP phone. Ensure that the template name is identical to one of the templates that you created.
Proxy Group	Registration (Proxy) Group to use with the managed IP phone template for this IP phone. If this value is blank, Managed IP Phone Assistant fills in the Registration Group defined in the template.
Extension	Primary appearance extension number for this IP phone. If this value is blank, it remains blank in Managed IP Phone Assistant.
Identification Address	SIP connection address in the form of `sip: xxx@ IPaddress: portnumber` . For example, `sip:320@172.17.238.68:5060`. If you do not know the IP address, leave this value blank. It populates when the phone registers with the CIC server following provisioning.
Label	(Polycom only) Label to use for the primary appearance of this IP phone and the associated SIP station. Typical values for "label" are the station extension or the user's extension (in the case where one user almost always uses the station). If this value is left blank, Managed IP Phone Assistant fills in the **Name** attribute (IP phone name).
Address	If you know the address for this IP phone ahead of time, specify one of the following: ○ **For Polycom phones:** MAC address of the IP phone. Polycom addresses start with `00:04:f2`. ○ **For SIP Soft Phones:** Full computer name for the IP phone. To ensure that you get the full computer name, click **My Computer>Properties>Computer Name**. For example: `PattyJ.acme.com`. ○ **For Interaction SIP Station I & II phones:** MAC address of the IP phone. Interaction SIP Station MAC addresses start with `00.26.fd`. ○ **For AudioCodes and Genesys phones:** MAC address of the IP phone. AudioCodes and Genesys phone addresses start with `00:90:8f`. If you do not know the address ahead of time: ○ **For Polycom phones:** Provision the phones manually using the provisioning IVR. ○ **For SIP Soft Phones:** Provision the phones manually using the SIP Soft Phone Provisioning wizard. ○ **For Interaction SIP Station I phones:** You cannot provision the Interaction SIP Station I through the provisioning IVR. If you didn't know the MAC address when you created the CSV list, specify the MAC address manually for each Interaction SIP Station I in the Interaction Administrator Managed IP Phone container before setting up and booting the corresponding phone. ○ **For Interaction SIP Station II phones:** Provision the phones manually using the provisioning IVR. ○ **For AudioCodes and Genesys phones:** Provision the phones manually using the provisioning IVR.

3. After specifying the IP phone information, **save the file**, selecting **.CSV** as the file type. The following message can appear when saving the CSV Managed IP Phone list:

Ignore this message and click **Yes**. The message explains that some of the macros in the Excel file cannot be preserved in the resulting `CSV` file. You don't need those macros when Managed IP Phone Assistant imports the file.

Following is an example of the resulting `CSV Managed IP Phone List.csv` file as it appears in Excel:

When opening the `CSV` file in a text editor, the IP phone information is separated by commas, with one IP phone listed per line.

```
CSV Managed IP Phone List-Template.csv - Notepad
File  Edit  Format  View  Help
Name,Template,Proxy Group,Extension,Identification Address,Label,Address
PolycomLobby1,My Polycom SA Template,,7101,,Lobby,0004F2AABBCC
PolycomAgent1,My Polycom Agent Template,,7201,,Line1,
AgentSSP-PC1,My Agent SSP Template,,7301,,,AgentPC1.lab.local
AgentISS1,My Agent ISS Template,,7401,,,0026FD0095FD
Agent420HD,My Agent 420HD Template,,7501,,,00908F123456
AgentISS2,My Agent ISS v2 Template,,7601,,Line2,00908F5635BB
```
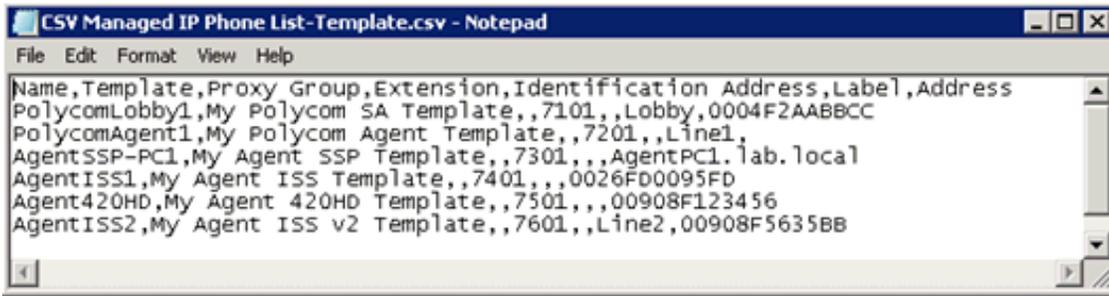
4. If you have not already done so, download your `CSV Managed IP Phone list-Template.csv` file to a secure location on the CIC server.

Create a CSV Managed IP Phone TMM List

Complete this procedure to create a CSV managed IP phone list based on type, manufacturer, and model.

To create a CSV Managed IP Phone-TMM list

1. **Download the most recent sample CSV Managed IP Phone Lists** from the <u>Utilities and Downloads page</u> to the CIC server or location accessible by the CIC server.

2. Open a **copy of** `CSV Managed IP Phone List-TMM.xlsx` and specify the attributes in the appropriate columns for each managed IP phone and associated SIP station to create.

	A	B	C	D	E	F	G	H	I
1	Name	Type	Manufacturer	Model	Proxy Group	Extension	Identification Address	Label	Address
2	PolycomLobby2	Stand-alone Phone	Polycom	IP330		7102		Lobby	
3	PolycomAgent2	Workstation	Polycom	IP601		7202		Line1	0004F204F6A9
4	AgentSSP-PC2	Workstation	ININ	Soft Phone	Remote Agent SSP - TLS	7302	7302		
5	AgentISS1	Stand-alone Phone	ININ	Interaction SIP Station	Remote Agent ISS - TCP	7402	7402		0026FDF00020
6	Agent420HD	Stand-alone Phone	AudioCodes	420HD		7502	7502		00908F123456
7	AgentISS2	Stand-alone Phone	ININ	Interaction SIP Station II	Remote Agent ISS - TCP	7602	7602	Line2	00908F5635BB

CSV Managed IP Phone List-TMM   ModelInformation

The following table provides descriptions of each attribute.

Attribute	Description
Name	(Required) Name of the IP phone.
Type	(Required) Type of IP phone - Workstation or Stand-alone Phone. **For Interaction SIP Station I and II phones**, specify `Workstation` as these types don't support stand-alone phones.
Manufacturer	(Required) IP phone manufacturer. The supported manufacturers are **AudioCodes**, **Genesys**, **ININ**, and **Polycom**.
Model	(Required) Phone model based on the manufacturer. **If the manufacturer is Polycom**, specify Polycom phone model. **If the manufacturer is ININ**, specify Soft Phone or Interaction SIP Station (for Interaction SIP Station 1), Interaction SIP Station II. **If the manufacturer is AudioCodes**, specify the AudioCodes phone model. **If the manufacturer is Genesys**, specify the Genesys phone model.
Proxy Group	Registration (Proxy) Group to use with the managed IP phone template for this IP phone. If this value is blank, Managed IP Phone Assistant fills in the default Registration Group.
Extension	Primary appearance extension number for this IP phone. If this value is blank, it remains blank in Managed IP Phone Assistant.
Identification Address	SIP connection address in the form of `sip:xxx@IPaddress:portnumber`. For example, `sip:320@172.17.238.68:5060`. If you do not know the IP address for this phone, leave this value blank. It populates when the phone registers with the CIC server following provisioning.
Label	(Polycom only) Label to use for the primary appearance of this IP phone and the associated SIP station. Typical values for **Label** are the station extension or the user's extension (in the case where one user almost always uses the station). If this value is blank, Managed IP Phone Assistant fills in the **Name** attribute (IP phone name).
Address	If you know the address for this IP phone ahead of time, specify one of the following: ○ **For Polycom phones:** MAC address of the IP phone. Polycom addresses start with `00:04:f2`. ○ **For SIP Soft Phones:** Full computer name for the IP phone. To ensure that you get the full computer name, click **My Computer>Properties>Computer Name**. For example: `PattyJ.acme.com`. ○ **For Interaction SIP Station I and II phones:** MAC address of the IP phone. Interaction SIP Station MAC addresses start with `00.26.fd`. ○ **For AudioCodes or Genesys phones:** MAC address of the IP phone. AudioCodes and Genesys phone addresses start with `00:90:8f`. If you do not know the address ahead of time: ○ **For Polycom phones:** Provision the phones manually using the provisioning IVR. ○ **For SIP Soft Phones:** Provision the phones manually using the SIP Soft Phone Provisioning wizard. ○ **For Interaction SIP Station I phones:** You cannot provision the Interaction SIP Station I through the provisioning IVR. If you didn't know the MAC address when you created the CSV list, specify the MAC address manually for each Interaction SIP Station I in the Interaction Administrator Managed IP Phone container before setting up and booting the corresponding phone. ○ **For Interaction SIP Station II phones:** Provision the phones manually using the provisioning IVR. ○ **For AudioCodes and Genesys phones:** Provision the phones manually using the provisioning IVR.

3. After you complete the IP phone information, save the file, selecting **.CSV** as the file type. The following message can appear when saving the CSV Managed IP Phone list:

Ignore this message and click **Yes**. The message explains that some of the macros in the Excel file cannot be preserved in the resulting `CSV` file. You don't need those macros when Managed IP Phone Assistant imports the file.

Following is an example of the resulting `CSV Managed IP Phone List.csv` file as it appears in Excel:

When opening the `CSV` file in a text editor, the IP phone information is separated by commas, with one IP phone listed per line.

4. If you have not already done so, download your `CSV Managed IP Phone list-TMM.csv` file to a secure location on the CIC server.

Create Managed IP Phones with Managed IP Phone Assistant

The Managed IP Phone Assistant guides you through the process of creating multiple managed IP phones and associated SIP stations from an imported CSV Managed IP Phone list. For more information, see the *Interaction Administrator Help*.

> **Note:**
> Genesys recommends that you run the Managed IP Phone Assistant outside of your core business hours because the procedure requires significant server resources.

To create managed IP phones with Managed IP Phone Assistant

1. In the **Managed IP Phones** container in Interaction Administrator, right-click in the right pane and then click **Managed IP Phone Assistant**.
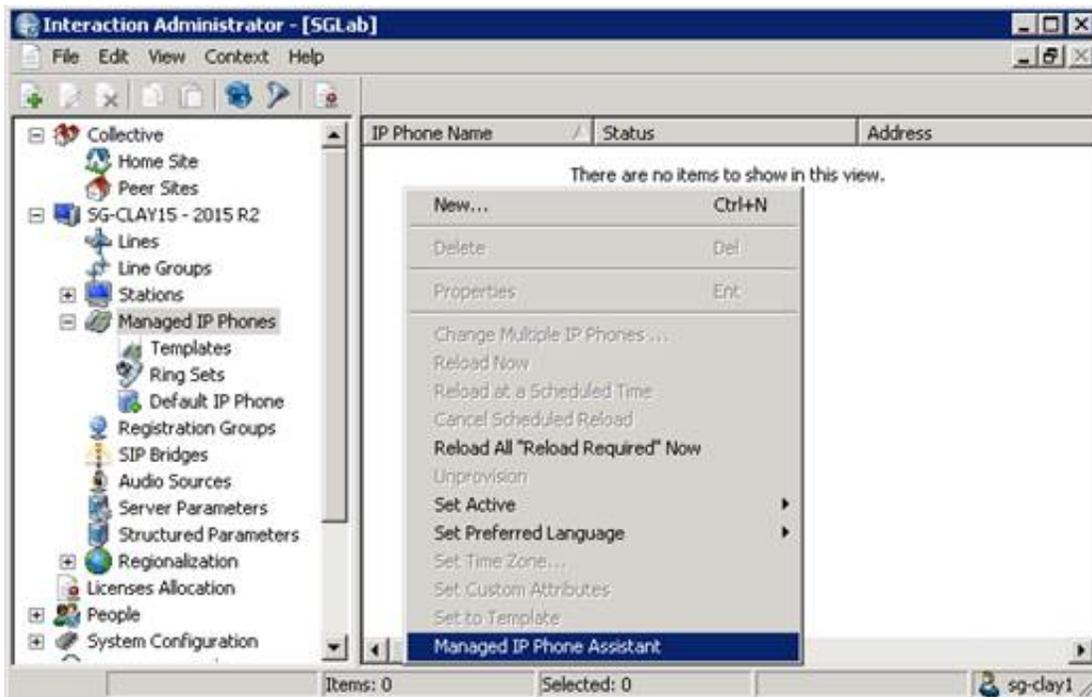
The **Managed IP Phone Assistant** Welcome page appears.

If you are running Managed IP Phone Assistant after business hours, a confirmation message appears, asking whether you want to run the wizard during business hours. Click **Yes**.

2. Click **Next**. The **Add Managed IP Phones** page appears.

3. Click **Create managed IP phones from a .CSV file**. The **Create Managed IP Phones from a CSV file** page appears.

4. **Browse** to the location of your `CSV Managed IP Phone List.csv` file(s) on the CIC server, select the `.csv` file to import, and then click **Next**.

If the assistant encounters errors while parsing the `CSV` file, a message appears, indicating such. Click **Errors** to view the status of the errors and a description.

**Warning** error: Managed IP Phone Assistant cannot verify one or more values. You can continue with the import, but some of those values don't import.

Severe error: Managed IP Phone Assistant detects no columns or it couldn't open the file. You cannot continue with the import.

If you imported a CSV list based on type, manufacturer, and model (`CSV Managed IP Phone List-TMM.csv`), the **Access Control** page appears.
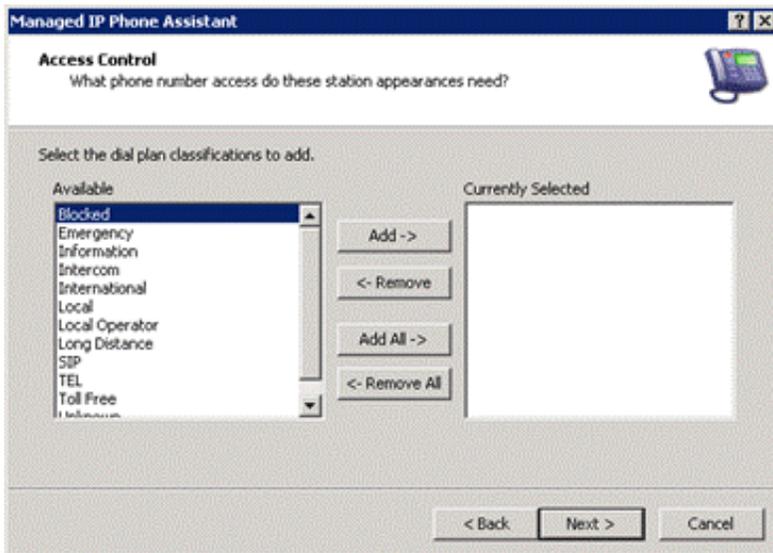
5. Select the dial plan classifications for the managed IP phones and station appearances in the `CSV` file. For more information, click Help (**?**).

> **Note:**
> If you imported a CSV list based on template, this page doesn't appear because you defined the dial plan classifications in the template.

6. Click **Next**. The **Saving Managed IP Phones** page appears. To change your settings, click **Back**.
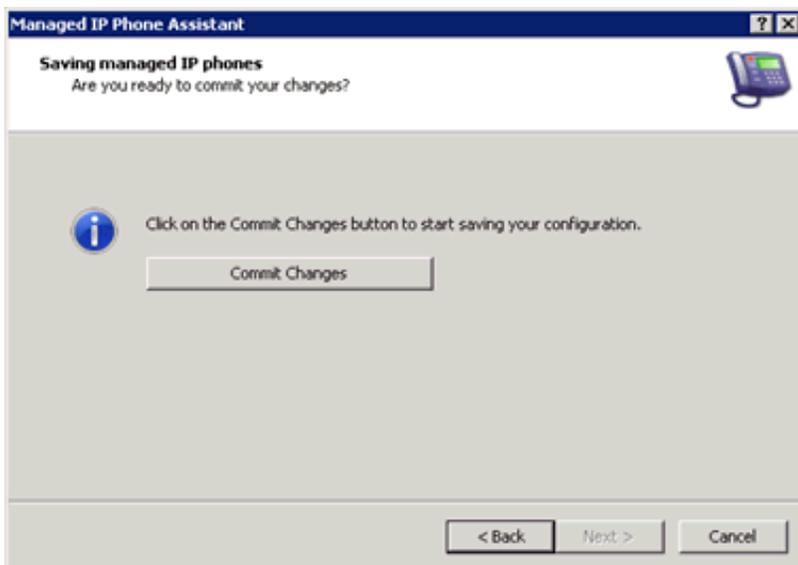
7. Click **Commit Changes**. If you imported a CSV list based on type, manufacturer, and model (`CSV Managed IP Phone List-TMM.csv`), the **Station Appearances Licenses** page appears.
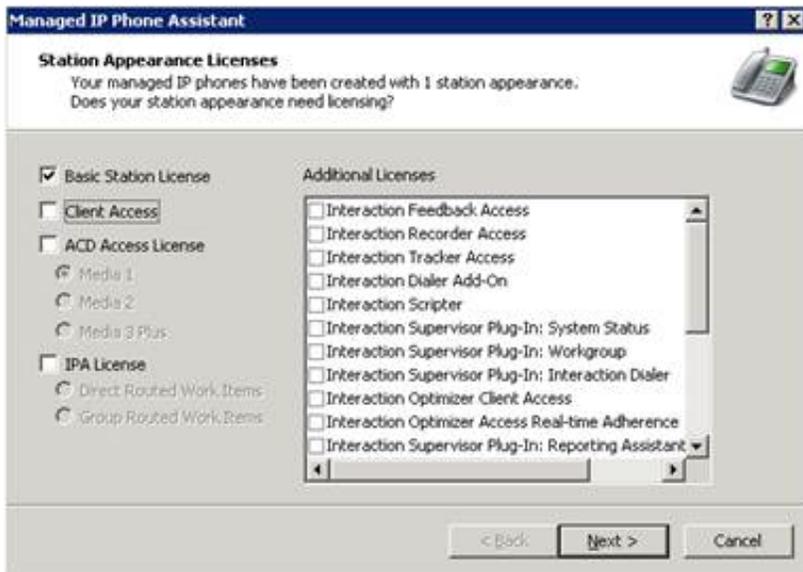
A Basic Station license is assigned by default to SIP stations associated with managed IP phones. If necessary, assign Client Access licenses, ACD Access licenses, and/or add-on licenses. For more information, click Help (**?**).

> **Note:**
> If you imported a CSV list based on template, this page doesn't appear because you defined the station appearance licensing in the template.

After you click **Commit Changes**, the assistant prepares the managed IP phones and applies licenses.

If you exceed the total count of licenses, a message appears: "Some or all licenses could not be allocated to each station appearance". Click **Review** to review the errors and then correct the information in the Interaction Administrator **License Allocation** container.

The **Completed the Managed IP Phone Assistant** page appears.

8. Click **Finish**. When the Managed IP Phone Assistant completes, the new managed IP phones appear in the **Managed IP Phones** container.
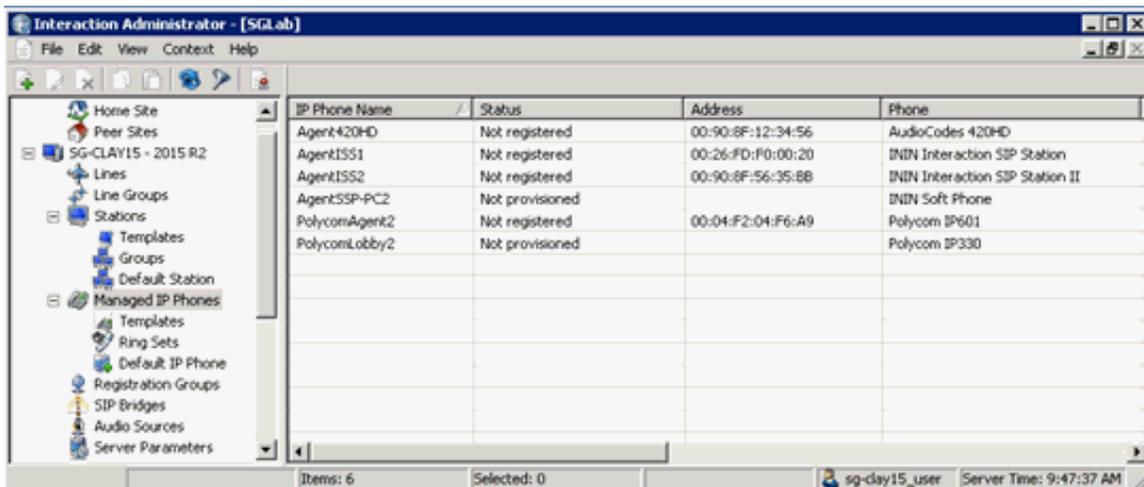
- ○ The new managed IP phones have a **Status** of **Not registered**. Their status changes to **Up-to-date** on the phones' next SIP registration.

- ○ If you did not specify the **Address** attribute for one or more managed IP phones in your CSV Managed IP Phone list, they have a status of **Not provisioned**. Their status changes to **Up-to-date** once you provision them using the Polycom phone, SIP Soft Phone, Interaction SIP Station II, AudioCodes, or Genesys phones provisioning IVR. For Interaction SIP Station I, you provision them by specifying the MAC address for each Interaction SIP Station in Interaction Administrator before setting up and booting the corresponding phone.

As part of creating managed IP phones, the associated SIP station appearances created on those IP phones appear in the **Stations** container.

The SIP stations associated with managed IP phones are of the type **Managed Workstation** or **Managed Stand-alone Phone**.

If necessary, you can assign Client Access licenses, ACD Access licenses, and add-on licenses to users on the **Licensing** tab now, or in the **Licenses Allocation** container.

Complete the Provisioning Process (Multiple Managed IP Phones)

CIC provisioning of managed IP phones connects the managed IP phones and downloads their configurations from the CIC server. The goal for CIC Managed IP Phone provisioning is to "automate" it as much as possible so that it requires no user participation.

If you followed the instructions to configure the network and to Create Managed IP Phones with Managed IP Phone Assistant, **automated provisioning occurs once you set up and boot (or re-boot)the phones**.

When a managed IP phone contacts the provisioning subsystem, the provisioning system matches the phone's computer name or MAC address with an existing managed IP phone configuration, and serves the configuration to the device without any extra steps.

Manual provisioning (multiple managed IP phones)

Manual provisioning of some or all managed IP phones are necessary when:
- You don't specify the MAC Address (Polycom, Interaction SIP Station I and II, AudioCodes, Genesys phones) or Full Computer Name (SIP Soft Phone) in the CSV list for some or all Polycom, Interaction SIP Station I and II, SIP Soft Phones, AudioCodes, and/or Genesys phones.

   > **Note:**
   > In an implementation of new Polycom and/or SIP Soft Phones, the MAC Address or Full Computer Name may not be known at the time the CSV list is created.

- The CIC system has no DHCP server or it cannot access the DHCP server.
- Some Polycom phones, Interaction SIP Station phones I and II, SIP Soft Phones, AudioCodes, and/or Genesys phones are in remote locations.

Typically, an administrator with required permissions does the manual provisioning. In some cases (for example, remote locations), users must do the manual provisioning.

To provision each managed phone's configuration manually
- **For Polycom phones**: Use the provisioning IVR.
- **For SIP Soft Phones**: Use the SIP Soft Phone Provisioning wizard.
- **For Interaction SIP Station I phones**: You cannot provision the Interaction SIP Station I phones through the provisioning IVR because they don't have a dial pad. If you didn't know the MAC address when you created the CSV list, specify the MAC address manually for each Interaction SIP Station in the Interaction Administrator Managed IP Phone container before setting up and booting the corresponding phone.
- **For Interaction SIP Station II phones**: Use the provisioning IVR since Interaction SIP Station II phones do have a dial pad.
- **For AudioCodes and Genesys phones**: Use the provisioning IVR.

Sometimes, you have to use non-standard manual provisioning procedures. For more information, see Appendix A: Non-standard Provisioning Scenarios.

Polycom Administration

Create Managed IP Phones from Existing (Polycom) SIP Stations

> **Note:**
> This information is for existing CIC systems with unmanaged Polycom phones migrating to CIC 2015 R1 or later.

If your CIC system has existing "unmanaged" Polycom phones (SIP stations), Genesys recommends that you convert them to managed IP phones. Then, you no longer need to maintain Polycom phone configuration (`.cfg`) files.

Before beginning these procedures, configure the network for managed IP phones. Genesys also recommends that you Create Individual Managed IP Phones for Test Purposes and ensure that you are satisfied with your testing first.

Add the SIP Phone Information Update server parameter

The SIP Phone Information Update server parameter allows the CIC system to populate the Manufacturer and Model automatically for a Polycom phone (SIP station) in Interaction Administrator. Set the SIP Phone Information Update server parameter to a value of **1**, **yes**, or **true**. When the phones reboot, Telephony Services derives the model and manufacturer information from the User-Agent strings in the registration message and updates the DS attributes dynamically for the corresponding SIP Station. All Polycom phones send this information each time they register, meaning that the data populates in Interaction Administrator as each phone re-registers with the CIC server. The system uses this data when you run Managed IP Phone Assistant to convert to managed IP phones and associated SIP stations.

For the **SIP Phone Information Update** server parameter to take effect, reboot the phones. It takes time for all of the phones to re-register with the server, so ensure that you check to see that the values populated before proceeding.

Update a common .cfg file to include sec.tagSerialNo="1" configuration parameter

Update the Polycom phone configuration files in the root directory of the TFTP or FTP server to include the **sec.tagSerialNo="1"** configuration parameter. This action allows the CIC system to populate the **MAC Address** automatically for a Polycom phone (SIP station) in Interaction Administrator. Choose a common file such as `xic.cfg` (when using files that the ININ IP Phone Configurator generated), or the `sip.cfg` file. When you update the `.cfg` file and reboot the phones, Telephony Services derives the MAC address from the User-Agent strings in the registration message and update the configuration dynamically for the corresponding SIP Station. All Polycom phones send this information each time they register, meaning that the data populates in Interaction Administrator as each phone re-registers with the CIC server. The system uses this data when you run Managed IP Phone Assistant to convert to managed IP phones and associated SIP stations.

For the `.cfg` file updates to take effect, reboot the phones. It takes time for all of the phones to re-register with the server, so ensure that you check to see that the values populated before proceeding.

Create managed IP phones from existing SIP stations using Managed IP Phone Assistant

Use the Migrate option in the Managed IP Phone Assistant to create managed IP phones and associated SIP stations from existing (Polycom phone) SIP stations. In this procedure, a sample SIP Workstation IP335 station migrates to a new managed IP phone and associated station.
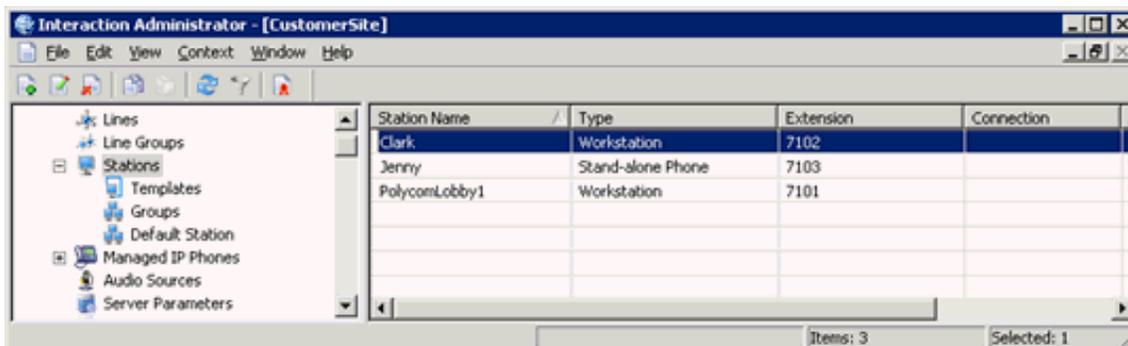
Notes:
- Genesys recommends that you run the Managed IP Phone Assistant outside of your core business hours because the procedure requires significant server resources.
- The complexity and variation of XML files means that your files can have problems migrating. If this situation occurs, contact PureConnect Customer Care and open an incident. Have your logs and copies of your XML files available. You can continue after you open your ticket by using the phones in an unmanaged fashion until you are able to complete the migration successfully.

To create managed IP phones from existing SIP stations

1. Before you begin, ensure that you know the location of the Polycom phone/SIP station configuration files (`.cfg`), as the Managed IP Phone Assistant asks for this information. The `.cfg` file location is typically the root directory of your TFTP server, either a local directory path or a network share. Ensure that it is accessible from the computer where you plan to run the Managed IP Phone Assistant.

2. Click the **Managed IP Phones** container and then click **Managed IP Phone Assistant**.

   Note:
   Run the Managed IP Phone Assistant *after* business hours
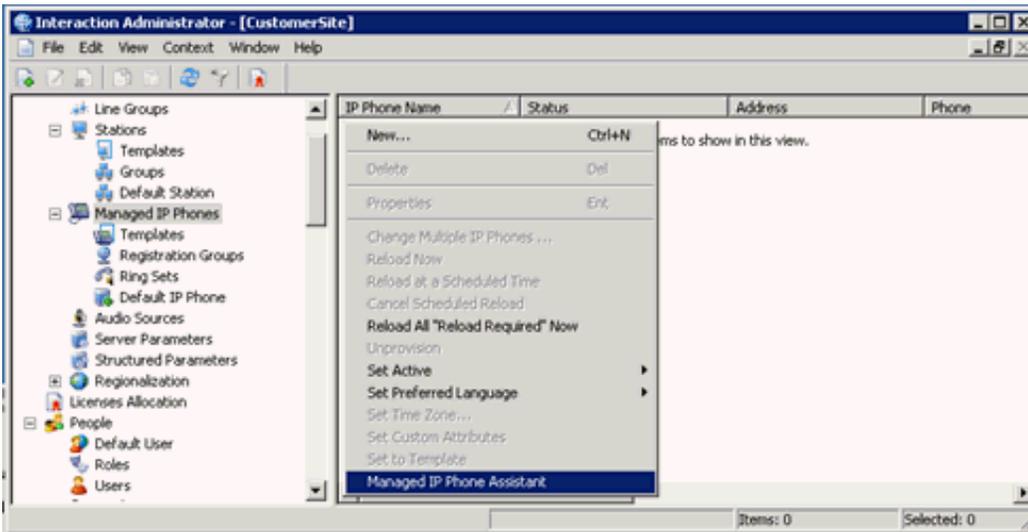
The Managed IP Phone Assistant **Welcome** page appears.
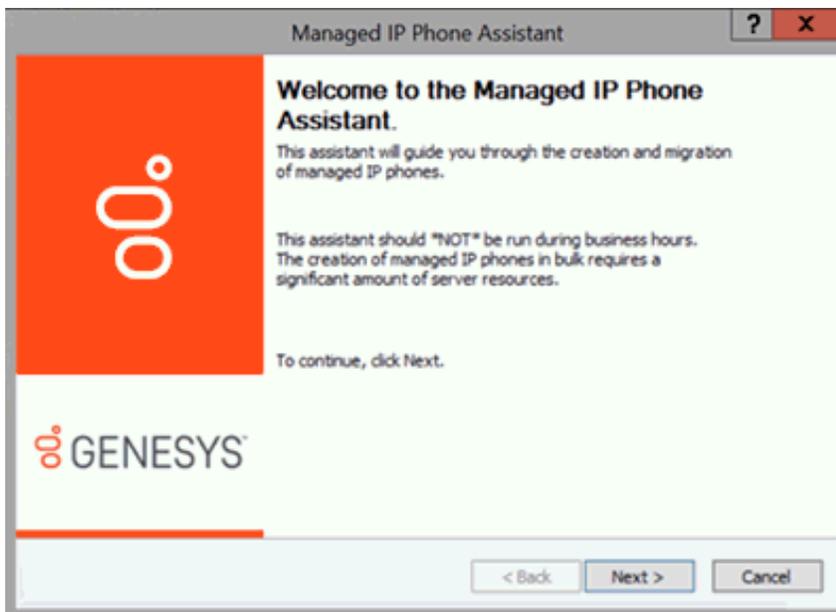
3. If you are running Managed IP Phone Assistant after business hours, click **Yes** to proceed past the warning message.

The **Add Managed IP Phones** page appears.
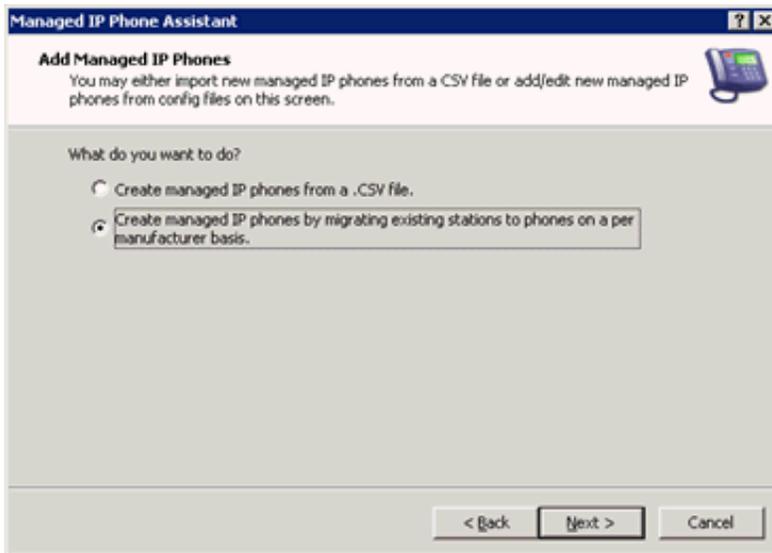
4. Click **Create managed IP phones by migrating existing stations to phones on a per manufacturer basis**. The **Select Manufacturer** page appears.
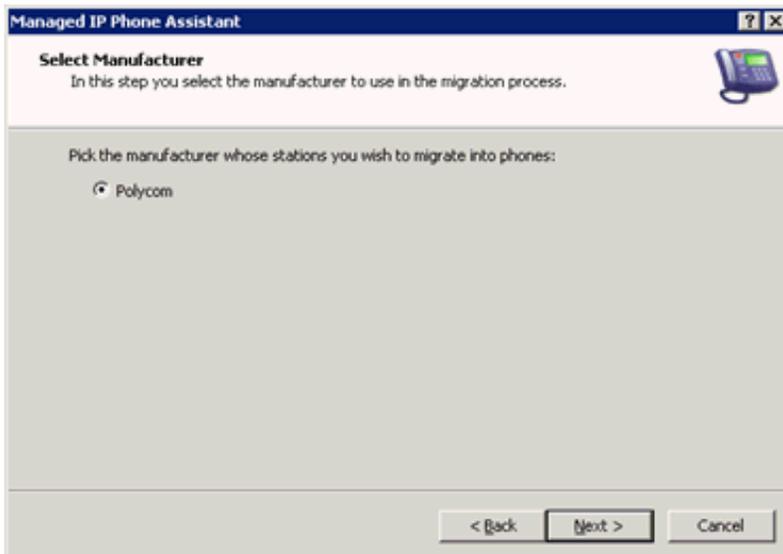
5. Choose the manufacturer of the stations to migrate. Currently **Polycom** is the only supported manufacturer. If you selected **Polycom** on the previous page, the **Select Default Model** page appears.

6. In the list box, click the default phone model to use when the system cannot derive the model from the stations on the phones that you are migrating.

   For example, SIP stations associated with a Polycom configuration file may not have a manufacturer or model set in Interaction Administrator. By selecting the default model, the Managed IP Phone Assistant knows what settings to use for the migration. In this example, the default model selected is an IP335.

   > **Note:**
   > If you set the SIP Phone Information Update server parameter as described in an earlier section, selecting the default model on this page is unimportant. The reason is that the server parameter ensures that each phone populates appropriately for manufacturer and model in Interaction Administrator.

   The **New Phone Naming** page appears.
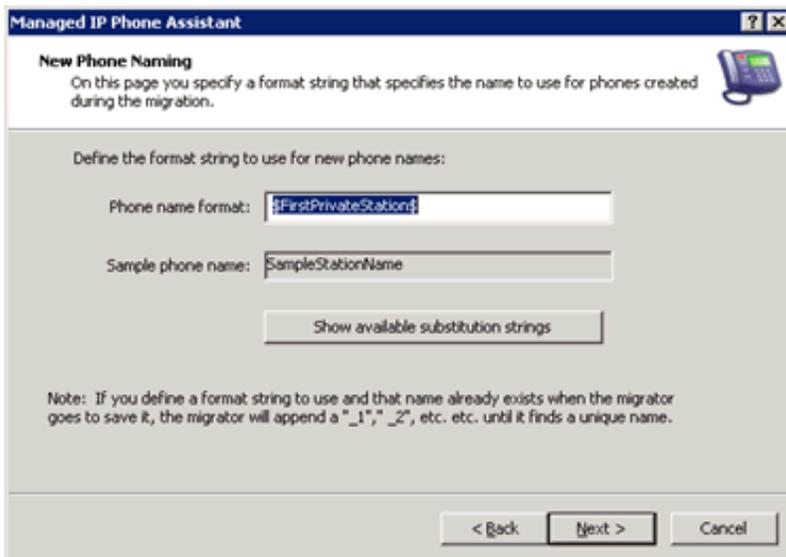
7. Specify a format string that indicates the name for the assistant to use for the migration item. The assistant creates an associated SIP station with the same name. There are two different substitution strings that you can use alone or with a format string to create phone names.

   Phone name format: Name of the first private station display name added to the migration item. Use a format string to define the phone's name when created. Use one or more substitution strings with a format string, so that every phone created has a meaningful and unique name.

   There are two different substitution strings:

   ○ **$FirstPrivateStation$**: Name of the first private station display name as defined in the `reg.x.displayName` attribute found in the SIP phone's `.cfg` file added to the migration item.

   ○ **MAC**: MAC address of the phone that you are migrating.

   By default, the assistant uses the $FirstPrivateStation$ substitution string. You can use a substitution string separately, in combination with the other substitution string, in combination with a format string, or use it alone. For example:

   ○ `$FirstPrivateStation$`

- MAC
- $FirstPrivateStation$-MAC-SecondFloor
- SecondFloorPhone **Using just text, the assistant appends the text with** _1, _2, **and so on after naming the first IP phone and associated SIP station.**

If a single format string of ManagedPhone is used without a substitution string, then all new phone names created would be named ManagedPhone_1, ManagedPhone_2, etc. If the substitution string MAC is used together with the format string of ManagedPhone, such as ManagedPhone – MAC, then the MAC address for each phone would be substituted in the new name, i.e., ManagedPhone – 0004f2008100.

> **Note:**
> The substitution strings are case-sensitive. For example, if Phone - mac is specified, mac will not resolve to the MAC address. The string must be entered as Phone – MAC for proper MAC address resolution.

Sample phone name: Indicates how the phone name format field resolves.

8. Click **Show Available Substitution Strings** to view the substitution strings. The **Phone Configuration File Directory** page appears.
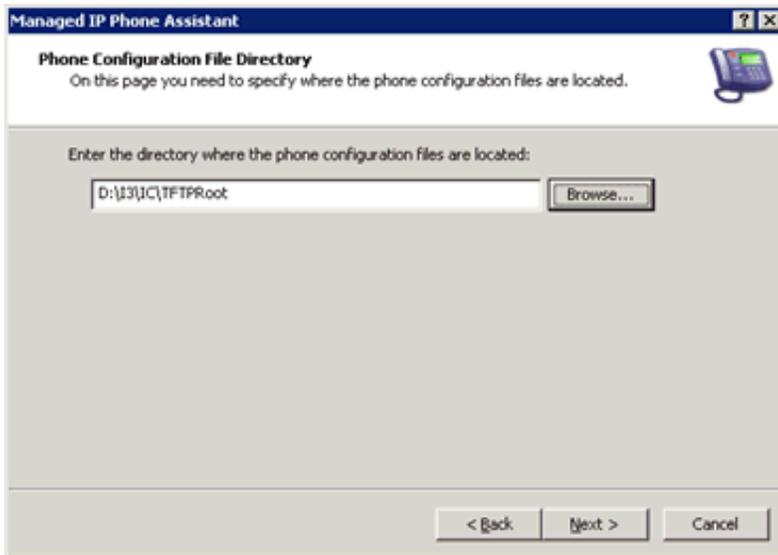
Type the directory or click **Browse** to select the directory where the existing IP phone configuration (.cfg) files are. The .cfg file is typically the root directory of your TFTP server, either a local directory path or a network share. Ensure that it is accessible from the computer where you plan to run the Managed IP Phone Assistant. In this example, the location is D:\I3\IC\TFTPRoot.

The Managed IP Phone Assistant searches this directory for .cfg files and displays a list of files found on the **Select the Items to Migrate** page. The assistant uses the settings in the selected .cfg files to create managed IP phone objects.

For the assistant to recognize a phone configuration file, the file must:

- Use the *XXXXXXXXXXXX*.cfg format, where *XXXXXXXXXXXX* is a 12-character alpha-numeric MAC address.
- Contain an APPLICATION XML element at the root that has a CONFIG_FILES XML attribute that specifies the other phone configuration files.

Each .cfg file that meets these criteria displays as a selectable item to migrate. The assistant needs read access to the phone configuration directory specified. For more information, click Help (**?**).

9. Click **Next**. The **Select the Items to Migrate** page appears.
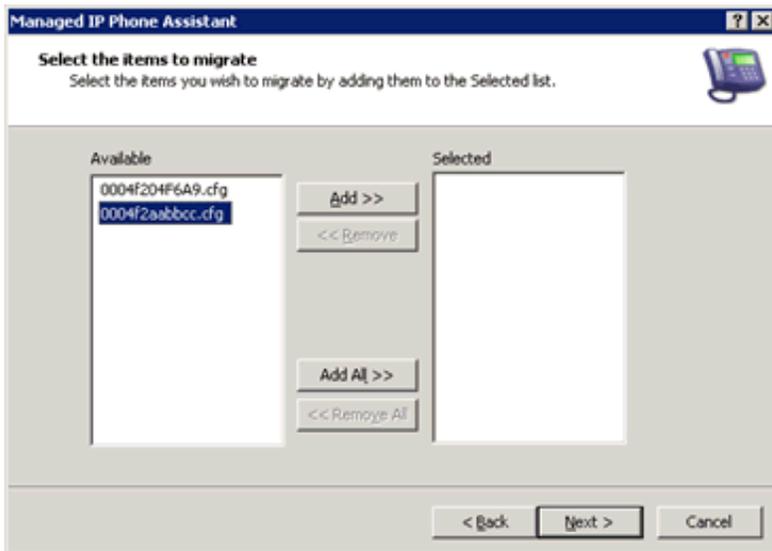
Select the items to migrate by clicking the items in the **Available** list and then clicking **Add** to move them to the **Selected** list. The items displayed in the **Available** list are the Polycom configuration files that the assistant found in the directory specified on the **Phone Configuration File Directory** page.

10. Select at least one item and then click **Next**. The **Build Migration Items** page appears.
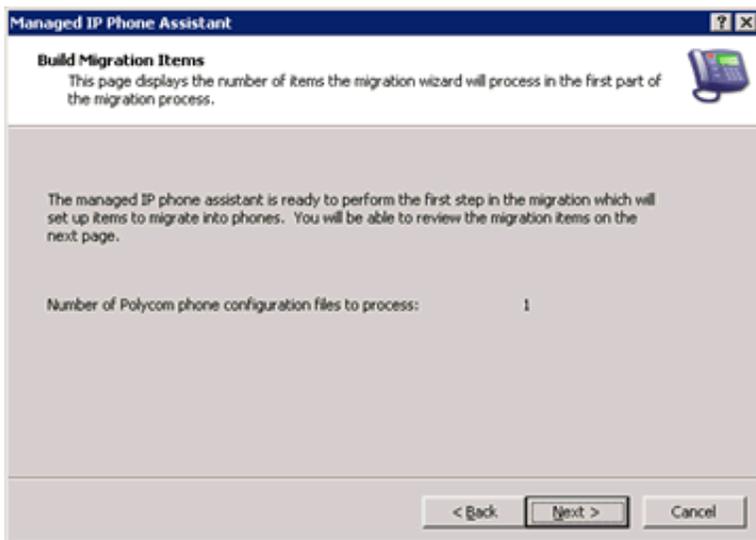
This page shows that the migration process is going to begin building migration items, and it lists the number of migration items included in the process. In this example, there is just one migration item.

11. Click **Next**. After the building the migration items, the **Current State of Migration Items** page appears.

The number of items with errors, warnings, or no errors or warnings appear.

Items with errors: The assistant *cannot* create a managed IP phone from these items. Click the **Show Item Details** button and review the **Build History** tab to aid in resolving the error.

Items with warnings: These items are candidates for migration at the current state, and the assistant *may* be able to create a managed IP phone from these items.

Items with no errors or warnings: These items are good candidates for migration at the current state, and the assistant will most likely be able to create a managed IP phone from these items.

> **Note:**
> Genesys strongly recommends that you click **Show Item Details** to review the migration item details. Item details include information about the new managed phone created for the selected migration item, and information about error and warnings listed for that migration item.

12. Click **Next**. The **Backup Directory Services** page appears.

Genesys strongly recommends that you complete a Directory Services backup. This step is the final step in the migration process before the Managed IP Phone Assistant starts creating the managed IP phones based on the migration items.

13. Click **Backup Directory Services** to do the backup.The Directory Services backup can take several minutes. There is a timeout set at 40 minutes. If the timeout is reached, the assistant displays **The CIC server was unable to perform a backup of Directory Services**. Either click **Try Again** to attempt another backup, or click **Continue** without making a backup.

When the backup is complete, the page displays the location of the backup file, for example, `D:\I3\IC\Backup\RegistryBackup_16-8-2011-458037`. Take note of this location.

This is the last page before the migration process starts. To learn more about what happens during the migration process, click Help (?).

A confirmation message appears.
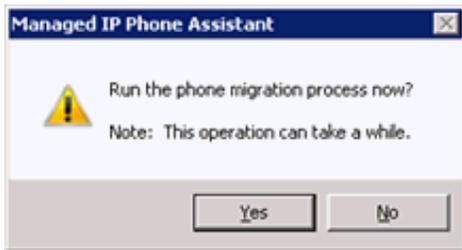
14. Click **Yes** to start the migration process. When the migration process is complete, the **Migration Results** page appears.
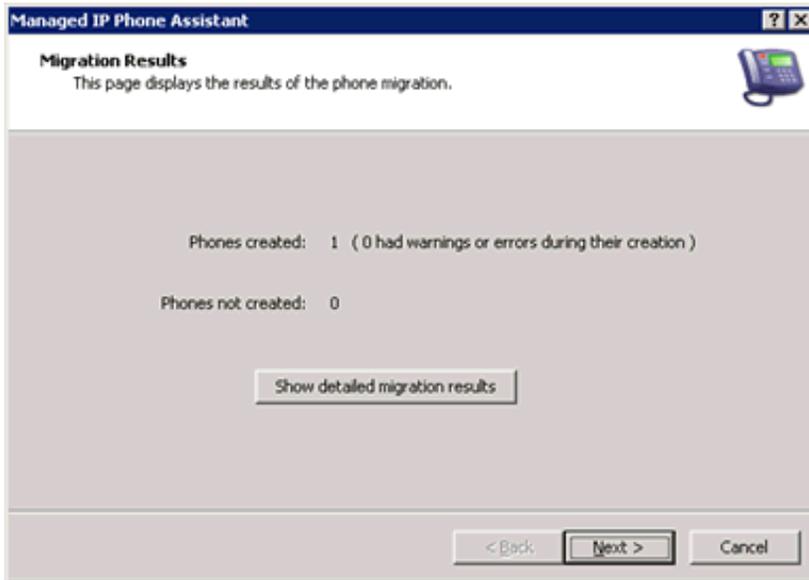
This page displays the number of managed IP phones created in the migration process and indicates whether warning or errors occurred.

15. Click **Show Detailed Migration Results** to view each step taken during the migration for each migration item.
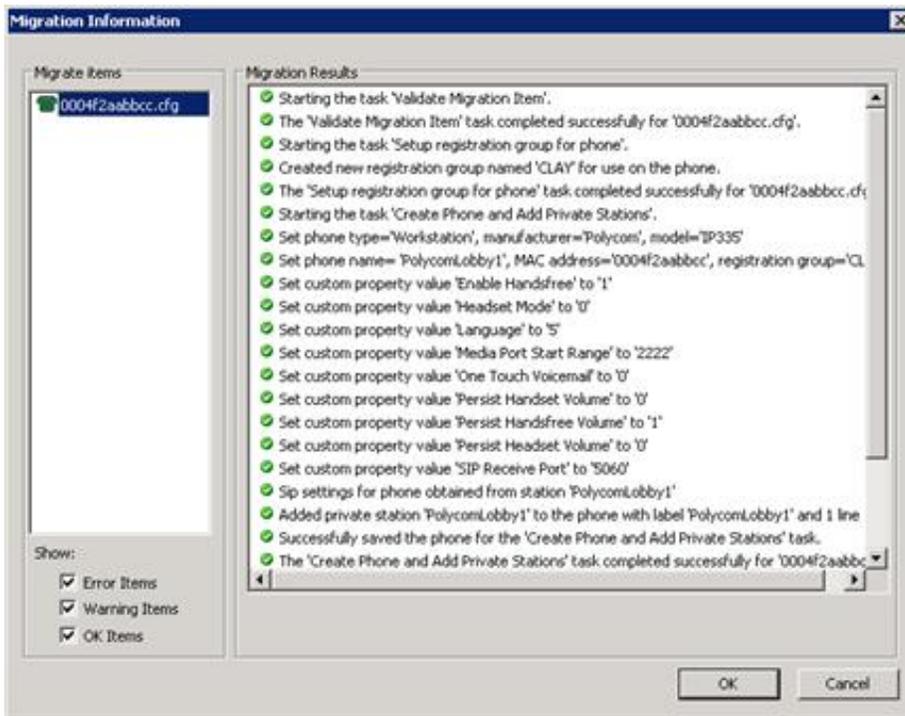
If errors and warnings occurred, you can see at what point they occurred and why. Depending on the migration results, the system might ask you to reload the phones; or you can make changes to individual IP phones in the Managed IP Phone container after the assistant completes and reload the phones then. For more information, click Help (?).

The Managed IP Phone Assistant process is now complete.

16. Click **Finish** to exit the assistant. The new managed IP phones appear in the **Managed IP Phones** container.
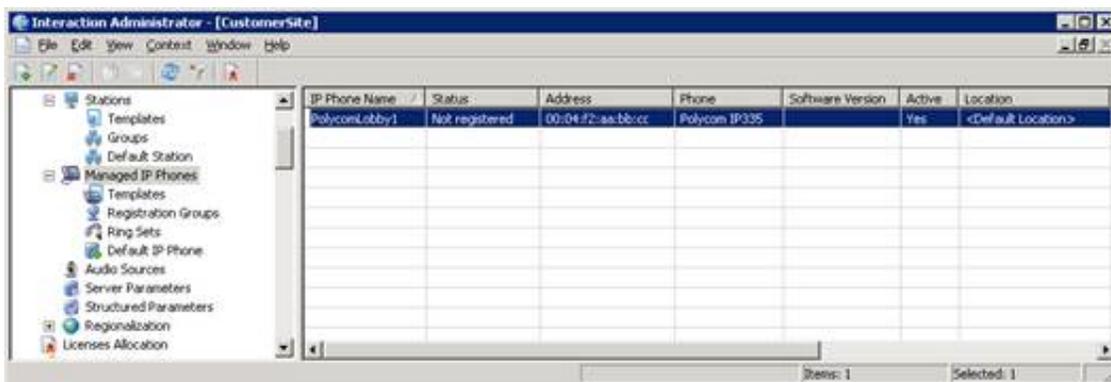
Notice that the new managed IP phone in this example has a status of **Not registered**. On the phone's next SIP registration, this status changes to **Up-to-date**.

As part of creating managed IP phones, the system creates associated SIP station appearances on those IP phones, of the type **Managed Workstation** or **Managed Stand-alone Phone**. You can view them in the **Stations** container.
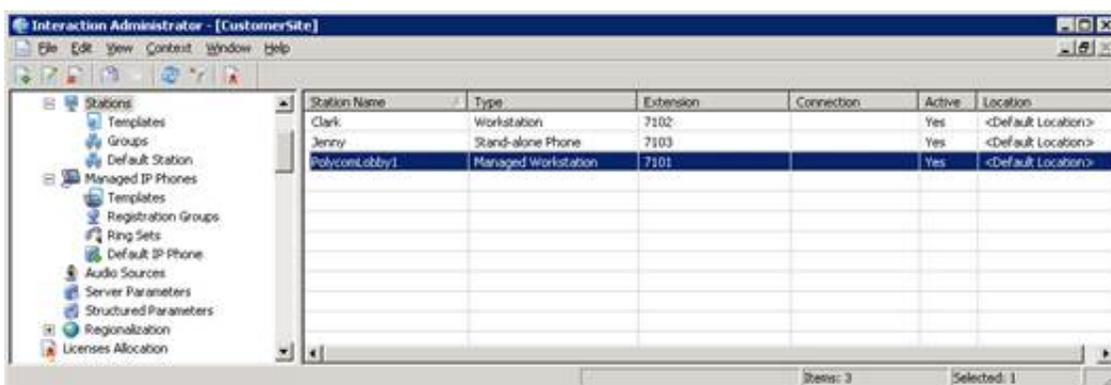
Complete the provisioning process

CIC provisioning of managed IP phones connects the managed IP phones and downloads their configurations from the CIC server. The goal for CIC managed IP phone provisioning is to *automate* it as much as possible so that it requires no user participation.

If you followed the instructions to configure the network and create managed IP phones from existing (Polycom) SIP stations using Managed IP Phone Assistant, **automated provisioning occurs once you set up and reboot the phones**.

When a managed IP phone contacts the provisioning subsystem, the provisioning subsystem matches the phone's computer name or MAC address with an existing managed IP phone configuration, and serves the configuration to the device without any extra steps.

Manual provisioning

Manual provisioning of some or all managed IP phones is necessary when:
- The CIC system has no DHCP server or it cannot access the DHCP server.
- Some Polycom phones are in remote locations.

Typically, an administrator with required permissions does the manual provisioning through the phone's provisioning IVR. In some cases (for example, remote locations), users do the manual provisioning. Do manual provisioning on each managed phone's configuration.

Polycom Firmware and Phones

Polycom Firmware for Supported and EOL Phones

Polycom Trio 8800 with Firmware 5.9.0 revision AD voicemail issue

Polycom Trio 8800 phones with Firmware 5.9.0 revision AD reboot or change to an idle state when attempting to access voicemail in one of the following ways:
- Using the one-touch feature
- Using the "Voicemail" option in the Feature menu

The workaround is to set the phone's "One Touch Voicemail" option to "No" or dial the voicemail extension from the phone.

Security notices

Remote Code execution vulnerability in UCS software

The Polycom Support Security Center posted a notification about a security vulnerability in UC software and possible remote code execution. This vulnerability affects VVX, Trio, SoundStructure, SoundPoint, and SoundStation phones when the web management interface is enabled. For more information, see https://support.polycom.com/content/dam/polycom-support/global/documentation/remote-code-execution-vulnerability-in-ucs-software-v1-2.pdf.

The following Firmware addresses this vulnerability (depending on your phone model):
- 5.9.3.2857 Rev G
- 5.9.0 Rev AD
- 4.0.14.1580

Voice endpoints - XSS and CSRF vunerabilities

The Polycom Support Security Center posted a notification about Cross-Site Scripting (XSS) and Cross-Site Request Forgery (CSRF) vulnerabilities on multiple Polycom voice endpoints. These vulnerabilities affect VVX, SoundPoint, and Trio phones. For more information, see https://support.polycom.com/content/dam/polycom-support/global/documentation/xss-and-csrf-on-ucs-1-0.pdf.

> **Important!**
> A firmware fix from Polycom is pending for Trio phones. Meanwhile, Genesys advises you to apply Polycom's mitigation recommendation. For more information, see https://support.polycom.com/content/dam/polycom-support/global/documentation/xss-and-csrf-on-ucs-1-0.pdf.

The following Firmware addresses these vulnerabilities (depending on your phone model):
- 5.9.6.2327
- 4.0.15.1009

Firmware for supported phones

The following table lists the Polycom firmware versions for supported phones.

Firmware Version	Initially Made Available in...	Phone Type	Phone Model
6.4.5.1251 (for New Hardware Revision)	2023r3 2023r2 Patch 5 2023r1 Patch 13 2022r3 Patch 17 2022r2 Patch 24 2022r1 Patch 32 2021r4 Patch 38	Desktop Phones	VVX250 VVX350 VVX450

6.4.5.1210 (for Legacy Hardware Revision)	2023r3 2023r2 Patch 5 2023r1 Patch 13 2022r3 Patch 17 2022r2 Patch 24 2022r1 Patch 32 2021r4 Patch 38	Desktop Phones	VVX250 VVX350 VVX450 VVX101 VVX150 VVX201, VVX301/311 VVX401/411 VVX501 VVX601
5.9.7.4477 Rev F	2023r3 2023r2 Patch 3 2023r1 Patch 11 2022r3 Patch 15 2022r2 Patch 22 2022r1 Patch 30 2021r4 Patch 37	Desktop Phones	D60 VVX101/201 VVX150 VVX250 VVX300/301 VVX310/311 VVX350 VVX400/401 VVX410/411 VVX450 VVX500/501 VVX600/601
5.9.6.3432	2023r3 2023r2 Patch 3 2023r1 Patch 11 2022r3 Patch 15 2022r2 Patch 22 2022r1 Patch 30 2021r4 Patch 37	Conference Phones	Trio 8500

5.9.6.2327	2020 R3	Desktop Phones	D60
	2020 R2 Patch 4		VVX101
	2020 R1 Patch 10		VVX 150
	2019 R4 Patch 16		VVX 201
	2019 R3 Patch 22		VVX 250
	2019 R2 Patch 29		VVX 300
	2019 R1 Patch 35		VVX 301
			VVX 310
			VVX 311
			VVX 350
			VVX 400
			VVX 401
			VVX 410
			VVX 411
			VVX 450
			VVX 500
			VVX 501
			VVX 600
			VVX 601
5.9.3.2857 Rev G	2019 R4	Desktop Phones	VVX101
	2019 R3 Patch 4		VVX 150
	2019 R2 Patch 11		VVX 201
	2019 R1 Patch 17		VVX 250
	2018 R5 Patch 23		VVX 300
	2018 R4 Patch 29		VVX 301
	2018 R3 Patch 35		VVX 310
			VVX 311
			VVX 350
			VVX 400
			VVX 401
			VVX 410
			VVX 411
			VVX 450
			VVX 500
			VVX 501
			VVX 600
			VVX 601

5.9.0 Rev AD	2019 R4 2019 R3 Patch 4 2019 R2 Patch 11 2019 R1 Patch 17 2018 R5 Patch 23 2018 R4 Patch 29 2018 R3 Patch 35	Conference Phones	Trio 8500 Trio 8800
5.8.1	2019 R2 2019 R1 Patch4 2018 R5 Patch10 2018 R4 Patch16 2018 R3 Patch23 2018 R2 Patch29 2018 R1 Patch35	Desktop Phones	VVX 150 VVX 250 VVX 350 VVX 450
5.8.0	2018 R5 2018 R4 Patch3 2018 R3 Patch10 2018 R2 Patch16 2018 R1 Patch23 2017 R4 Patch29 2017 R3 Patch36	Desktop Phones (with support of VVX D60 wireless handset pairing)	VVX 101 VVX 201 VVX 300/301 VVX 310/311 VVX 400/401 VVX 410/411 VVX 500/501 VVX 600/601 VVX D60
5.5.4	2018 R4 2018 R3 Patch5 2018 R2 Patch11 2018 R1 Patch18 2017 R4 Patch24 2017 R3 Patch31 2017 R2 Patch36	Conference Phones	Trio 8500 Trio 8800
5.4.5G	2017 R3 2017 R2 Patch6 2017 R1 Patch12 2016 R4 Patch17 2016 R4 Patch15 + ES 2016 R3 Patch23 2016 R2 Patch23 2016 R1 Patch27	Desktop Phones	VVX 101 VVX 201 VVX300/301 VVX310/311 VVX400/401 VVX410/411 VVX500/501 VVX600/601

4.0.15.1009	2020 R3 2020 R2 Patch 4 2020 R1 Patch 10 2019 R4 Patch 16 2019 R3 Patch 22 2019 R2 Patch 29 2019 R1 Patch 35	Desktop Phones Conference Phones	**SoundPoint** IP321 IP331 IP335 IP450 IP550 IP560 IP650 IP670 **SoundStation** IP6000 IP7000
4.0.14.1580	2019 R4 2019 R3 Patch 4 2019 R2 Patch 11 2019 R1 Patch 17 2018 R5 Patch 23 2018 R4 Patch 29 2018 R3 Patch 35	Desktop Phones Conference Phones	**SoundPoint** IP321 IP331 IP335 IP450 IP550 IP560 IP650 IP670 **SoundStation** IP6000 IP7000
4.0.8.2058.I (Required for new phones after June 2016 with MAC address range of 64:16:7f. See KB article.)	2017 R1 Patch1 2016 R4 Patch5 and Patch6 2016 R3 Patch12 2016 R2 Patch17 2016 R1 Patch22 2015 R4 Patch22 2015 R3 Patch25 2015 R3 Patch24 + ES	Desktop Phones Conference Phones	**SoundPoint** IP321 IP331 IP335 IP450 IP550 IP560 IP650 IP670 **SoundStation** IP6000 IP7000

5.2.2 (Default for VVX phones.)	2015 R2 Patch8 2015 R3 Patch2	Desktop phones	VVX300 VVX310 VVX400 VVX410 VVX500 VVX600
4.1.6h	SU6, SU5-ES, SU4-ES	Desktop phones	VVX300 VVX310 VVX400 VVX410 VVX500 VVX600
4.0.8c	2015 R2 Patch8 2015 R3 Patch2	Desktop phones Conference phones	IP321 IP331 IP335 IP450 IP550 IP560 IP650 IP670 IP5000 IP6000 IP7000
4.0.1b	SU3	Desktop phones Conference phones	IP321 IP331 IP335 IP450 IP550 IP560 IP650 IP670 IP5000 IP6000 IP7000

Firmware			Phone Type	Phone Model
3.3.2		2020 R2 2019 R3 Patch 14 2020 R1 Patch 2 2019R4 Patch 8 2019 R3 Patch 16	Desktop phones Conference phones	IP5000 IP335 IP321 IP331 IP450 IP550 IP560 IP650 IP670 IP6000 IP7000
3.2.7		SU4	Desktop phones Conference phones	IP331 IP321 IP335 IP450 IP550 IP560 IP650 IP670 IP5000 IP6000 IP7000

Firmware for End of Life (EOL) phones

The following table lists the Polycom firmware versions for EOL phones. Genesys no longer supports the firmware and phone provisioning; however the phones are still available from the manufacturer.

Firmware	Phone Type	Phone Model	Limitations
4.0.2b	Wi-Fi phones	SL8440 SL8450	
3.2.7	Desktop phones	IP320 IP330 IP430	
3.1.6	Desktop phones Conference phones	IP301 IP501 IP600 IP601 IP4000	Except for the IP4000, does not support SRTP.
2.1.4	Desktop phones	IP300 IP500	Does not support SRTP, DND syncing, idle screen configuration, or voice-quality monitoring.

Polycom firmware history

The following table shows Polycom firmware released (CIC 2015 R2 or later) and supported previously and their initial availability.

Note: Starting with CIC 2016 R4 Patch 2, CIC 2016 R3 Patch 8, CIC 2016 R2 Patch 14, CIC 2016 R1 Patch 20, CIC 2015 R4 Patch 20, CIC 2015 R3 Patch 24, and CIC 2015 R2 Patch 24+, Genesys provides support for Polycom phone models VVX 301, 311, 401, 411, 501, and 601. *

Firmware Version	Initially Made Available in...	Phone Type	Phone Model
5.4.3.2036 (Required for new phones after June 2016 with MAC address range of 64:16:7f.. See KB article.)	2017 R1 Patch 1 and later 2016 R4 Patch 6 and Patch 7 2016 R3 Patch 13 2016 R2 Patch 18 2016 R1 Patch 22 2015 R4 Patch 22 2015 R3 Patch 25 2015 R3 Patch 24 + ES	Desktop Phones	VVX 101 VVX 201 VVX300/301 VVX310/311 VVX400/401 VVX410/411 VVX500/501 VVX600/601 *
5.4.0a Use 5.4.3.2036 instead.	2016 R4	Desktop Phones	VVX 101 VVX 201
5.4.3.1014 (Required for new phones after June 2016 with MAC address range of 64:16:7f.. See KB article.) Use 5.4.3.2036 instead.	2016 R3 Patch 6 2016 R2 Patch 11 2016 R1 Patch 17 2015 R4 Patch 19 2015 R3 Patch 23 2015 R2 Patch 24 + IC-138829	Desktop Phones	VVX300/301 VVX310/311 VVX400/401 VVX410/411 VVX500/501 VVX600/601
4.0.8.1972.H (Required for new phones after June 2016 with MAC address range of 64:16:7f.. See KB article.) Use 4.0.8.2058.I instead.	2016 R3 Patch 6 2016 R2 Patch 11 2016 R1 Patch 17 2015 R4 Patch 19 2015 R3 Patch 23 2015 R2 Patch 24 + IC-138829	Desktop Phones Conference Phones	**SoundPoint** IP321 IP331 IP335 IP450 IP550 IP560 IP650 IP670 **SoundStation** IP6000 IP7000
5.2.2 (Default for VVX phones.)	2015 R2 Patch 8 2015 R3 Patch 2	Desktop Phones	VVX300 VVX310 VVX400 VVX410 VVX500 VVX600

Polycom Firmware Update Required

Starting in June 2016, Polycom introduced a new MAC address range for VVX; SoundPoint IP and SoundStation phones; and SoundStructure Installed Audio products. The new address range replaces the `00:04:xx:xx:xx:xx` MAC address range. Phones with a MAC address in the range `64:16:7f:xx:xx:xx` require a firmware update, included in supported CIC patch releases starting with CIC 2015 R2. For more information, see the [PureConnect KB article](#).

Selectable Polycom Firmware

Starting with CIC 4.0 SU 3, administrators can select from a list of supported Polycom firmware versions for a specified model to apply to a managed IP phone or group of managed IP phones in Interaction Administrator.

The selectable firmware feature allows you to:

- Control CIC release deployment to managed IP (Polycom) phones. For example, you can leave the phones on an older firmware version when deploying a release. Then, set a few phones to the new firmware to test it. Then, push out all the phones when ready.
- Test for a regression by pushing a test phone back to an older firmware version for verification purposes.

> **Note:**
> If the selected Polycom IP Phone model(s) do not support the selectable firmware feature option, the option doesn't appear.

To select a firmware version for a single managed IP phone

1. Double-click a managed IP phone in the **Managed IP Phones** container. The **Managed IP Phone Configuration** dialog box appears.
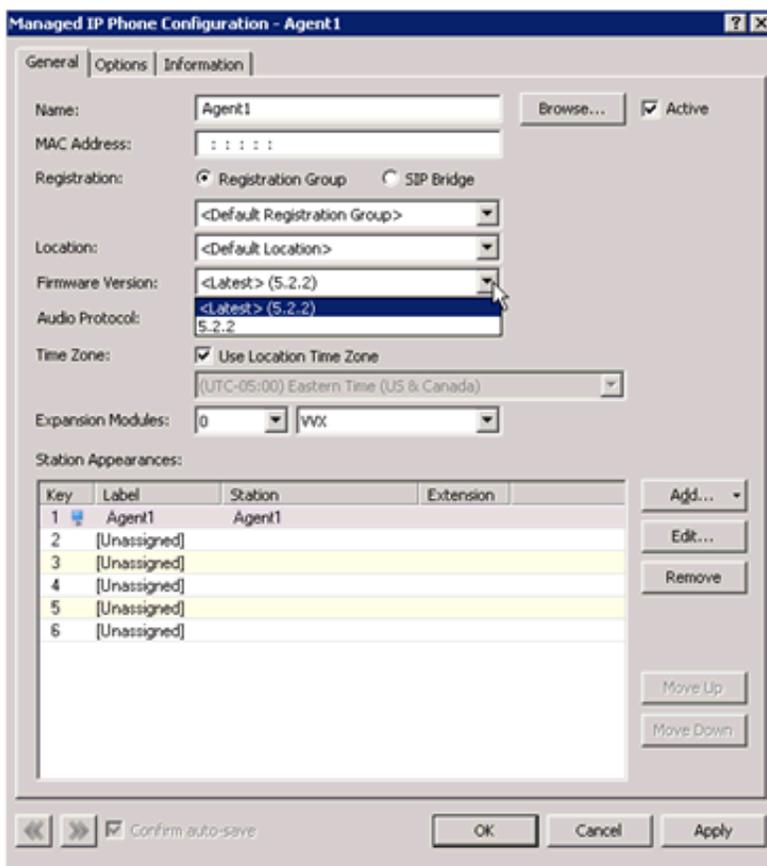
2. In the **Firmware Version** list box, click the firmware version. By default, the recommended option of <Latest> is selected.

   Older firmware versions installed on this system previously can also appear in the list box. In certain scenarios, you can select older approved firmware for this Polycom IP phone model. For example, to control the rollout of new firmware to a managed IP phone or group of managed IP phones during a release update. If the selected Polycom IP phone model does not support this feature, this option doesn't appear.

To select a firmware version for multiple IP phones

1. Click two or more managed IP phones in the **Managed IP phones** container, right-click, and then click **Change Multiple IP Phones**.
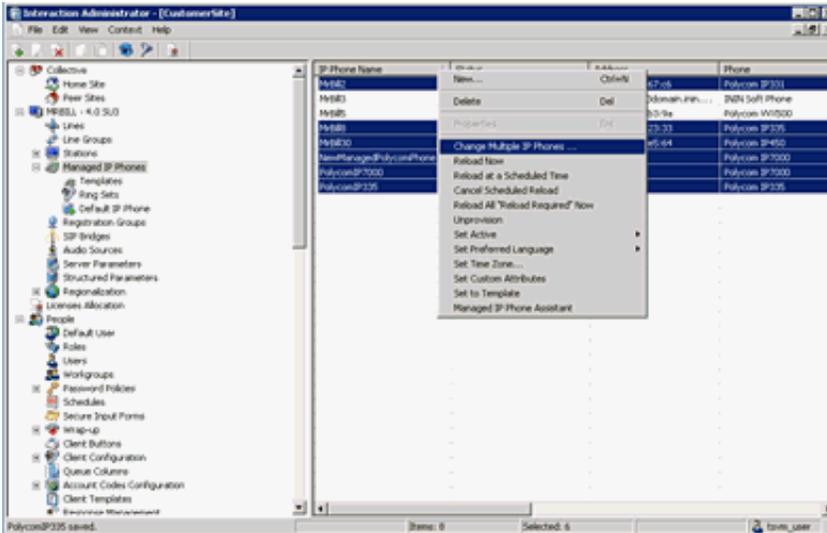
2. From the **Firmware Version** menu, click the firmware version. By default, the recommended option of <Latest> is selected.
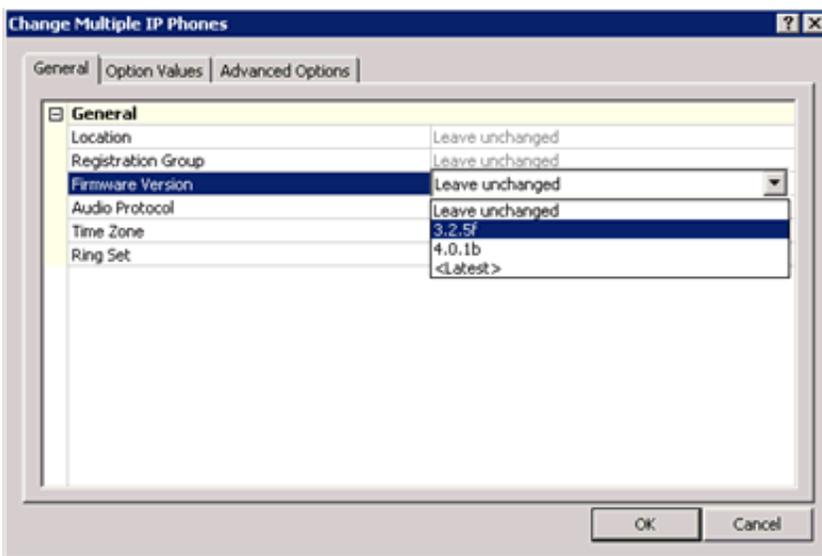
SpectraLink Wi-Fi Phone Considerations

The SpectraLink SL8440 and SL8450 Wi-Fi models differ significantly from the other Polycom phone models. The most basic difference is the wireless connectivity capability of the SpectraLink phones through Wi-Fi. Along with supporting 802.11b/g/n, they also support WEP, WPA, WPA2-PSK, and WPA2-Enterprise for wireless security. They also require business class wireless access points that support WMM (Wi-Fi Multimedia) protocols, requiring extra steps of Wi-Fi connectivity configuration before DHCP/DNS setup.

It is also important to note that the firmware and provisioning of these devices are "End of Life" (EOL), so newer supported firmware versions are not available. For a complete list of supported standards and recommended practices, see the Polycom product documentation.

Support for End of Life Devices

SIP Handset manufacturers, such as Polycom, move their products to an "End of Life" (EOL) status sometime after they stop production of that particular product model. Once this happens, they also stop maintaining the firmware used with those models, typically. Genesys has opted to attempt to support our customers who still operate on these EOL handsets by continuing to make them available as Managed IP Phones. This support, however, is limited.

To date, all EOL phones are still available when selecting the model of your Managed IP Phone, their configuration files still generate, and the CIC server still includes the last supported firmware files. While Genesys opted to support the administration, configuration, and interoperability of these devices with CIC, Genesys is unable to address any issues determined to exist at the device level. Genesys doesn't have the ability to create firmware, nor obtain newer firmware from the manufacturers for products listed as EOL. Also, Genesys can discontinue the support of any EOL device for any reason in the future.

To check the status of a particular phone model, consult the manufacturer. For the latest phones that Genesys tested, see the Testlab site.

Polycom Provisioning FTP Adapter

Polycom phones running 2.1.4 firmware (the IP300 and IP500) use the **Provisioning FTP Adapter** for their initial firmware requests. This adapter by default listens on Port 21, but you cannot disable it by creating the server parameter `Provision FTP Enabled` and setting it to `No`.

Managed IP Phone (Polycom) Configuration Options

Availability of configuration options for Polycom managed IP phones depends on the model of the phone. Newer models tend to have more options.

Options tab

The managed IP phone configuration options are available on the **Options** tab when editing one or more managed Polycom phone(s) from the **Managed IP Phones** container. The **Options** tab contains configuration options such as port settings, localization, volume persistence, and emergency information. For more information about the configuration options and their meaning, see the *Interaction Administrator Help*.
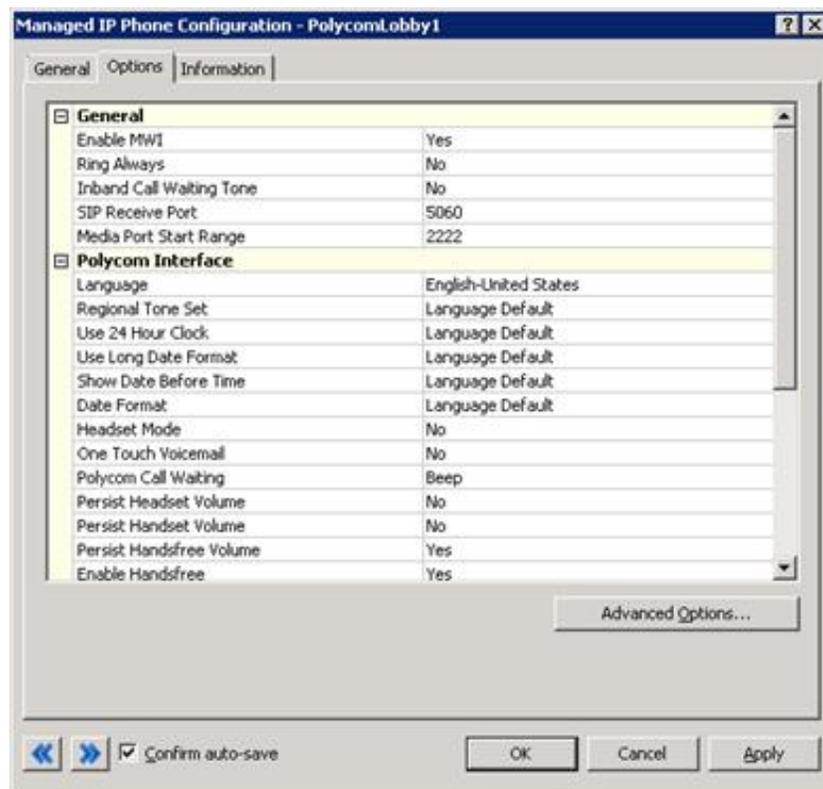
Advanced Options

The managed IP phone configuration advanced options are available on the **Advanced Options** tab when editing one or more managed Polycom phone(s) from the **Managed IP Phones** container. The **Advanced Options** tab contains configuration options for timeout, Polycom features, echo/noise suppression, gain settings, auto-dial, local Polycom dial plan, Network Address Translation (NAT), flash parameters, syslog, voice quality monitoring, and phone-specific SIP security. For more information about the advanced configuration options and their meaning, see the *Interaction Administrator Help*.
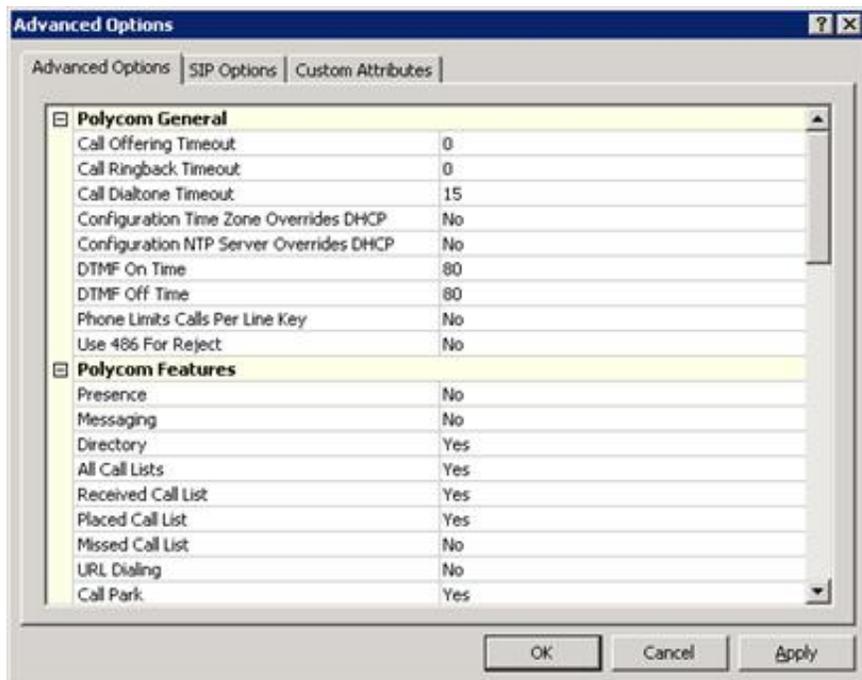
Other Managed IP Phone (Polycom) Features

Following are some of the unique features that Polycom phones offer over other managed IP phones types.

Call parking / Zone paging

Call parking places a parked call (held call) into a specific call queue, or orbit. Any other Polycom phone can pick up the parked call by specifying which orbit to pick up from.

Zone paging allows one phone to page an entire dial group (through an extension). It passes one-way audio to all phones reachable at the specified extension.

Dial string	Function
*901 <extension>	Zone page
*902 <orbit#>	Park call in orbit <orbit#>
*903 <orbit#>	Pick up parked call from orbit <orbit#>
*904 (<orbit#>)	List parked calls (in orbit <orbit#>)

Dial options for Call parking and Zone paging

Shared line appearances

Polycom phones support shared line appearances, allowing mirroring of one line over multiple stations. For example, an assistant has an appearance of the manager's phone line on their phone so that the assistant can see whether the manager is on the phone, and answer the manager's calls.

The first step for setting up shared lines is to set the desired stations as sharable. Only stations on Polycom phones support this feature. Use Interaction Administrator to edit the station appearance on a managed Polycom IP phone.

After you set the station to *Sharable*, you can add it to the Polycom managed IP phone as a shared station appearance. Select the station appearance and a dialog box appears with the configuration of the shared appearance. Afterward, reboot the phone with the shared appearance.
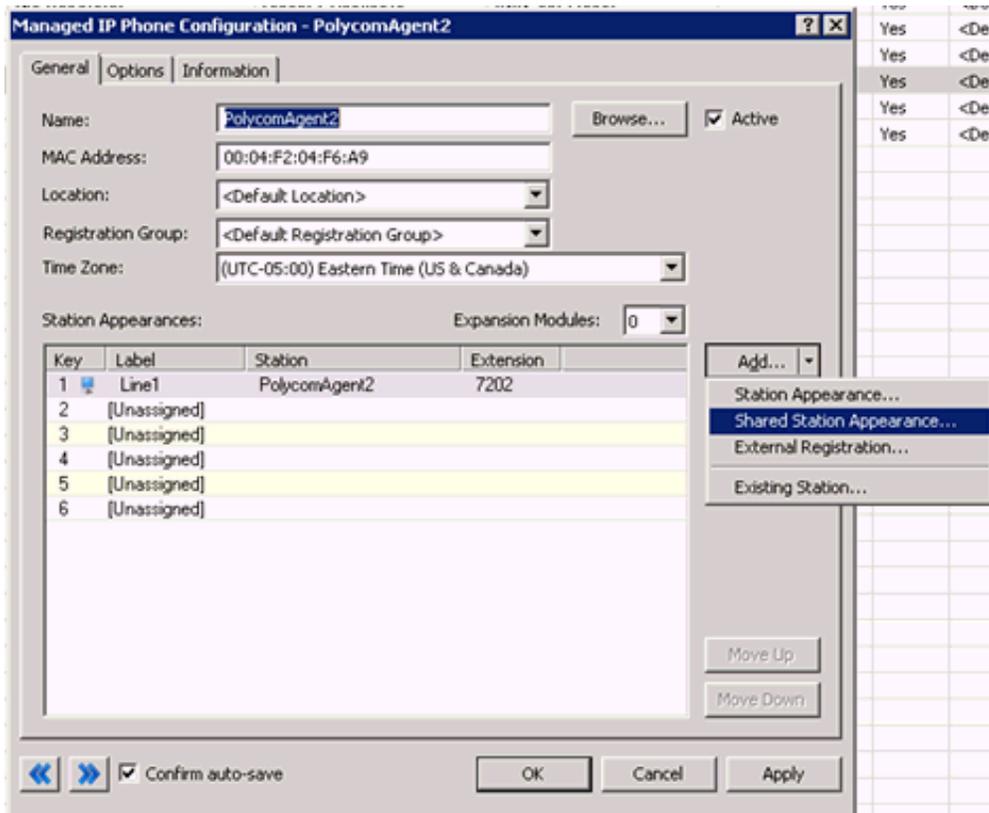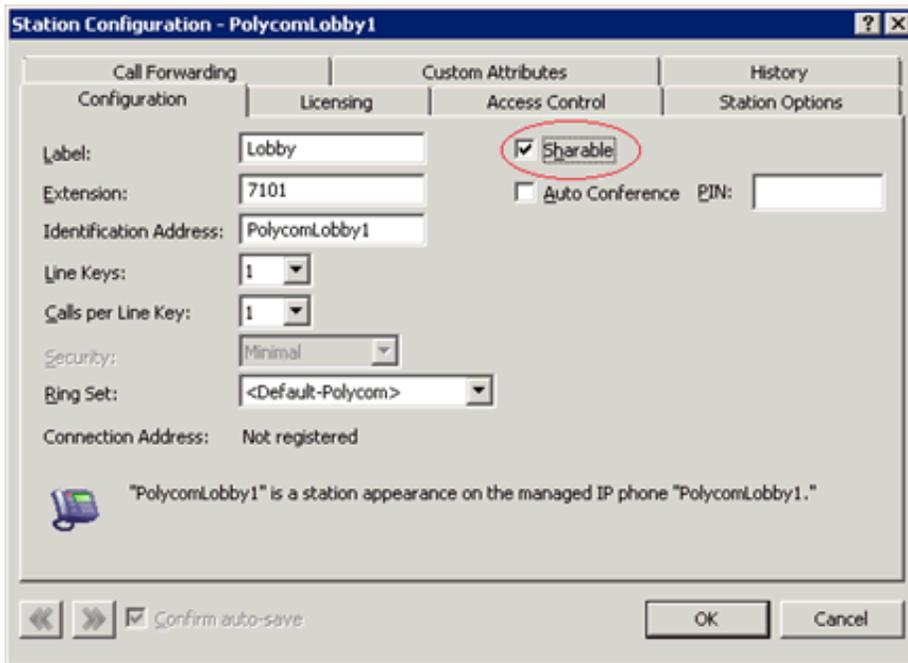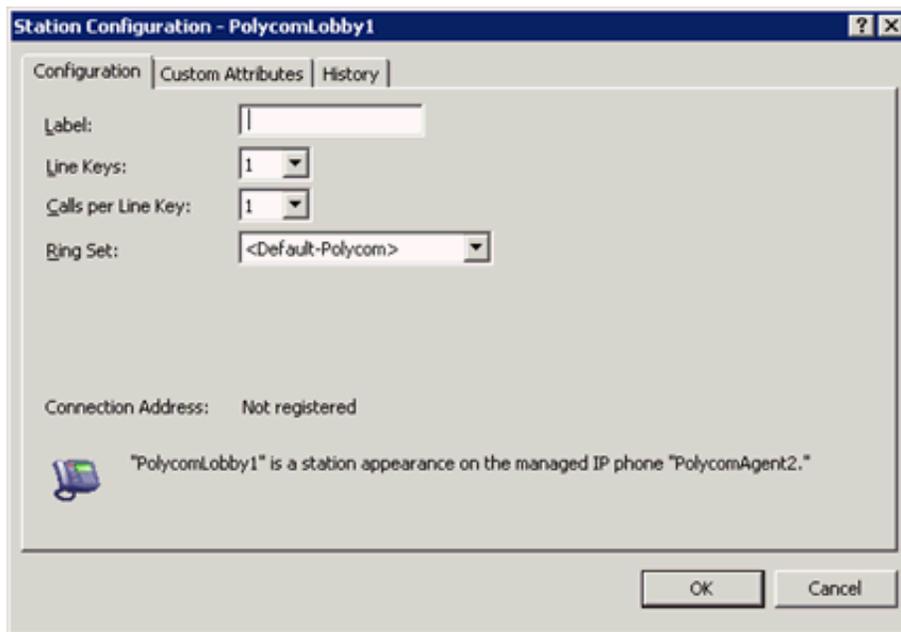
Custom configuration files (Polycom)

The provisioning server supports the ability to add a custom attribute to Polycom phones named config_files (case-insensitive), which must contain a comma-separated list of files. If the file exists in the `\i3\ic\provision\polycom` directory on the CIC server, provisioning includes the files in the config_files list for Polycom phones. The system requests these files after provisioning generates the config files and allows administrators to set custom config file attributes that provisioning doesn't handle (for example, microbrowser settings). For more information about the file inheritance model that Polycom phones use in the provisioning sequence, see Boot and Provision Sequences (Polycom).

The `config_files` custom attribute supports [MACADDRESS] and [MODEL] placeholders, with the value being substituted automatically (for example, *[MACADDRESS]*`-config.cfg` changes to `0004f2000000-config.cfg` and *[MODEL]*`-config.cfg` changes to `IP330-config.cfg`). With this mechanism, you can set custom configuration on a per-phone and a per-model basis.
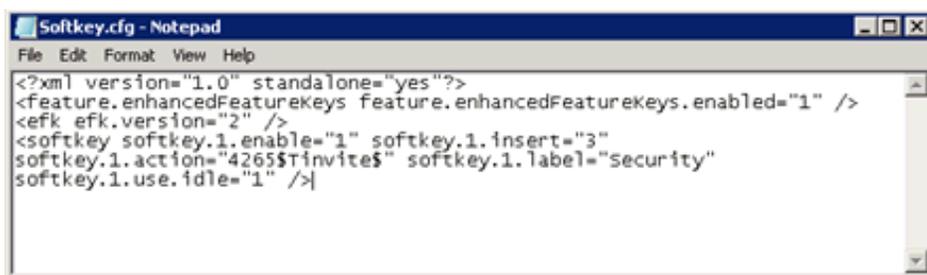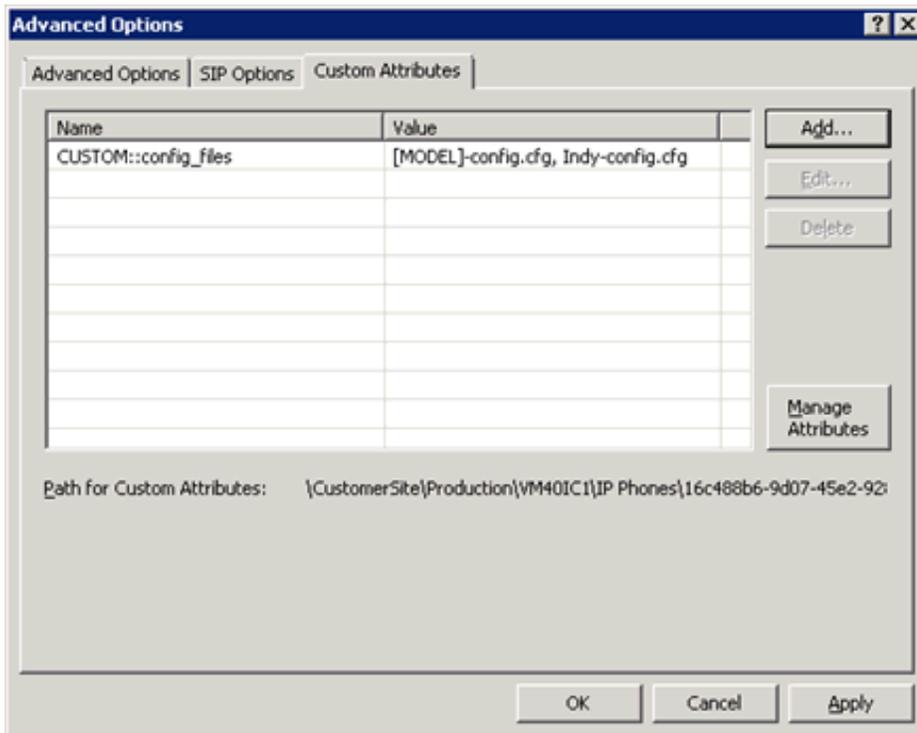
Supported languages for Polycom phones

Language support for Polycom phone models depends on the version of your Polycom firmware. For information about your firmware version, see Polycom's VoIP SIP Software Release Matrix.

Following is a list of all known languages supported as of Polycom 3.2.5c:

Chinese, China (for IP 450, 550, 560, 650, 670 and IP 6000, 7000 only)	French, France	Polish, Poland (all phones except IP 301)
Danish, Denmark	German, Germany	Portuguese, Portugal
Dutch, Netherlands	Italian, Italy	Russian, Russia
English, Canada	Japanese, Japan (for IP 450, 550, 560, 650, 670 and IP 6000, 7000 only)	Slovenian, Slovenia (all phones except IP 301 and IP 4000)
English, United Kingdom	Korean, Korea (for IP 450, 550, 560, 650, 670 and IP 6000, 7000 only)	Spanish, Spain
English, United States	Norwegian, Norway	Swedish, Sweden

> **Note:**
> The IP301 model does not support *any* languages other than its internal default (English, United States).

You configure language support for managed Polycom phones in Interaction Administrator in **Managed IP Phones Configuration**...**Options** under **Polycom Interface**.
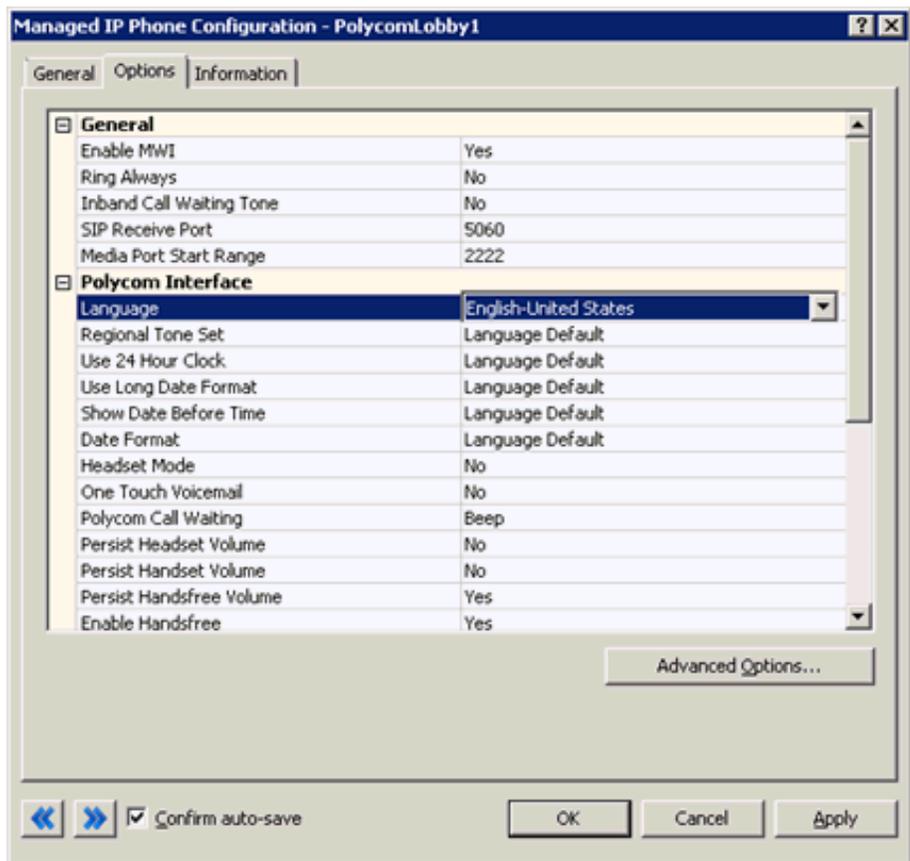
External registrations (Polycom)

Polycom phones support the use of the external registrations feature, allowing a specific line to register differently than all other lines on that phone. You add an external registration to a phone using the **Managed IP Phone Configuration** dialog box in Interaction Administrator. For more information, see Appendix B: How Registrations Work (Proxy Settings).

Phone Simulator

Phone Simulator - Simple but powerful tool that allows you to point to an IC Server and emulate a Polycom, Interaction SIP Station, or AudioCodes 420HD phone's requests for configuration files. Also, the simulator performs all inheritance calculations and displays the result for all attributes in a searchable/filterable pane.

Troubleshooting (Polycom)

Following are a few of the more common problem scenarios encountered with Polycom phones. In all these scenarios, it is useful to obtain a packet capture to see exactly what traffic passes to and from the phone, including DNS, HTTP, and SIP traffic.

Polycom Phone is set to use TLS but can't receive calls

If a Polycom phone cannot connect to a network time server (through SNTP) to determine the time, it cannot register or validate a certificate sent from the CIC Server to validate the call connection. All calls fail silently. Also, if the Polycom phone cannot connect to a network time server when booting, the phone fails.

To resolve this issue, ensure that each Polycom phone using TLS can connect to an NTP server. Do the following:
- Set the NTP server on the DHCP server through either DHCP Option 004 or 042.
- If no domain controller is available, set the CIC server as the SNTP server.

For more information, see Configure the Time Server.

Polycom phone cannot locate boot server

By default, the phone's Boot Server setting is **Custom + Option 66**, with Custom's default value set to **160**. You can set both these values on the phone. When the phone looks at options that the DHCP server supplies, it looks for Option 160 first. If that option does not define a boot server, the phone looks at Option 66. As described in Polycom DHCP Record Notes, you normally do not need to set Option 66 on a Polycom phone for the phone to locate the boot server.

> **Notes:**
> - Steps often differ depending on the phone model used and whether the phone is starting or already running. If these steps don't match your phone, see the Polycom documentation for configuration instructions.
> - The SpectraLink phones do not have a menu key; rather, the main screen has an icon called **Settings** that gives access to the configuration menus. Otherwise, the menus are similar to the configuration menus on other models. The **Network Settings** differ, though the difference is because of the SpectraLink phones running Polycom's 4.x firmware rather than to the model itself.

To verify that the phone's Boot Server settings are correct
1. On the phone's **Network Configuration** screen, do the following:
   a. Press `Menu`, press `3` for Settings, and then press `2` for Advanced.
   b. Type the password (the default is `456`) and then press `Enter`.
   c. Press `1` for **Admin Settings**.
   d. Press `1` for **Network Configuration**.
2. On the DHCP menu screen, do the following:
   a. Choose **DHCP Menu** and then choose **Select**.
   b. Choose **Boot Server** and then press `Edit`.
   c. Use the arrow keys to select **Custom + Opt 66** and then press `OK`.
   d. Choose **Boot Srv Opt** and then press `Edit`.
   e. Using the keypad, type `160`.
   f. Press `OK` and then press **Exit** twice.
3. Select **Save Config**. The phone saves your configuration changes and reboots. If it still cannot locate the boot server, verify through DNS that the hostname configured in DHCP resolves to the IP address of the desired CIC server.

Polycom phone contacts boot server but cannot register

If the non-provisioned phone can contact the boot server, it displays a line labeled Setup. If the phone is provisioned, it displays the line label set in Interaction Administrator.

If the phone is not registered:
- When you go off-hook, the phone displays the message *URL call is disabled*.
- The phone icon next to the line label appears hollow.

If you set up a Polycom phone using the Configure a Polycom phone's boot server manually procedure (in Appendix A) but the phone fails to register
1. Repeat Steps 1 and 2 of the procedure for configuring the phone manually. The screen shows the boot server's URL or IP address followed by the port number.
2. Verify that the boot server and port information is correct.

To determine whether registration failure is because the phone cannot resolve the short name of its server through DNS
1. Open Windows Explorer on your CIC server.
2. Locate the Polycom log under `\I3\IC\Logs\`*`<yyyy-mm-dd>`*`\phones`. The filename is the MAC address of the phone-`boot_`*`<log sequence number if any>`*.
3. Open the log in Windows Notepad or a text editor.
4. In the log, go to the section beginning with **DHCP returned result**. The log should look like the following:

```
 0101000017|app1 |3|00|DHCP returned result 0x38F from server 10.250.0.2.
0101000017|app1 |3|00|Phone IP address is 10.250.0.92.
0101000017|app1 |3|00|Subnet mask is 255.255.255.0.
0101000017|app1 |3|00|Gateway address is 10.250.0.1.
0101000017|app1 |3|00|Boot server address is http://lab1ic.sbsdomain.local:8088.
0101000017|app1 |3|00|DNS server is 10.250.0.2.
```

```
0101000017|app1 |3|00|DNS alternate server is 10.250.0.1.
0101000017|app1 |3|00|DNS domain is sbsdomain.local.
```

5. In the DHCP returned result section:

   In the line that begins `DNS server is`, note the URL of the DNS server. If this value is not present, then DHCP Option 6 is not defined.

   In the line that begins `DNS domain is`, note the name of the DNS domain. If this value is not present, then DHCP Option 15 is not defined.

6. Do the following to determine whether the DNS server can resolve the DNS domain:

   a. Open a command prompt, type `nslookup`, and then press **Enter**. The command window appears as follows:

   ```
   C:\>nslookup
   Default Server:nighthawk.example.com
   Address:172.16.1.2
   ```

   b. If the default server address differs from the one in the Polycom phone's log, type `server`, the address from the Polycom phone's boot log, and then press `Enter`. Windows changes the server address to the address that you provided. The command window appears as follows (except that your server address is different):

   ```
   > server 10.250.0.2
   Default Server:[10.250.0.2]
   Address:10.250.0.2
   ```

   c. Type the short name of your CIC server (such as `lab1ic`) and the domain name suffix in the Polycom phone's boot log above (such as `sbsdomain.local`), and then press `Enter`. If the DNS server can resolve the CIC server name, the command window appears as follows:

   ```
   > lab1ic.sbsdomain.local
   Server:[10.250.0.2]
   Address:10.250.0.2
   Name:lab1ic.sbsdomain.local
   Address:10.250.0.51
   ```

Boot and Provision Sequences (Polycom)

The behavior of a Polycom phone when powering on is helpful while troubleshooting a phone to understand what it is doing during startup. When the phone powers up, it runs through the bootloader (boot sequence), checking its current firmware. It then starts the SIP application and gets its configuration (provision sequence).

Polycom boot sequence

> **Note:**
>
> Starting with CIC 2017 R3, Interaction Administrator includes the following advanced options for Polycom phones capable of 4.0 or newer firmware:
> - Boot Server Type
> - Boot Server Option
> - Boot Server Option Type
> - Provisioning URL
>
> If the **Boot Server Type** option is set to **Static**, the phone uses the value of the **Provisioning URL** option instead of the DHCP option during the Polycom boot sequence. For more information, see the *Interaction Administrator Help*.

Following is the Polycom boot sequence:
- Phone powers up and does CDP/LLDP VLAN discovery
  - Troubleshoot using packet capture
  - If the switch returns a response, it uses that VLAN
- Runs through the DHCP discovery
  - Troubleshoot using packet capture, syslog, or bootlog

- DHCP *must* return
  - IP Address
  - Option 1:Subnet Mask
  - Option 6: DNS server(s) - These servers must be able to resolve the name of the CIC server (short name + option 15). If using switchover, they must also be able to resolve SRV entries for the phone's domain.
- DHCP *should* return
  - Option 3: Router - used when the phone needs to contact any resources not within its subnet.
  - Option 15: Domain Name - used for DNS A Record lookup.
  - Option 160: Custom Provisioning Server - The phone defaults to `Custom + Opt 66` with `Custom` defined as `160`. It checks for a provisioning server in Option 160 of the DHCP response first, then falls back to Option 66.
    - If 160 or 66 isn't set:
      - If a boot server value resides in flash memory and the value is not `0.0.0.0`, it uses the value in flash memory.
      - Otherwise, the phone sends out a DHCP INFORM query.
      - If it fails, it reports on the screen that it `Failed to contact boot server.`
- DHCP *may* return
  - Option 4/42: Time Server(s) - defined so that the phone can do SNTP queries to determine time. If you defined this option in the Configuration files generated from IA, these DHCP Options are optional.
  - Option 2: Time Server Offset - If you have your time server derived from the Option 4 or 42 values, you must also define the offset.
- Phone asks for model-specific `bootrom.ld` (example `/2345-12500-001.bootrom.ld`)
  If it has a different version, it downloads and saves the new version; and then reboots
- Phone asks for *<MAC>*`.cfg` (example: `/0004f218dece.cfg`)
  - This file contains a list of files that the phone should request to obtain configuration
  - If there is no managed phone with this MAC address
    - Provision Server creates a provisional station
    - *<MAC>*`.cfg` populates with provisional configuration file names
- Phone asks for model-specific `sip.ld` (example `/2345-12500-001.sip.ld`)
  If it has a different version, it downloads and saves the new version; and then reboots.

Provision sequence

- Phone starts the SIP application and query DHCP again
- Phone asks for model-specific `bootrom.ld` again
- Phone asks for *<MAC>*`.cfg` again
- Phone asks for model-specific `sip.ld` again
- Phone asks for the files from *<MAC>*`.cfg` in order, generally:
  - `/server/cert/ca.cfg` - CIC certificate authority for use with TLS connections
  - `/phone/`*<guid>*`.cfg` - custom configuration for that managed phone
  - `/proxy/`*<registration_group>*`.cfg` - registration information for the phone's registration group
  - `/server/xic.cfg` - dialplan and feature-specific configuration
  - `/phone1.cfg` - Polycom-default configuration
  - `/sip.cfg` - Polycom-default configuration
  - `/overrides/`*<MAC>*`-phone.cfg` - phone-specific configuration that a user changed from the phone's user interface
- Phone asks for default files; it's possible that these files are not available
  - `/`*<MAC>*`-license.cfg` - phone-specific license
  - `/contacts/`*<MAC>*`-directory.xml` - contacts directory
- Phone REGISTERS with the CIC server

You can verify proper provisioning and registration using either a Wireshark capture, or the combination of Provision Server and SIPEngine logs.

Precedence example

Polycom phones give precedence to whatever configuration file it acquires first; therefore, any configuration received in `server/xic.cfg` overrides configuration received in `sip.cfg`.

As an example, say that a phone has the `CUSTOM::config_files` attribute set to `config_me.cfg`. Assuming that the file exists, the phone requests this `config_me.cfg` immediately before it requests `phone1.cfg`. The contents of `config_me.cfg` are:

```
<?xml version="1.0" standalone="yes"?>
<mb>
<idleDisplay>
<home mb.idleDisplay.home="http://www.inin.com"/>
<refresh mb.idleDisplay.refresh="4"/>
</idleDisplay>
reg.1.address="blahblahblah"
</mb>
```

The phone ignores the `reg.1.address` attribute configuration, since it receives it before this file in `phone/<guid>.cfg`. However, the phone doesn't receive the other two attributes until "sip.cfg", so the phone applies these two attributes accordingly.

Phone simulator

A quick way to check what configuration passes to a phone is to use `PhoneSim.exe`, a PureConnect tool that simulates the provisioning requests for managed IP phones. The tool is available on the [Utilities and Downloads page](#). All the configuration attributes display and allow filtering by file, attribute name, or attribute value. The system saves the configuration files that the provisioning server passes in the specified output directory.

This tool applies configuration settings according to file precedence. To determine whether the system applies a specific attribute from the proper file, see the filename associated to that file.
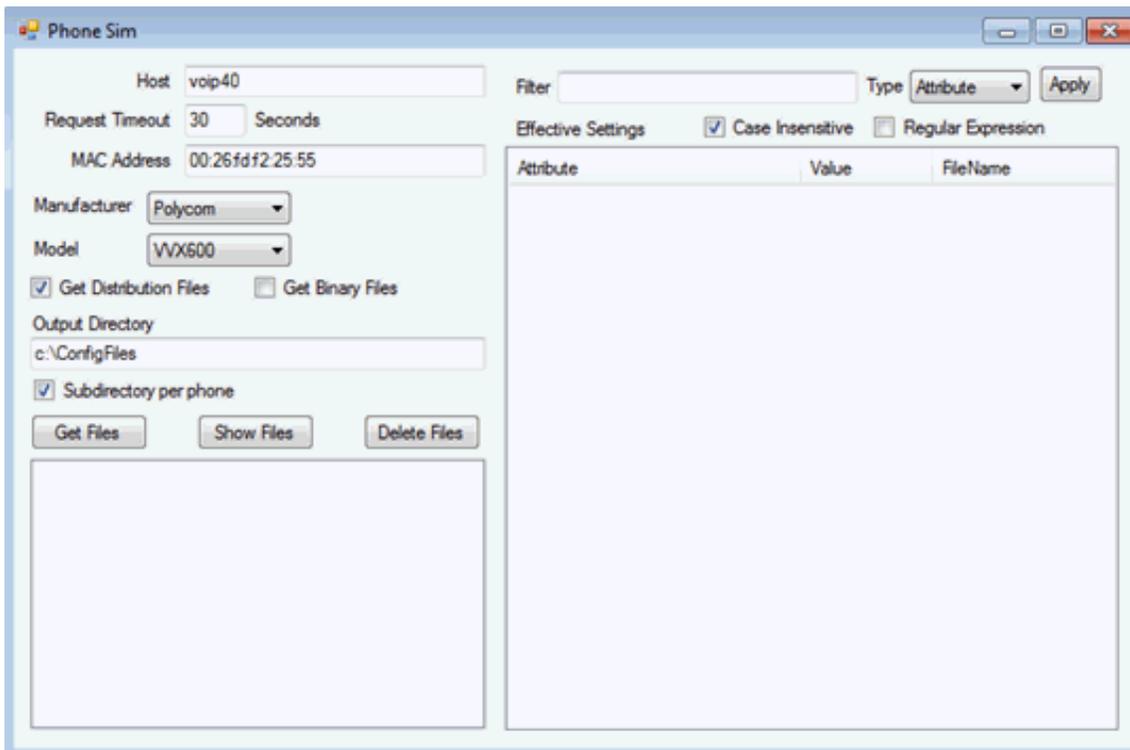
Interaction SIP Station I and II Administration

Set Up Interaction SIP Station I and II Phones

Before setting up one or more Interaction SIP Station I and II phones, ensure that you completed the network configuration for Interaction SIP Station I and II.

Unpack the package contents

Your Interaction SIP Station I or II package includes the following items. Ensure that all these parts are available in the box before you proceed.

- One Interaction SIP Station I or II
- One Ethernet patch cable
- A desk-mounting plate with two desk-mounting screws (Interaction SIP Station I only)
- A headset hanger (Interaction SIP Station I only)

When unpacking, ensure that all the following items are present and undamaged. If anything is missing or broken, contact the distributor from whom you purchased the phone for assistance.

Interaction SIP Station Phone I with hanger	Two Desk-Mounting Screws
Ethernet Patch Cable	Desk Mounting Base Plate

Fasten the phone to the desk (optional-Interaction SIP Station I only)

Optionally, you can secure the Interaction SIP Station I to a desk or other appropriate service using the included base plate that you can mount to the bottom of the phone with the two screws provided.

To attach the base plate to the phone

1. On the bottom of the phone, remove the two screws in the center.
2. Optionally, insert the hanger before mounting the plate in step 3 (hanger not shown in the diagram).
3. Using the provided screws, attach the base plate to the bottom of the phone, as shown.

> **Note:**
> The indented side of the holes faces out.

---

4. Place the phone on the mounting surface and mark the surface through the holes on each end of the base plate.
5. Drill or punch a small hole on these marks to receive the screws.
6. Place the base plate over the holes and fasten it to the surface.

For instructions on using Interaction SIP Station, see the *Interaction SIP Station Quick Reference card*.

Connect the network cable

This procedure describes how to cable your phone. After you connect the network cable, the Status LED flashes orange as it cycles through the start-up process. Genesys recommends provisioning the phone through Interaction Administrator.

To connect the network cable
1. Connect a RJ-9 headset to the RJ-9 headset jack.
2. Connect the PoE LAN port on the Interaction SIP Station I or II to your available LAN jack using the provided Ethernet patch cord.
3. If you have only one LAN jack, you can use the included PC port on the Interaction SIP Station I or II to provide a LAN connection to your PC.

Interaction SIP Station I and II Firmware for Supported Phones

The following tables list the Interaction SIP Station I and II firmware versions for supported phones.

Firmware for supported phones

Firmware Version	Initially Made Available in...	Phone Model
2.0.4.15.7	2015 R2	Interaction SIP Station I Interaction SIP Station II

Managed IP Phone (Interaction SIP Station I and II) Configuration Options

The following options reflect the state of Interaction SIP Station I and II configuration as of CIC 2015 R2.

Options tab (SIP Station I and II)

These configuration options are available from the Managed IP Phones container when editing one or more managed Interaction SIP Station I and II phone(s). This tab allows for configuration of speed dial (Interaction SIP Station 1 only), emergency information, among other options. For more information about these configuration options, see the *Interaction Administrator Help*.

Advanced Options (Interaction SIP Station I and II)

The Advanced Options are available from the **Advanced Options** button in the **Options** tab, if configuring just one phone, or in the **Advanced Options** tab, if configuring more than one phone. This tab allows configuration of provisioning, syslog, gains, LAN and VLAN, and audio quality diagnostics. For more information about these configuration options, see the *Interaction Administrator Help*.
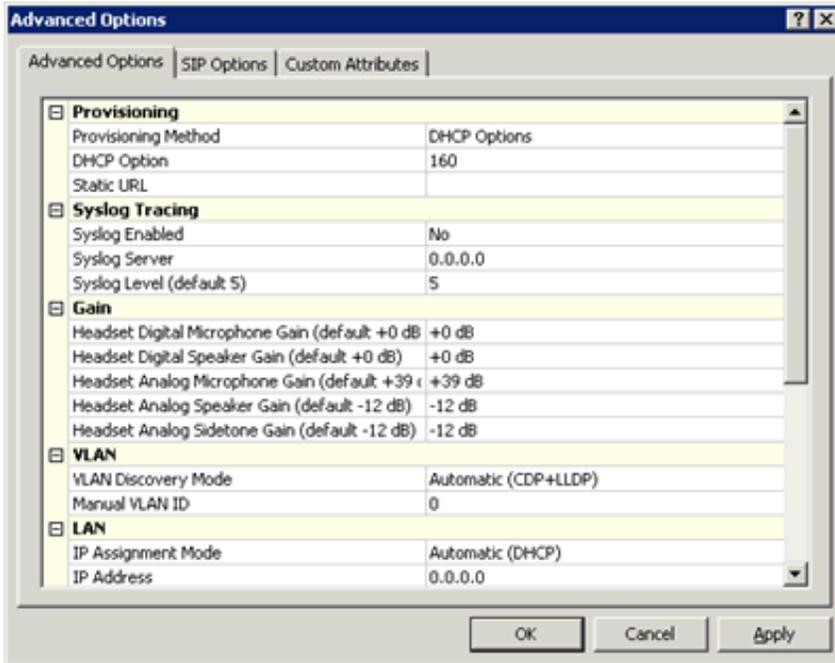
Other Managed IP Phone (Interaction SIP Station I and II) Features

Following are two unique features that Interaction SIP Station phones offer over other Managed IP Phones, supported by CIC: Configurable speed dials and LED status light.

Configurable speed dials (Interaction Station 1 only)

Interaction SIP Station I phones can have two speed-dial numbers configured in the **Options** tab in Interaction Administrator. They can map to any valid number or extension. **Button 1 Speed Dial** maps to the *exclamation point* (`!`) key on the Interaction SIP Station I and defaults to `911`. Generally, you leave this key at its default to allow placing emergency calls. **Button 2 Speed Dial** maps to the pickup key and defaults to `*`. To initiate a speed-dial call, press and hold the appropriate key until the call places and connects.

LED status light

Interaction SIP Station phones I and II have a multi-color LED in the center of the phone. This light indicates various states for the phone by color and flashing.

In general, if the LED is solid blue, the Interaction SIP Station can receive calls. If it is not solid blue, it cannot receive calls. If the phone does not have an active call and you adjust the volume, the LED flashes once for each button press: orange if adjusting the volume down, red if adjusting the volume up. When the CIC server signals a reboot, the LED flashes orange rapidly before proceeding to the boot sequence.

Boot sequence

When the phone boots up, the light is solid purple and then solid orange while the phone powers on. When the phone starts querying DHCP and getting its configuration, the LED flashes orange quickly. If at this point the Interaction SIP Station obtains new firmware, it downloads the firmware and then transitions to solid orange as it saves the new firmware before rebooting. Once the phone obtains its configuration, the LED transitions to solid orange pending registration. If the registration fails, the LED blinks orange slowly. If the registration succeeds, the LED is solid blue.

	Boot Status		2 seconds
1	Power up		
2	Linux boot up		
3	Linux boot up end		
4	Restore to default indication during boot up (Around 15 seconds after power up)		
5	CDP/LLDP discovery		
6	Acquiring IP Address		
7	Idle (Not registered)		
8	Downloading new firmware.		
9	Upgrading firmware.		
10	Stuck (Boot loader was damaged during upgrade. Unit must RMAed.)		
11	Infinite loop in recovering mode (Waiting for options 66/67)		
12	Obtained 66/67 and downloading firmware		
13	Upgrading in recovery mode		
14	Registration in progress		
15	Registration succeed		
16	Idle (Registered).		
17	Registration failure (Incorrect password or can't reach the proxy).		

Call sequence

When the phone receives a call, it flashes red to indicate that a call is waiting for you to answer. Once you pick up the call, it transitions to blinking blue, indicating that a call is in progress. If you place the call on hold, the LED continues to flash blue. If you adjust the volume of a call, the LED flashes orange once for each button press. If you mute the phone's microphone, the LED transitions to solid red until either the call disconnects or you un-mute the microphone. Once the call disconnects, the LED returns to solid blue.

	Call Status (no change)	
18	Alerting	
19	Active call	
20	Mute (During active call).	
21	Volume key pressed	
22	Volume key pressed (Min/Max volume level reached)	

Phone Simulator

Phone Simulator - Simple but powerful tool that allows you to point to an IC Server and emulate a Polycom, Interaction SIP Station, or AudioCodes 420HD phone's requests for configuration files. Also, the simulator performs all inheritance calculations and displays the result for all attributes in a searchable/filterable pane.

Troubleshooting (Interaction SIP Station I and II)

Troubleshooting addresses two of the more common issues with the Interaction SIP Station I and II. Whenever troubleshooting a network issue with the Interaction SIP Station, packet captures are invaluable.

Cannot hear audio through headset

The Jabra GN 1200 headset supports a variety of telephones with different models requiring different switch settings. If you cannot hear audio through the headset (for example, no ring tone when you press the volume up button), check the switch position on the inline switch on the headset. Interaction SIP Station I and II use switch position 1

Interaction SIP Station cannot start or connect to the network

If you specified incorrect network settings, such as an incorrect VLAN ID or static IP address settings, to a point to where the Interaction SIP Station unit can no longer start or connect to the network (as verified through packet captures), you can reset the unit to factory default network settings.

> **Warning!**
> Only the network administrator should do this procedure, not an agent.

To reset Interaction SIP Station to factory default network settings

1. Disconnect the network connection from the unit, which, in turn, removes power from the unit.
2. Wait a few seconds.
3. Do one of the following:
   - For Interaction SIP Station I, press and hold both round buttons on the top of the unit.
   - For Interaction SIP Station II, press and hold the `Mute` and `Volume down` buttons.
4. Plug in the network connection into the unit. The unit restarts.
5. Continue holding the two round buttons until the indicator light on the unit blinks three times with a purple color. The unit resets to factory default network settings.

Boot and Provision Sequences (Interaction SIP Station I and II)

Unlike a Polycom phone, the Interaction SIP Station I and II do not have separated boot and provision sequences. The unit powers on and then goes directly into the provisioning sequence.

Firmware boot/provisioning sequence

Following is the firmware boot and provisioning sequence for SIP Station I and II:

- Phone powers on and does CDP and/or LLDP discovery, depending on configuration
  - Gets port-mode configuration
  - Gets VLAN information (if any)
- Phone runs through DHCP discovery:
  - DHCP *must* return:
    - IP Address
    - Option 1 - Subnet Mask
    - Option 15 - DNS server
  - DHCP *should* return:
    - Option 3 - Router to contact devices outside of subnet
    - Option 66 - TFTP Server Address
    - Option 67 - Bootfile Name (usually `sip100.img`)
    - Option 160 - Provision server address (usually the CIC server or a provisioning proxy)
  - DHCP *may* return:
    - Option 4/42 - Network time servers
    - Option 2 - Time offset
- Phone queries DNS for the provision server found in Options 160
  - If not found, the phone boots with its old configuration
- Phone sends HTTP GET request for the Bootfile to the Provision Server
  - It pulls a portion of the file and checks whether it matches its current firmware
  - If it doesn't match, the phone GETS the complete file, flashes the firmware to its memory and reboots
- Phone sends HTTP GET request for *<MAC>*`.cfg` to the Provision Server
- Phone attempts to register with its configured SIP Proxy
- If the configuration file changed, the phone reboots

Phone simulator

A quick way to check what configuration passes to a phone is to use `PhoneSim.exe`, a PureConnect tool that simulates the provisioning requests for managed IP phones. For more information, see "Phone simulator" in [Boot and Provision Sequences (Polycom)](#).

SIP Soft Phone Administration

SIP Soft Phone Requirements

CIC supports the SIP Soft Phone on workstations running on **Microsoft Windows 7** (32-bit and 64-bit), **Microsoft Windows 8** (32-bit and 64-bit), and **Microsoft Windows 8.1**(32-bit and 64-bit), which support the PureConnect QoS driver.

> **Notes:**
> - CIC doesn't support the SIP Soft Phone on workstations running Microsoft Windows XP (32-bit and 64-bit), which does not support the PureConnect QoS driver.
> - CIC 2016 R2 and later supports Microsoft Windows 10.
> - CIC 2015 R1 and later supports Microsoft Windows 8.1.
> - CIC 2015 R1 and later *no longer* supports Microsoft Windows XP.

SIP Soft Phone Installation

The SIP Soft Phone is a selectable feature on the **Custom Setup** page of the **IC User Applications install (32-bit and 64-bit)**.
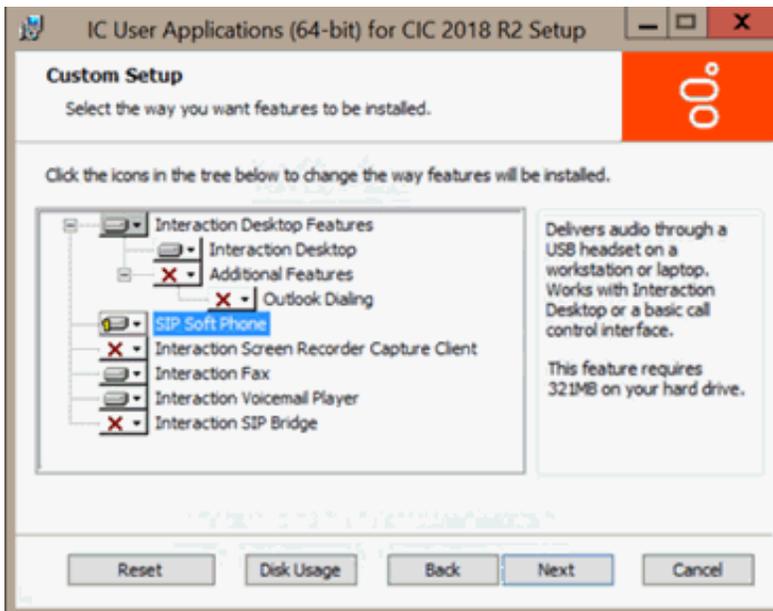
Ensure that you select this feature when installing IC User Applications on workstations that use the SIP Soft Phone. If not installed during initial installation, click **IC User Applications** in the **Control Panel -> Programs and Features**, and then click **Change** to open the **Custom Setup** page. Click the **SIP Soft Phone** feature.

On the **QoS Requirement** page, click **PureConnect QoS driver** to add the PureConnect certificate to the Trusted Publishers list and install the PureConnect QoS driver.
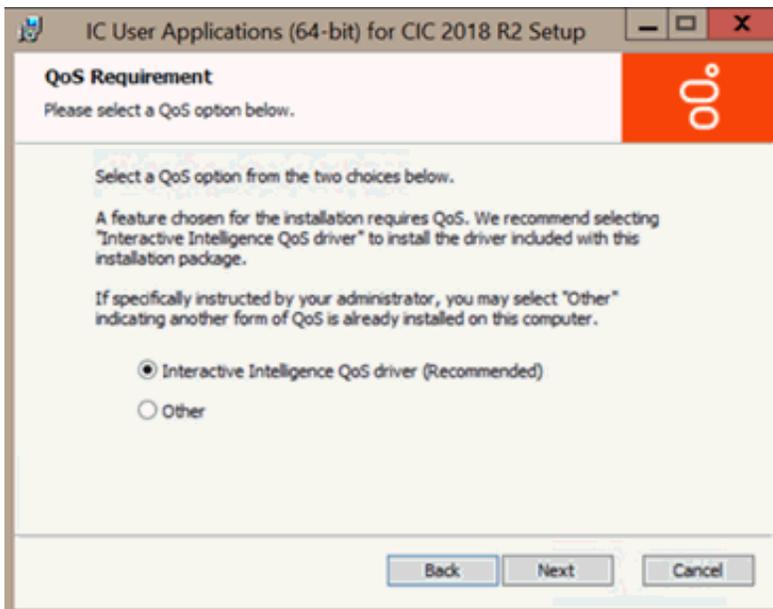
For more information:

- For instructions on installing IC User Applications, see the *PureConnect Installation and Configuration Guide*.
- For more information about the PureConnect QoS driver, see Implement QoS in Your Environment and the PureConnect KB article.

SIP Soft Phone Audio Device Requirement

For audio devices, audio quality and personal preference are subjective. When you select a USB headset, choose a quality headset with noise and echo cancellation. You can discuss personal experience with headsets and recommendations from other partners and customers on the PureConnect Community Forum.

SIP Soft Phone Setup Process

If you followed the instructions to configure the network for SIP Soft Phones and to create multiple Managed IP Phones using Managed IP Phones Assistant (with the full computer names filled in the CSV list), **automated provisioning occurs when the SIP Soft Phone starts on a workstation.**

Following are **other required or recommended setup procedures**, depending on the circumstances:

- If the workstation contains more than one NIC or if the NIC was altered following the SIP Soft Phone installation, specify the **network adapter** to use with the SIP Soft Phone. For more information, see SIP Soft Phone Network Adapter Configuration and Auto-detection.
- Genesys highly recommends that SIP Soft Phone users configure their SIP Soft Phones to **use the default devices configured in the Windows Multimedia Control Panel**. For more information, see SIP Soft Phone Audio Configuration.
- If you installed **CIC Language Packs** on the CIC system, SIP Soft Phone users can display the user interface in another language. For more information, see Set the User Interface Language for SIP Soft Phone.
- If one of the following is true, you might need to provision the phones manually using the **SIP Soft Phone Provisioning wizard**:
  - There is no DHCP server.
  - The system cannot access the DHCP server.
  - You didn't specify the full computer name in the CSV list for some or all SIP Soft Phones.

  For more information, see SIP Soft Phone Provisioning Wizard.

SIP Soft Phone Network Adapter Configuration and Auto-detection

If the workstation contains more than one NIC or if the NIC was altered after the SIP Soft Phone installation, specify the **network adapter** to use with the SIP Soft Phone after the phone is provisioned. You can configure the SIP Soft Phone to use a specific network adapter for Real-time Transfer Protocol (RTP) and SIP communications.

To specify the network adapter

1. Right-click the SIP Soft Phone icon in the system tray and then click **Options**.
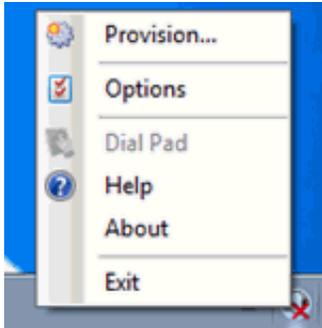
2. In the **Options** dialog box, click the adapter that best suits the RTP or SIP traffic.
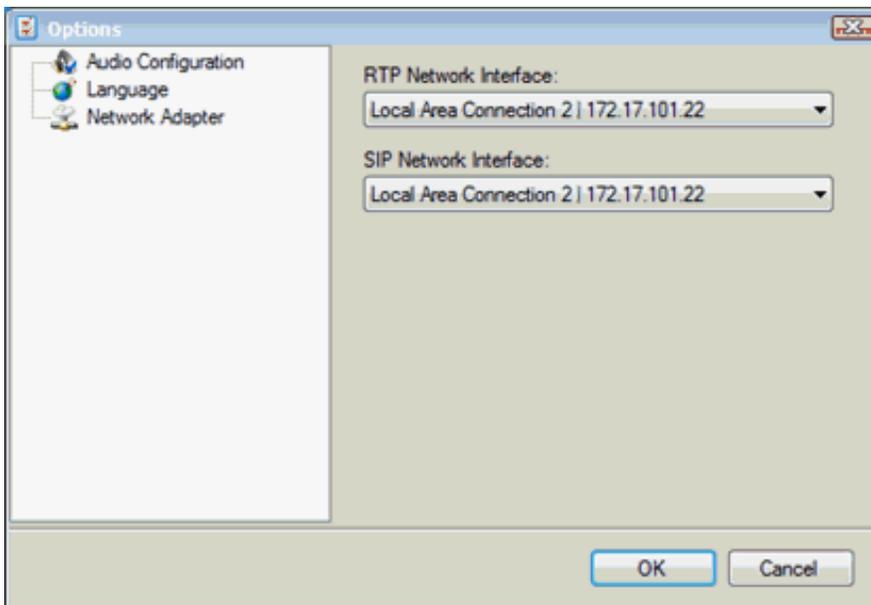
Auto-detection of changes to the network adapter

The SIP Soft Phone detects when a selected network adapter for SIP or audio connection is unavailable or disconnects; and then terminates the connections. When the selected adapters are available again, the SIP Soft Phone establishes the connections again.

SIP Soft Phone Audio Configuration

Genesys highly recommends that SIP Soft Phone users configure their SIP Soft Phone to **Use default communication devices**, which is the default configuration set in the Windows Multimedia Control Panel.

To configure the audio device

1. Right-click the SIP Soft Phone icon in the system tray and the click **Options**.
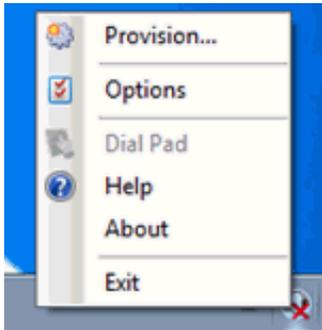
2. In the **Options** dialog box, click **Audio Configuration**.
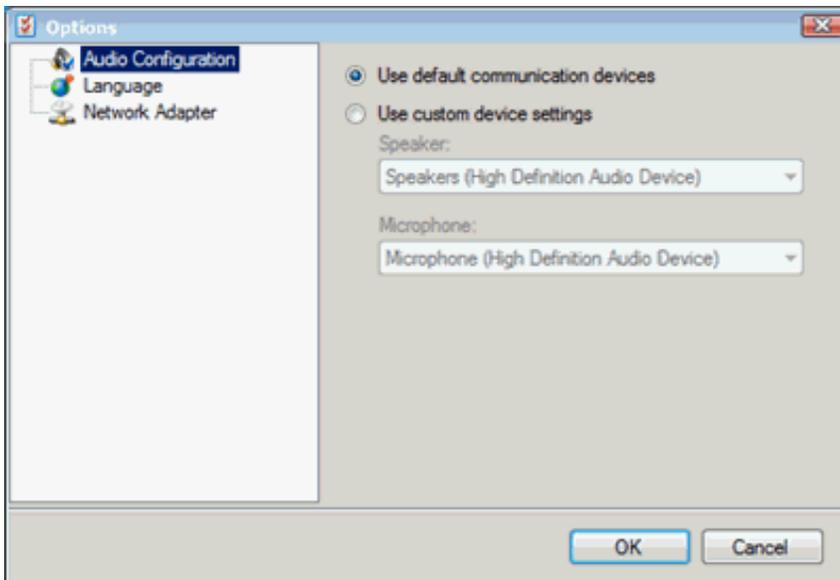
3. Click **Use default communication devices** to use default configuration set in the **Windows Multimedia Control Panel**.

    The administrator needs to instruct users to configure the Windows Multimedia Control Panel to set the default speakers and microphone to a USB headset device:

    a. Right-click the **Speaker** icon in the notifications area of the Windows taskbar and then click **Sounds**.
    b. Select and configure the headset's speaker on the **Playback** tab.
    c. Select and configure the headset's microphone device on the **Recording** tab.

    Alternatively, users can select **Use custom settings** on the **Audio Configuration** page to specify the speakers and microphone to use.

Set the User Interface Language for SIP Soft Phone

If you installed **CIC Language Packs** on the CIC system, SIP Soft Phone users can display the user interface in another language.

To configure the audio device

1. Right-click the SIP Soft Phone icon in the system tray and then click **Options**.
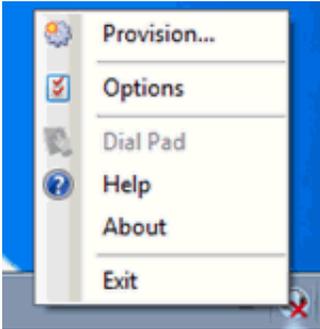
2. In the **Options** dialog box, click **Language**.
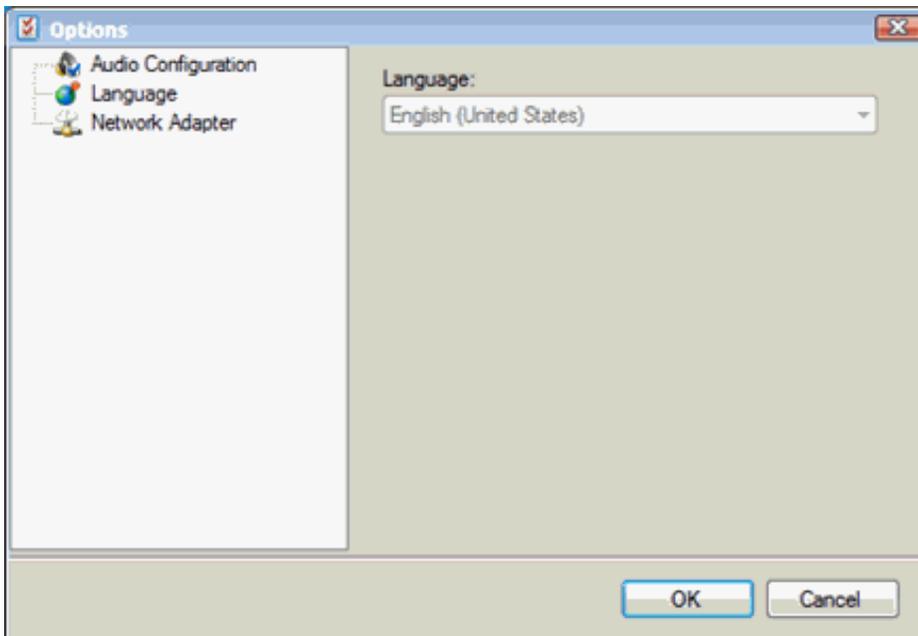
3. In the **Language** list box, click the appropriate language for the user interface.
4. Exit and restart the SIP Soft Phone.

SIP Soft Phone Provisioning Wizard

If there is no DHCP server, the system cannot access the DHCP server, or you didn't specify the full computer name in the CSV list for some or all SIP Soft Phones, provision the SIP soft phone using the SIP Soft Phone Provisioning wizard.

To provision SIP soft phones using the SIP Soft Phone Provisioning wizard

1. Right-click the **SIP Soft Phone icon** in the system tray and then click **Provision**. The SIP Soft Phone Provisioning wizard **Welcome** page appears.

2. Click **Next**. The **Get SIP Soft Phone Configuration** page appears. Specify how to load the phone configuration file. The configuration file contains information to provision the SIP Soft Phone.

Automatic Discovery: If selected, the DHCP lookup provides the location of the provisioning server for you. If you configured the network for the SIP Soft Phone (DHCP Option 160), this option is selected by default. An information icon appears next to this option. Rest the pointer on the icon to display either the reason the option is not available, or the location of the provisioning server that the DHCP lookup provides.

Configure Settings: If selected, displays the **Phone Provisioning Server** box to allow you to specify the provisioning server. If the configuration file is not available or an error occurs during automatic discovery, you can select this option to load configuration information manually.

Phone Provisioning Server: Name of the phone provisioning server. If you select the **Configure Settings** option, this box appears.

3. Do one of the following:

- **To load the configuration file automatically**, do the following:

a. With the **Automatic Discovery** option selected, click **Next**.

b. If there are any errors when loading the configuration, an error message appears on the **Get SIP Soft Phone Configuration** page. Do one of the following:

   ○ If the configuration loads successfully and you have multiple network adapters, select the appropriate option on the **Select Network Adapters** page.

- If the configuration loads successfully and you do not have multiple network adapters, click **Finish** to exit the **SIP Soft Phone** Provisioning wizard.

- **To load the configuration file manually**, do the following:

  a. Click **Configure Settings**.
  b. In the **Phone Provisioning Server** box, type the name of the provisioning server (CIC server).
  c. Click **Next** to load the specified configuration file.
  d. Do one of the following:

       - If the configuration loads successfully and you have multiple network adapters, select the appropriate option on the **Select Network Adapters** page.
       - If the configuration loads successfully and you do not have multiple network adapters, click **Finish** to exit the SIP Soft Phone Provisioning wizard.
       - If the configuration file is not available on the server, click **Next** to complete the auto provisioning portion of the SIP Soft Phone Provisioning wizard.

4. If the Full Computer Name of the computer running the SIP Soft Phone does not match the Full Computer Name for any of the Managed IP Phones configured in the **Managed IP Phones** container in Interaction Administrator, the **Use Auto Provisioning** page appears.

This page allows you to provision the SIP Soft Phone using the provisioning IVR. The wizard initiates a call to the CIC server to do the provisioning. The server then generates the appropriate configuration and triggers the SIP Soft Phone to retrieve its new configuration.

5. If the **Use Auto Provisioning** page appears, do the following:

   a. Click **Provision**.

   > **Note:**
   > If the **Provision button** is not available, see Troubleshooting (SIP Soft Phone).

   b. Following the voice prompts, provide the requested information using the dial pad. For example, the type of provisioning (by administrator or user), the extension number for the phone, and the PIN number. Administrators need to specify both administrator and station information.

   The CIC server generates the appropriate configuration and triggers the SIP Soft Phone to retrieve its new configuration.

   c. After provisioning the SIP Soft Phone, click **Next**.

   The **SIP Soft Phone Provision Complete** page appears whenever SIP Soft Phone provisioning is complete, depending on the option used for loading the configuration file and whether provisioning was necessary.

6. Click **Finish**.

SIP Soft Phone Help

For instructions on using the SIP Soft Phone, see the *SIP Soft Phone documentation*.

Managed IP Phone (SIP Soft Phone) Configuration Options

The managed IP phone options reflect the state of SIP Soft Phone configuration as of CIC 4.0 GA.

Options tab

These configuration options are available from the Managed IP Phones container when editing one or more managed SIP Soft Phones. This tab allows for configuration of diagnostic captures and emergency information, among other options. For more information about these options, see Overview of configuration settings for managed IP phones and templates in the *Interaction Administrator Help*.

SIP Soft Phone and Remote Survivability

Remote survivability is the ability of a remote location to continue to operate in some (typically limited) capacity when the connection to the main location is down. In this scenario, the CIC server is unreachable, meaning that the CIC client application cannot function, even though the SIP Soft Phone is still running. The local SIP Proxy server and SIP gateways are available also.

For more information about remote survivability, see [Appendix B: How Registrations Work (Proxy Settings)](#).

In a remote survivability scenario

1. SIP Soft phone users can place **outbound calls** using the SIP Soft Phone dial pad. To access the dial pad, do the following:
   a. Right-click the **SIP Soft Phone icon** in the system tray and then click **Dial Pad**.
   b. Using the dial pad, you can provide phone numbers to dial and have call control over those outbound calls.

   The calls go directly from the SIP Soft Phone to the proxy server, which routes them out the local gateway.

2. SIP Soft Phone users can receive **inbound calls**, but the SIP Soft Phone cannot alert users about the calls. The SIP Soft Phone relies on the CIC client for this functionality, and CIC client is not available. Users need to have the SIP Soft Phone Dial Pad open to check for calls.

Because the SIP Soft Phone does not alert or pop for inbound calls, the SIP Soft Phone is not practical, and therefore, not supported as a remote survivability solution. Physical devices such as Polycom phones and Interaction SIP Station phones are more practical in a remote survivability situation.

Troubleshooting (SIP Soft Phone)

Troubleshooting addresses a few of the more common problem scenarios encountered with SIP Soft Phones. In these scenarios, it is useful to obtain a packet capture to see exactly what traffic is passing to and from the SIP Soft Phone, including DNS and SIP traffic.

SIP Soft Phone cannot obtain configuration from provisioning server

The SIP Soft Phone uses DHCP Option 160, which it queries from the DHCP server to obtain the name of the provisioning server. There are a few reasons why the SIP Soft Phone cannot obtain the configuration file from provisioning server.

> **Note:**
> Obtaining a packet capture from both the client and server may be a necessary troubleshooting step.

To ensure that the SIP Soft Phone can obtain configuration from provisioning server
1. Verify DHCP Option 160 configuration on the DHCP server. (For more information, see Configure DHCP Option Records.)
2. Verify through DNS that the hostname configured in DHCP resolves to the IP address of the correct CIC server. (For more information, see Configure DHCP Option Records.)
3. Verify that the Windows User account that is running the SIP Soft Phone has full access to the following registry keys:

   ```
   HKLMSoftwarePoliciesMicrosoftSystemCertificates
   HKLMSoftwarePoliciesMicrosoftCryptography
   ```

4. Verify that the Windows User account that is running the SIP Soft Phone has full access to the following file path:

   ```
   %ALLUSERSPROFILE%\Application Data\Microsoft\Crypto\Keys
   ```

5. If there is a firewall between the client workstation running the SIP Soft Phone and the CIC servers, verify that the firewall rules allow communication to the CIC servers on ports 8088 (http) and 8089 (https).

SIP Soft Phone contacts provisioning server but cannot register

If the SIP Soft Phone cannot place calls, it typically indicates that it is not registered and results in the following behavior:
- Placing outbound calls using the CIC client doesn't complete.
- The CIC client cannot pick up incoming calls.

> **Note:**
> Obtaining a packet capture from both the client and server is a helpful troubleshooting step.

To ensure that the SIP Soft Phone can register
1. Ensure that the appropriate network adapter and VPN are selected on the **Options** page. For more information, see SIP Soft Phone Requirements.
2. Ensure that the audio device is working properly by running the **Audio Tuning Wizard** in Interaction Desktop. For more information, see Audio Tuning Wizard.
3. Ensure that the SIP Soft Phone can resolve the name of its proxies correctly (through DNS). (For more information, see Appendix B: How Registrations Work (Proxy Settings).
   Verify that the SIP Soft Phone resolves the applicable DNS record correctly (for example, DNS SRV, server hostname, SIP Proxy name).

Startup and Provision Sequences (SIP Soft Phone)

The behavior of a SIP Soft Phone upon powering on is helpful while troubleshooting a phone to understand what it is doing during startup.

Startup sequence

There are two ways to start the SIP Soft Phone. You can start the SIP Soft Phone by manually running it from the desktop shortcut;

or by attempting to log on to CIC client and then selecting SIP Soft Phone as the station type. (Only a single instance of the SIP Soft Phone can run at a time.) In either case, the startup sequence is the same.

In the startup sequence, the SIP Soft Phone attempts to:

- Initialize the audio input and output devices based on registry keys:

  - ```
    HKEY_CURRENT_USER\
    Software\
      Interactive Intelligence\
        SIP Soft Phone\
          <fullcomputername>\
            4.0 Selected Audio Input Device
    ```

  - ```
    HKEY_CURRENT_USER\
    Software\
      Interactive Intelligence\
        SIP Soft Phone\
          <fullcomputername>\
            4.0 Selected Audio Output Device
    ```

- Bind to the network adapter based on a registry key:

  ```
  HKEY_CURRENT_USER\
   Software\
     Interactive Intelligence\
       SIP Soft Phone\
         <fullcomputername>\
           4.0 Selected Network Adapters
  ```

- Download a configuration file based on a registry key:

  ```
  HKEY_CURRENT_USER\
   Software\
     Interactive Intelligence\
       SIP Soft Phone\
         <fullcomputername>\
           4.0 ConfigUrl
  ```

- If the previous step fails, the SIP Soft Phone attempts to download a configuration file based on a DHCP Option 160.

Provision sequence

- If the SIP Soft Phone provisioned successfully during a previous startup, it looks in the registry first for a key that specifies the configuration file.

  ```
  HKEY_CURRENT_USER\
   Software\
     Interactive Intelligence\
       SIP Soft Phone\
         <fullcomputername>\
           ConfigUrl
  ```

  (for example, https://provision.lab.voip:8088/AgentPC1.lab.local.i3sipcfg)
- If the configuration file registry key does not exist, the SIP Soft Phone queries the DHCP server for Option 160 to determine whether a boot server definition exists and then attempts to download the configuration file.
- If that option does not define a boot server, you need to provision the SIP Soft Phone manually using the SIP Soft Phone Provisioning wizard.

Phone simulator

A quick way to check what configuration passes to a phone is to use `PhoneSim.exe`, a PureConnect tool that simulates the provisioning requests for managed IP phones. For more information, see "Phone simulator" in <u>Boot and Provision Sequences</u>

(Polycom).

AudioCodes and Genesys Phone Administration

AudioCodes and Genesys Firmware for Supported Phones

The following table lists the AudioCodes and Genesys firmware versions for supported phones.

Firmware for supported phones

Firmware Version	Initially Made Available in...	Phone Model
2.2.16.643	2023r3 2023r2 Patch 6 2023r1 Patch 14 2022r3 Patch 18 2022r2 Patch 25 2022r1 Patch 33 2021r4 Patch 39	AudioCodes 405 405HD 420HD 430HD 440HD Genesys 405 405HD 420HD
2.2.16.408	2020 R4 2020 R3 Patch 2 2020 R2 Patch 8 2020 R1 Patch 14 2019 R4 Patch 21 2019 R3 Patch 27	AudioCodes 405 405HD 420HD 430HD 440HD Genesys 405 405HD 420HD
2.2.16.142.41	2019 R3 2019 R2 Patch 4 2019 R1 Patch 10 2018 R5 Patch 16 2018 R4 Patch 22 2018 R3 Patch 29 2018 R2 Patch 35	AudioCodes 405 405HD 420HD 430HD 440HD Genesys 405 405HD 420HD

2.2.16.142	2018 R5 Patch 5	**AudioCodes**
	2018 R4 Patch 11	405
	2018 R3 Patch 18	405HD
	2018 R2 Patch 24	420HD
	2018 R1 Patch 31	430HD
	2017 R4 Patch 37	440HD
		Genesys
		405
		405HD
		420HD
2.2.8.61	2017 R2	420HD
2.2.277.2.3	IC 4.0 SU 6	420HD
2.0.0.18_6	IC 4.0 SU 5	420HD

Managed IP Phone (AudioCodes and Genesys Phones) Configuration Options

AudioCodes and Genesys phones work with the CIC provisioning subsystem and you configure them in Interaction Administrator in the same way as Polycom phones, Interaction SIP Station I and II phones, and SIP Soft Phones. You *must* implement each AudioCodes or Genesys phone as a managed IP phone.

AudioCodes and Genesys phones are similar to the Interaction SIP Station I and II phones in their network and Interaction Administrator configuration. However, they have a headset, LCD display, a dial pad, and more function keys. Interaction SIP Station I has none of these features; Interaction SIP Station II has a dialpad and more function keys only.

Options tab

These configuration options are available from the Managed IP Phones container when editing one or more managed AudioCodes or Genesys phones. For more information about these options, see the *Interaction Administrator Help*.
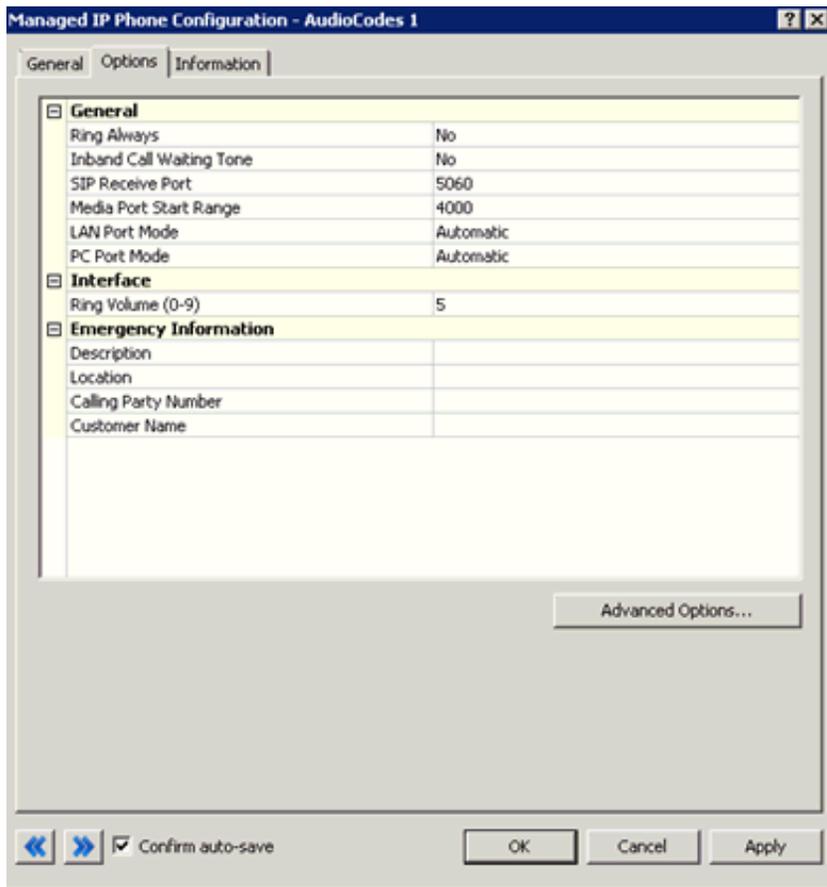
Advanced options

The Advanced Options are available from the **Advanced Options** button in the **Options** tab, if configuring just one phone, or in the **Advanced Options** tab if configuring more than one phone. This tab allows configuration of provisioning, syslog, gains, LAN and VLAN, and audio quality diagnostics. For more information about these options, see the *Interaction Administrator Help*.
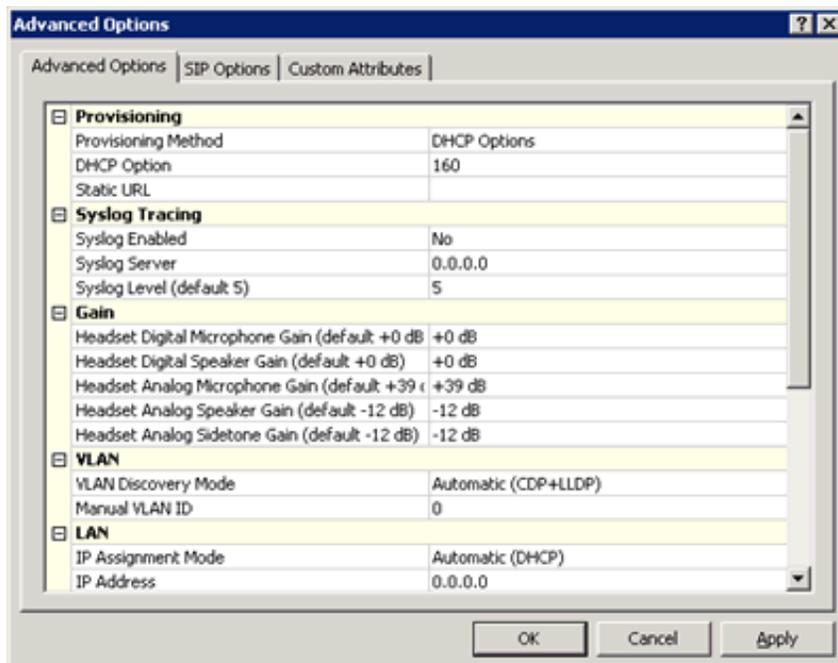
Other Managed IP Phone (AudioCodes and Genesys Phone) Features

Download files

- [Customizing Call Progress Tones](#) - Contains "How do I customize my Call Progress Tones file.doc", "usa_tones_09_no_hold.dat", and "usa_tones_09_no_hold.ini". See instructions in the ".zip" file.
- [Sample Call Progress Files](#) - Contains sample files and instructions.
- [AC Tools](#) - Contains instructions and a utility to allow logging on the AudioCodes gateway.
- [Phone Simulator](#) - Simple but powerful tool that allows you to point to an IC Server and emulate a Polycom, Interaction SIP Station, or AudioCodes 420HD phone's requests for configuration files. Also, the simulator performs all inheritance calculations and displays the result for all attributes in a searchable/filterable pane.

For more information about the AudioCodes Phone Series, see the [AudioCodes website](#).

Boot and Provision Sequences (AudioCodes and Genesys Phones)

The behavior of AudioCodes and Genesys phones when powering on is helpful while troubleshooting a phone to understand what it is doing during startup. Unlike a Polycom phone, AudioCodes and Genesys phones do not have separate boot and provision sequences. The units power on and then go directly into the provisioning sequence.

Firmware boot/provisioning sequence

Following is the firmware boot and provisioning sequence for AudioCodes and Genesys phones:
- Phone powers on and does CDP and/or LLDP discovery, depending upon configuratio
  - Gets port-mode configuration
  - Gets VLAN information (if any)
- Phone runs through DHCP discovery
  - DHCP *must* return:
    - IP Address
    - Option 1 - Subnet Mask
    - Option 15 - DNS server
  - DHCP *should* return:
    - Option 3 - Router to contact devices outside of subnet
    - Option 66 - TFTP Server Address
    - Option 67 - Bootfile Name (usually 420HD.img)
    - Option 160 - Provision server address (usually the CIC server or a provisioning proxy)
  - DHCP *may* return:
    - Option 4/42 - Network time servers
    - Option 2 - Time offset
- Phone queries DNS for the provision server found in Option 160
  - If not found, the phone boots with its old configuration
- Phone sends HTTP GET request for the Bootfile to the Provision Serve
  - It pulls a portion of the file and checks whether it matches its current firmware
  - If it doesn't match, the phone GETS the complete file, flashes the firmware to its memory, and then reboots
- Phone sends HTTP GET request for other config files listed with the `CUSTOM::config_files` attribute specified
- Phone sends HTTP GET request for `<MAC>.cfg` to the Provision Serve
- Phone attempts to register with its configured SIP prox
- If the configuration file changed, the phone reboot

Custom configuration files

The provisioning server supports the ability to add a custom attribute to AudioCodes and Genesys phones named `config_files` (case-insensitive), which must contain a comma-separated list of files. If the file exists in the `\i3\ic\provision\AudioCodes` directory on the CIC server, provisioning includes the files in the `config_files` list for AudioCodes phones. If the file exists in the

`\i3\ic\provision\Genesys` directory on the CIC server, provisioning includes the files in the `config_files` list for Genesys phones.

The system requests these files after provisioning generates the config files and allows administrators to set custom config file attributes that provisioning doesn't handle.

Following is the format of the custom configuration files.

The Configuration file is a `*.cfg` file with the file name specified in the list under the `config_files` custom attribute.

Ensure that the configuration file adheres to the following guidelines:
- No spaces on either side of the equals (=) sign.
- Each parameter is on a new line.
- Each parameter has the following syntax: *<parameter name>= <value>*

Following is an example of part of a configuration file:

```
voip/line/0/enabled=1
voip/line/0/id=1234
voip/line/0/description=310HD
voip/line/0/auth_name=1234
voip/line/0/auth_password=4321
```

Phone simulator

A quick way to check what configuration passes to a phone is to use `PhoneSim.exe`, a PureConnect tool that simulates the provisioning requests for managed IP phones. For more information, see "Phone simulator" in Boot and Provision Sequences (Polycom).

AudioCodes Documentation

Gateway Backup and Restore - How to backup and restore your gateway's ini file

Configure Audiocodes Gateway Setup - How to set up AudioCodes Gateway with 7.2+ firmware

Configure AudioCodes FXS and FX0 Gateway - How to configure AudioCodes FXS and FXO Gateway with 6.0+ firmware

Configure AudioCodes Mediant 1000 or 2000 - How to configure the Mediant 1000 or 2000 Gateway

Appendixes

Appendix A: Non-standard Provisioning Scenarios

This appendix describes several non-standard provisioning scenarios when there is no DHCP server or the system cannot access the DHCP server.

Remote phones

If you deploy a managed IP phone at a remote site, it is possible to avoid excess WAN traffic associated to provision phones. The Interaction SIP Proxy offers a Provision Proxy feature when using CIC to manage it. Phones can send their HTTP provisioning requests to the SIP Proxy, instead of over the WAN to the CIC server.

When an HTTP request comes in from a phone, the SIP Proxy forwards the HTTP requests to the CIC server and then caches whatever the Provision Server returns. The next time a request comes in for the same file, the SIP Proxy checks with the Provision Server to see whether the file changed, and then returns the file to the phone.

You enable the Provision Proxy feature from the **System -> CIC Integration** tab in the SIP Proxy web interface.

Configure a Polycom phone's boot server manually

On a Polycom phone, follow these steps to point the phone to the CIC (boot) server. As needed, use the buttons and arrow keys next to the choices displayed on the phone's LCD screen.

> **Note:**
> Steps for some phones differ slightly from those described here, and steps often differ depending on whether a phone is just starting or running already. If these steps don't match your phone, see the Polycom documentation for instructions about how to configure your phone.

To configure a Polycom phone's boot server
1. On the phone's **Network Configuration** screen, do the following:
   a. Press the **Menu** button, press `3` for **Settings,** and then press `2` for **Advanced**.
   b. Key in the password (the default is `456`) and then press the `Enter` button.
   c. Press `1` for Admin Settings and then press `1` for **Network Configuration**.
2. On the screen where you specify the CIC server's address, do the following:
   a. Choose **DHCP Menu** and then press `Select`.
   b. Choose **Boot Server** and then press `Edit`.
   c. Use the arrow keys to select **Static.**
   d. Press `OK` and then press `Exit` to go up a level.
   e. Use the arrow keys to select **ServerMenu** and then press `Select`.
   f. Select **ServerType:**, use the arrow to select **HTTP**, and then press `OK`.
   g. Select **ServerAddr:** and then press `Edit`.

3. Specify the CIC server's URL and the provisioning port number:

   a. If needed, press the phone's Alphanumeric toggle button (see previous image) to toggle from letters to numbers. Some phones differ slightly from the one shown here.

   b. Provide the CIC server's computer name or IP address, then a colon, and then port number **8088**.

   An example is `10.20.0.157:8088`. **For IP300/IP500 models, specify the full URL.**

   > **Note:** Specify letters by pressing the corresponding number key multiple times:
   > - For a period (`.`), press the `*` key once.
   > - For a colon (`:`), press the `#` key twice.
   > - For a slash (`/`), press the `#` key three times.
   > - If these instructions do not work on your phone, see the Polycom documentation for your phone.

4. Press **OK**, **Exit**, and then **Save** to reboot the phone and connect with the designated server.

Configure an Interaction SIP Station through the web interface

The Interaction SIP Station has a web interface to allow configuration in the instance of inability to connect to the DNS or a provisioning server. Do not use the web interface with CIC Managed IP Phones.

Firmware Version	Web Interface Address
1.2.2	http://*<ISS_ip_addr>*:80

The default user name for the interface is `admin` and the default password is `1234`. The web interface allows for configuration of network parameters; speed dial; and diagnostic tools, and allows for manual updating of the phone's configuration.
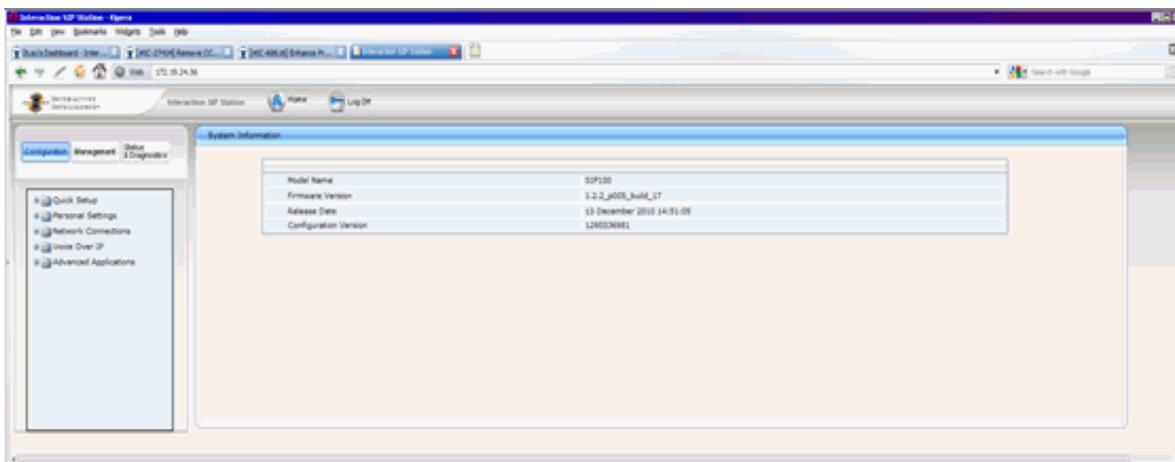
Appendix B: How Registrations Work (Proxy Settings)

This appendix describes how registration groups function.

Registrations overview

When you configure a phone to use a particular registration group, all station appearances on that phone use the same SIP registration settings, according to the contents and ordering of the registration group. Each "registration" serves as a mapping to a hostname (or IP address), a port, and a protocol for use when the phone registers. Configuring more than one registration in a registration group serves to provide SIP redundancy according to two paradigms: **failover** and **fallback**.

Registration types

Each registration group consists of an ordered list of one or more registrations of the following types:
- **Line:** The most basic registration type, the Line allows for selection of a line that you defined for the particular CIC server. All the transport settings derive from the configuration of the selected line. If there is a Line within the registration group, ensure that the Line is the first entry. You can only have **one** Line per registration group.

- **Manual:** Manual registrations allow for administrator-specified registrations. You must specify the address, port, and transport protocol. The address is either a hostname, FQDN, or IP address. It can point to anything, but use caution. Customers used manual registrations for secondary SIP proxies, primarily, but with the ability to use a **Managed Proxy**, the use of manual registrations declined.
- **Managed Proxy:** The Managed Proxy feature of the Interaction SIP Proxy allows phones in a remote location to register with the SIP proxy while maintaining some functionality of, and communication with, the CIC server. You must specify the transport protocol, but the system derives the contact address and port from the configuration of the associated Managed Proxy. To function properly, ensure that the managed phone and its Managed Proxy are in the same CIC location. Customers use Managed Proxies for remote survivability situations. For more information, see Remote survivability (fallback).
- **DNS SRV:** The DNS SRV records are a way for the phone to track multiple registration addresses without CIC needing to do it also. It's common for customers to use this registration type in Switchover settings. For more information, see Switchover (failover).

Switchover (failover)

Switchover operates according to the failover paradigm: the primary registration is actually a pair of CIC servers (a Switchover pair) in which both servers have the same capabilities. When an agent places a call, the phone attempts to contact the last known functional server. If the active server fails, the other server in the pair comes up. When an agent places another call, the phone attempts to contact the "down" server first, and then contacts the "up" server when the previous attempt fails. For each subsequent call, the phone remembers which server was functional last and routes its call attempts accordingly.

For Polycom phones and for SIP Soft Phones, you configure this option through a DNS SRV registration in the registration group. Since the DNS SRV record can point to more than one registration address, the phones can track both members of the Switchover pair without CIC needing to notify them of a Switchover event.

Remote survivability (fallback)

Remote survivability is the ability of a phone to maintain a SIP registration when a connection to the main CIC server severs. Remote survivability operates according to the fallback paradigm: there is a single main registration and one or more limited-functionality registrations. The system treats any other registration entries in a registration group as such and assumes them to have less than the full CIC functionality.

When connection to the main CIC server severs, the phones try each of the fallback registrations successively whenever an agent places an outbound call.

Following is a common scenario for remote survivability:

Your implementation has a distributed architecture where remote offices only have phones; proxy servers; and SIP gateways, and connect to the CIC servers at a central location through MPLS/WAN connections. If the WAN fails, the phone attempts to contact the CIC server first. If that fails, the phone calls through the Proxy, which routes the call out the local SIP gateway.

External registrations (proxy settings)

For Polycom phones only, external registrations allow a specific line/station appearance on a phone to register differently than all the other lines on that phone. For example, external registrations are useful in a large company where CIC deploys for a certain sector like customer support, and some other SIP call system deploys elsewhere. If an employee needs to call into the other SIP call system directly, you can add an external registration to that user's phone.

There are two parts to setting up external registrations:
- Set up the external registration group
- Add the external registration to the phone

Set up the external registration group

To designate a registration group as external, select a registration group type of **External** rather than **Regular** when creating the registration group. The registration group itself is then much like a regular registration group, with the exception that the only registration types allowed are **Manual** and **DNS SRV**. As before, there is no limit on the number of entries the group, but more than three is impractical.

Once created, you can only use the external registration group for external registrations. It is not available for selection as the main Registration Group of a managed IP phone.

Add the external registration to the phone

After creating an external registration group, you can add external registrations to Polycom managed IP phones in a similar manner to adding a shared station appearance. Configure the station with whatever Identification Address and credentials that the SIP server in the corresponding external registration group expects.

Change Log

The following table list the changes to the *CIC Managed IP Phones Administrator's Guide* since its initial publication.

Date	Changes
01-June-2012	- In Chapter 3: "Create individual managed IP phones for test purposes": - Updated "Registration group configuration" for default protocol changed from TCP to UDP in IC 4.0 SU1. - Updated "Polycom and SIP Soft Phone registration group configuration" to add screenshot to "Switchover settings". - Updated Chapter 6: "Additional Configuration (Polycom)" to add "SpectraLink Wi-Fi phone considerations" subsection to "Polycom firmware and model limitations". - Updated Chapter 7: "Troubleshooting (Polycom)" under "Polycom phone cannot locate boot server", added note about SpectraLink phones.
11-January-2013	In Chapter 14: "Setup and Configuration (SIP Soft Phone)," revised "Audio device requirement" statement.
24-January-2013	- In Chapter 1: "Introduction to IC Managed IP Phones," updated "About this guide" to refer to the IC Migration Guide for the procedures for migrating IC 3.0 managed IP phones to IC 4.0 managed IP phones. - In Chapter 3: "Create individual managed IP phones for test purposes", updated the screenshots in "Create individual managed IP phones for test purposes" procedure to include the selectable firmware version option on the **Managed IP Phone Configuration – General** tab, added in IC 4.0 SU 3. - In Chapter 6: "Additional Configuration (Polycom)", renamed the subsection on Polycom firmware to "Polycom firmware and phones (IC 4.0 SU 3 and later)" and updated it for the selectable firmware version feature and other firmware changes in IC 4.0 SU 3.
25-July-2013	In Chapter 3: "Create individual managed IP phones for test purposes", added note in the "Default registration group" subsection that a registration group cannot have more than one SIP line or DNS SRV registration.
30-October-2013	- In Chapter 6: "Additional Configuration (Polycom)", updated the "Polycom firmware for supported and EOL phones" section for firmware supported in IC 4.0 SU 3 and later. - In Chapter 2: "Configure the network for managed IP phones", in "Required DHCP option provisioning records", updated Option 132 for Interaction SIP Station. - Updated the following sections for Interaction SIP Station support for DNS SRV in IC 4.0 SU 4 and later: - Chapter 2: "Configure the network for managed IP phones", sections "IP phone network configuration task list" and "Create DNS service location (SRV) records for Switchover". - Chapter 3: "Create individual managed IP phones for test purposes", section "Interaction SIP Station registration group configuration". - Appendix B: How registrations work (proxy settings).
13-January-2014	In Chapter 6: "Additional Configuration (Polycom)", added entry Polycom firmware for VVX600 phones in "Polycom firmware and phones (IC 4.0 SU 3 and later)", to be available in IC 4. SU 5 and pre-release support available in IC 4.0 SU 3 and IC 4.0 SU 4.
27-January-2014	- In Chapter 6: "Additional Configuration (Polycom)": - Modified table in "Polycom firmware and phones (IC 4.0 SU 3 and later)" to show official support in IC 4.0 SU 5 for VVX300, 310, 400, 410, and 600 phones. - Added new section "Interaction Firmware" for IC 4.0 SU 5 and later. - In Chapter 11: "Additional Configuration (Interaction SIP Station)", added new section "Interaction Firmware" for IC 4.0 SU 5 and later. - Updated guide for support for AudioCodes 420HD IP phone in IC 4.0 SU 5 and later. Created new "AudioCodes Administration" section.
16-May-2014	In Chapter 6: "Additional Configuration (Polycom)", modified table in "Polycom firmware and phones (IC 4.0 SU 6 and later)": - To show official support in IC 4.0 SU 6 for Polycom firmware 4.1.6h - Add "IC Release" column showing what SUs and ESs a firmware version is supported in.

15-September-2014	Updated documentation to reflect changes required in the transition from version 4.0 SU# to CIC 2015 R1, such as updates to product version numbers, system requirements, installation procedures, and copyright and trademark information.

05-December-2014	• Made updates throughout the entire document for Interaction SIP Station II and associated firmware, available in CIC 2015 R2 and later.
	• In Chapter 2: "Configure the network for managed IP phones," made the following updates:
	  ○ Added new section "Configure DHCP records for multiple phone models."
	  ○ In "Configure the TFTP server," removed section on manually copying the firmware file to the TFTP server for a Switchover pair, as this has been fixed for CIC 2015 R2 and later.

17-June-2015	In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware and phones (CIC 4.0 SU 6, CIC 2015 R4 and later)" for new Polycom SoundPoint IP and VVX firmware in CIC 2015 R4.
01-July-2015	In Chapter 14: "Setup and configuration (SIP Soft Phone)," updated "SIP Soft Phone requirements" to include support for 32-bit as well as 64-bit for Windows 8 and Windows 8.1.
15-September-2015	• Updated document to reflect transition to Interaction Desktop in CIC 2015 R3 and removal of Interaction Client .NET Edition in CIC 2016 R1. • Updated document for rebranding in CIC 2016 R1.
15-October-2015	In Chapter 2: "Configure the network for managed IP phones," in the "Required DHCP option provisioning records" table, updated Option 067 to provide more information and refer to the "Configure DHCP records for multiple phone recovery" section.
15-February-2016	• In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" to remove older, obsolete firmware versions. • Updated the copyright page.

10-April-2016	• In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" for support for VVX 100 and 200 phones in CIC 2016 R4 and later.
	• Updated the following chapters for TLS/SRTP support for Interaction SIP Station and AudioCodes in CIC 2016 R4 and later:
	  ○ Chapter 1: "Introduction to IC Managed IP Phones", updated "Managed IP phones and SIP security."
	  ○ Chapter 3: "Create individual managed IP phones for test purposes", updated the following sections:
	    ■ "Create individual managed IP phones for test purposes" – Audio Protocol field
	    ■ "Registration group configuration"
	• In Chapter 14: "Setup and configuration (SIP Soft Phone)," updated "SIP Soft Phone requirements" to include support for Windows 10 in CIC 2016 R2 and later.

21-June-2016	In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" to update firmware versions.
02-August-2016	In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" to include new VVX models 301, 311, 401, 411, 501, 601.
26-September-2016	In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" to update firmware versions.
26-October-2016	In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" to update firmware versions to remove 5.4.0a and replace 5.4.3.1014 with 5.4.3.2036 and replace 4.0.8.1972.H with 4.0.8.2058.I.
07-March-2017	• Added note to "Polycom boot Sequence" topic to mention advanced options for boot sequence. • Added note to "Limitations" section to indicate that starting with CIC 2016 R1, firmware updates with a version of 2.2.2.77 or higher for Interaction SIP Station I & II remove the constraint of registering the phone with a single server.
28-April-2017	In Chapter 6: "Additional Configuration (Polycom)," updated "Polycom firmware for currently supported phones table" to include firmware 5.4.5G.
19-September-2017	Rebranded to Genesys.
20-March-2018	Updated document format and removed chapter numbers.

31-May-2018	Added support for Polycom phone models Trio8500 and Trio8800 in [Polycom Firmware for Supported and EOL Phones](#).
06-September-2018	Added a link to the "CIC Utilities and Downloads" page for "Phone Simulator" in [Boot and Provision Sequences (Polycom)](#).
21-September-2018	Added support for Polycom phone model VVX D60 and Polycom 5.8. Firmware in [Polycom Firmware for Supported and EOL Phones](#).
08-January-2019	Added support for AudioCodes 405; 405HD; 420HD; 430HD; and 440HD, and Genesys 405, 405HD, and 420HD.
05-March-2019	Added support for the following Polycom phone models in [Polycom Firmware for Supported and EOL Phones](#): • VVX 150 • VVX 250 • VVX 350 • VVX 450
29-March-2019	Added link to the SIP Soft Phone help in [SIP Soft Phone Help](#).
29-April-2019	Reorganized the content only, which included combining some topics and deleting others that just had an introductory sentence such as, "In this section...".
07-May-2019	Removed "New CIC Distribution Model" as the model is no longer new.
11-June-2019	Updated Firmware to 2.2.16.142.41 for the following phones in [AudioCodes and Genesys Firmware for Supported Phones](#): • AudioCodes and Genesys 405 • AudioCodes and Genesys 405HD • AudioCodes and Genesys 420HD • AudioCodes 430HD • AudioCodes 440HD
25-September-2019	TLS/STRP support for Interaction SIP Station I and II phones is not available, as indicated previously. For more information, see [Managed IP Phones and SIP Security](#).
27-September-2019	Updated Polycom Firmware to address a vulnerability in the web-based management interface of VVX, Trio, SoundStructure, SoundPoint, and SoundStation phones running Polycom UC Software. If exploited, the vulnerability could allow an authenticated, remote attacker with administrator privileges to cause a denial of service (DoS) condition or execute arbitrary code. Following are the Firmware updates: • 5.9.3.2857 Rev G • 5.9.0 Rev AD • 4.0.14.1580 For more information, see [Polycom Firmware for Supported and EOL Phones](#).
14-January-2020	Updated links to KB articles in the following: • [Implement QoS in Your Environment](#) • [Polycom Firmware Update Required](#)
16-March-2020	Added support for Firmware 3.3.2 and added a server parameter to fix a firmware downgrade issue. For more information, see [Polycom Firmware for Supported and EOL Phones](#).
01-April-2020	Changed path to *Security Features Technical Reference* in [Managed IP Phones and SIP Security](#).
29-April-2020	• Update or remove links to "my.inin.com" as appropriate. • Added links to AudioCodes documentation that were on the "my.inin.com" site. For more information, see [AudioCodes Documentation](#).

03-June-2020	Removed "CIC 2.4/3.0 to 2015 R1 or Later Migrations." Most, if not all, customers have already migrated from CIC 2.4/3.0 to a later release.
08-June-2020	Added support for Firmware 5.9.6.2327 and 4.0.15.1009. For more information, see Polycom Firmware for Supported and EOL Phones.
13-July-2020	Changed the link in Managed IP Phone (SIP Soft Phone) Configuration Options to point to the actual configuration page in the *Interaction Administrator* help.
10-September-2020	Added support for Firmware 2.2.16.408. For more information, see AudioCodes and Genesys Firmware for Supported Phones.
21-July-2023	Added support for Firmware 2.2.16.643. For more information, see AudioCodes and Genesys Firmware for Supported Phones.