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# Interaction SIP Bridge

## Installation and Configuration Guide

### Abstract

This document provides installation and configuration information for Interaction SIP Bridge. Interaction SIP Bridge is a client software product that enables at-home workers to use a physical SIP-enabled telephone to make and receive calls from a Customer Interaction Center environment through a remote connection, such as through a virtual private network. For the latest version of this document, see the PureConnect Documentation Library.

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

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# Introduction of Interaction SIP Bridge

## Purpose of Interaction SIP Bridge

Interaction SIP Bridge enables agents to use a hardware-based IP telephone at a remote location for sending and receiving calls through TCP/IP network communications to Customer Interaction Center (CIC). Interaction SIP Bridge resides on the computer of the remote agent and relays SIP, RTP, and HTTP messages between the CIC network and the IP telephone. To communicate with the remote CIC telephony network, Interaction SIP Bridge uses an existing software-based virtual private network (VPN) connection on the computer of the remote agent. After using the VPN to connect to the CIC network, the remote agent can use the IP telephone as if it is locally connected to the CIC network.

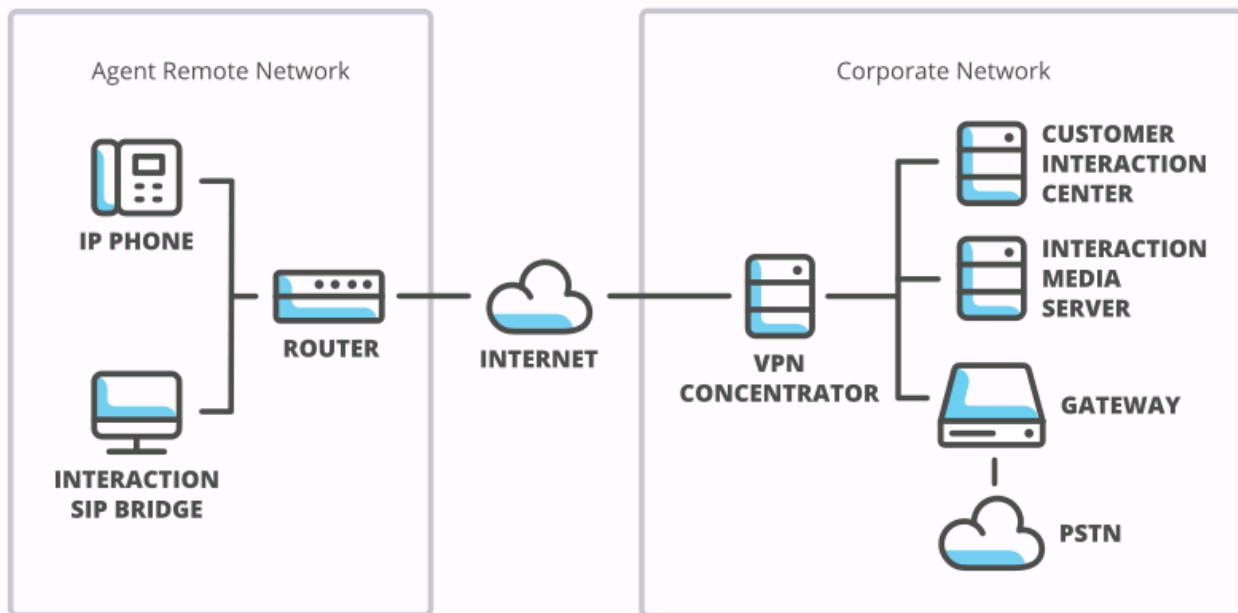
## Network overview of Interaction SIP Bridge

Interaction SIP Bridge requires no additional network hardware than that necessary to connect a remote computer over the Internet to a standard VPN concentrator. Interaction SIP Bridge provides the following network features to the IP telephone:

- DHCP Option functionality, which enables out-of-the-box managed IP telephones to discover an Interaction SIP Bridge workstation without manual intervention
- Relay of RTP audio and SIP call control messaging
- Relay of HTTP and HTTPS provisioning messages to and from the Customer Interaction Center Provisioning server, which enables remote firmware updates and configuration enforcement

## Interaction SIP Bridge with Customer Interaction Center

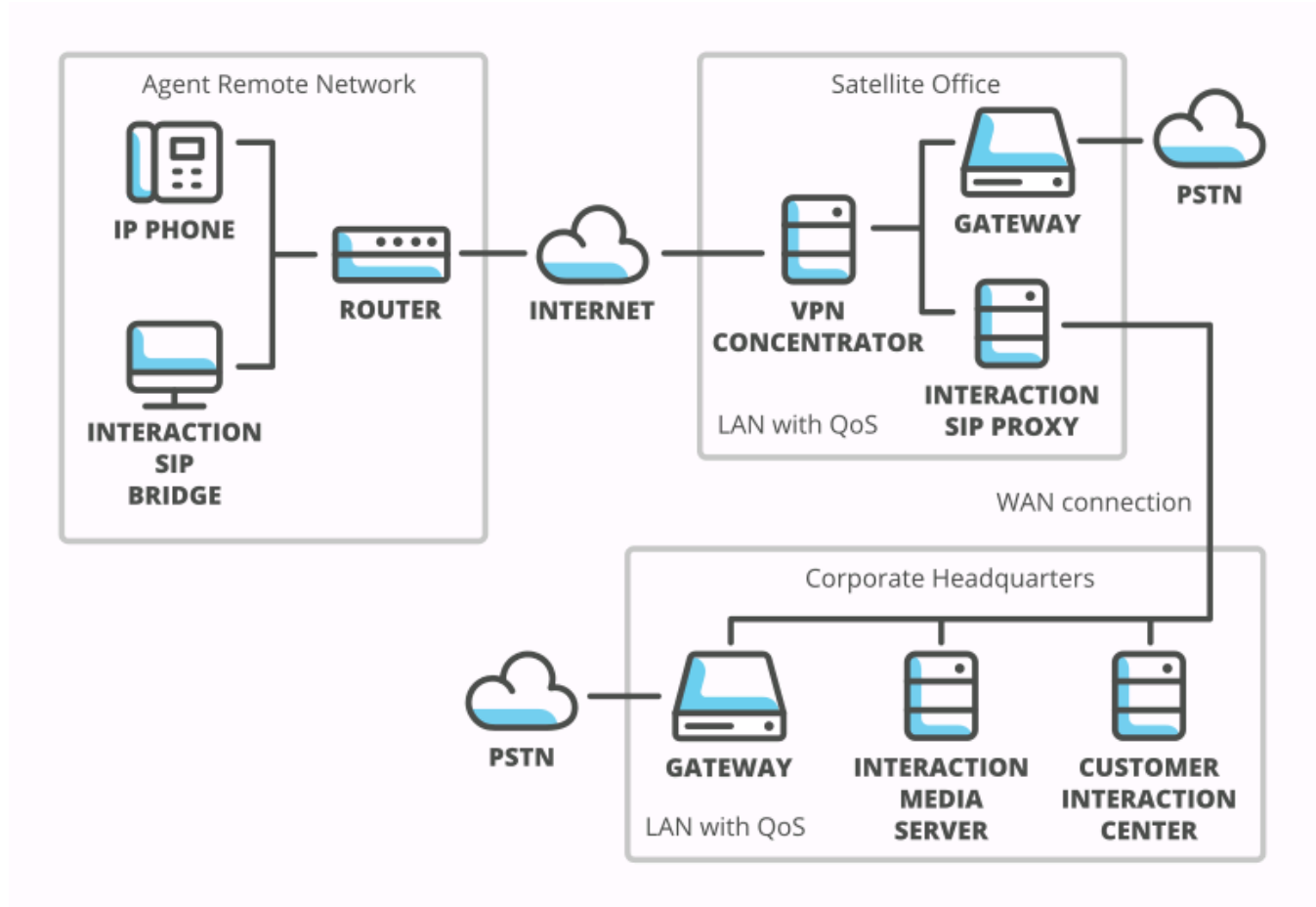
Interaction SIP Bridge connects to the Customer Interaction Center server through a software-based VPN connection on the computer in the network of the remote agent. Interaction SIP Bridge, through the local area network (LAN) of the remote agent, serves as a relay for all RTP, SIP, and HTTP communications to and from the IP telephone.



Interaction SIP Bridge network architecture

## Interaction SIP Bridge with Interaction SIP Proxy

Alternatively, you can configure Interaction SIP Bridge to register its associated IP telephone with an Interaction SIP Proxy instead of a Customer Interaction Center server.



### Interaction SIP Bridge with Interaction SIP Proxy



If you use an Interaction SIP Proxy server as a registration source, Interaction SIP Bridge can register its associated IP telephone through the software-based VPN connection. This configuration has the following limitations:

- Interaction SIP Bridge cannot use Digest Authentication with Interaction SIP Proxy. Interaction SIP Proxy is aware only of the managed IP telephone definitions that you define through Interaction Administrator. For more information about managed IP telephones, see [Managed IP telephones](#) and also read *CIC Managed IP Phones Administrator's Guide* in the PureConnect Documentation Library. Interaction SIP Proxy does not receive Interaction SIP Bridge workstation definitions from Customer Interaction Center.
- Interaction SIP Proxy cannot provision firmware to the IP telephone that is associated with an Interaction SIP Bridge workstation. To have the satellite office provide firmware updates to the IP telephone, you must provide a separate provisioning server.

# Interaction SIP Bridge requirements

The following sections provide the requirements for installing and using Interaction SIP Bridge:

[Computer hardware](#)

[Computer software](#)

[Customer Interaction Center environment](#)

[Remote agent local area network](#)

[IP Telephone](#)

[Restrictions](#)

## Computer hardware

Interaction SIP Bridge uses the same computer hardware requirements as those for all CIC user applications. The hardware requirements for the computer of the remote agent are listed in "Client workstation hardware requirements" in *PureConnect Installation and Configuration Guide*, which is available in the PureConnect Documentation Library in the **Installation and Configuration** section.

## Computer software

The software on the computer of the remote agent must meet the following requirements:

- Interaction SIP Bridge uses the same software requirements as those for all CIC user applications. For information about software requirements, see "Client workstation software requirements" in *PureConnect Installation and Configuration Guide*. This document is available in the PureConnect Documentation Library in the **Installation and Configuration** section.
- The computer on the remote network of the agent must have a VPN client software application that enables connection to the corporate network.
- The VPN client software application must define a virtual network adapter in Windows. Use the **Change adapter settings** hyperlink in the **Network and Sharing Center** window to view all defined network adapters of the Windows operating system.
- Interaction SIP Bridge supports only the PureConnect Quality of Service (QoS) driver, which the installation process provides and configures for existing network connection definitions. Interaction SIP Bridge does not support other QoS solutions, such as Microsoft QoS Packet Scheduler. However, network connections can support both the PureConnect QoS driver and Microsoft Packet Scheduler without conflicts.

### Important!

Interaction SIP Bridge receives and responds to broadcast messages from the IP telephone on a local area network. Because IP telephone broadcasts do not extend past the local area network, Interaction SIP Bridge does not function on a terminal services instance, a remote desktop session, or a remote virtual machine.

## Customer Interaction Center environment

The Customer Interaction Center environment must meet the following requirements:

- Customer Interaction Center 2015 R1 or later
- The necessary Interaction Media Servers are installed, registered with the Customer Interaction Center server or servers, and configured for your VoIP communications specifications.

## Remote agent local area network

The local area network of the remote agent must meet the following requirements:

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## Internet connection

The Internet connection must meet the following requirements:

- Speed rating - Both the download and the upload speeds must be sufficient for receipt and transmission of RTP (voice) packets for the assigned codec of the IP telephone (G.711=64 kbit/sec, G.729=8 kbit/sec, G.726=16, 24, 32, or 40 kbit/sec)

### **Important!**

The Internet connection of the remote agent network must also support the additional bandwidth requirements of other necessary activities. Such activities can include file transfers, e-mail synchronization (including attachments), client software communications with corporate servers, web-based applications on the corporate network, and so on.

- Latency - Audio quality of VoIP communications degrades with latency values greater than 150 milliseconds
- Consistency - The Internet connection must not experience consistently recurring service outages.

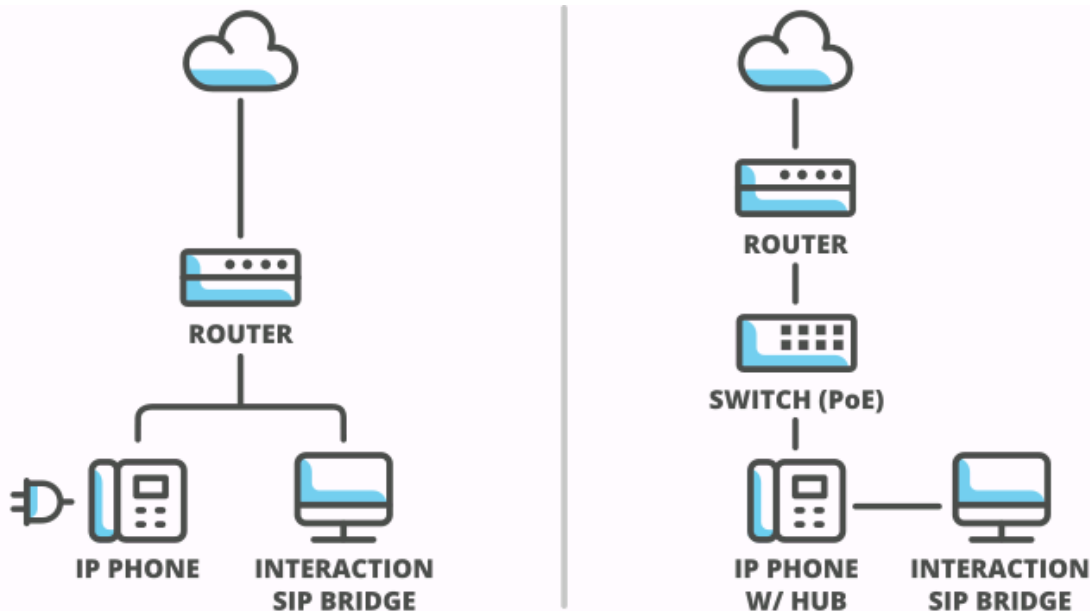
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## Port configuration

Ports 343, 8088, and 8089 must be open throughout the network connection, which ranges from the remote computer of the agent to the corporate network. If you modify the ports that Interaction SIP Bridge uses for communications, you must also ensure that these ports are available throughout the network connection. Include the remote computer, firewalls, and other network devices, such as session border controllers, in ensuring port availability.

## Network devices

The following diagram displays common agent local area network environments that are commonly used with Interaction SIP Bridge:



Interaction SIP Bridge home network examples



The agent remote network must include one of the following devices:

- Internet router with Dynamic Host Control Protocol (DHCP) capability  
The router must be able to support dynamic IP addressing for the IP telephone and the computer hosting Interaction SIP Bridge, if you do not configure static IP addresses for the computer hosting Interaction SIP Bridge and the IP telephone.
- Ethernet switch (optional)  
If you require Power over Ethernet (PoE) to run the IP telephone, you must connect the IP telephone to an Ethernet switch that supports PoE. In this configuration, Genesys recommends that you connect the computer hosting Interaction SIP Bridge to the Ethernet hub on the IP telephone with an Ethernet cable instead of the Ethernet switch or Internet router.

### Important!

Both the computer hosting Interaction SIP Bridge and the IP telephone must reside within the same Ethernet subnet on the agent remote network. Communication between the IP telephone and Interaction SIP Bridge can be problematic if you connect the computer to the Internet router and the IP telephone to an Ethernet switch. You must also ensure that the IP telephone and the computer hosting Interaction SIP Bridge do not have overlapping subnets.

## Cables

Category 5 or better Ethernet networking cables to connect the devices, including the IP telephone, to the remote local area network



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## Supported networking features

- Internet Protocol version 4 (IPv4)
- TCP, UDP, and TLS transport protocols
- RTP and SRTP control protocols
- SIP signaling protocol
- NTP clock synchronization protocol
- Software-based Virtual Private Network (VPN) connections
- HTTP Digest Authentication
- RFC 2833 for Dual-Tone Multifrequency (DTMF) transmission
- All codecs supported by Customer Interaction Center

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## Unsupported networking features

- Virtual Local Area Networks (VLANs)
- Internet Protocol version 6 (IPv6)
- H.323 specification
- In-band DTMF transmission

## IP Telephone

**Note:**

For use with Interaction SIP Bridge, your IP telephone must support HTTP Digest Authentication.

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## Managed IP telephones

*Managed IP phones* are IP telephones that can receive configurations from Customer Interaction Center. You configure all necessary options through Interaction Administrator.

Interaction SIP Bridge requires a Polycom IP telephone that Customer Interaction Center supports. The IP telephones that Interaction SIP Bridge supports as managed IP telephones in Customer Interaction Center are in the **Recommended** section at <http://testlab.genesys.com>.

For more information about managed IP phones, see *CIC Managed IP Phones Administrator's Guide* in the PureConnect Documentation Library.

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## Unmanaged IP telephones

At this time, Interaction SIP Bridge does not officially support unmanaged IP telephones, also known as *stations* in Interaction Administrator. PureConnect Customer Care representatives cannot provide assistance for this feature.

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## Unsupported IP telephones

Interaction SIP Bridge does not support the following voice over IP (VoIP) solutions:

- Interaction SIP Station
- Interaction SIP Softphone or any software-based VoIP solution, also known as *softphones*
- Polycom IP300 and IP500 model series of IP telephones

## Restrictions

Interaction SIP Bridge has the following restrictions regarding installation and operation:

- A personal computer can host only one instance of Interaction SIP Bridge. You cannot run multiple instances of Interaction SIP Bridge on the same computer.
- An instance of Interaction SIP Bridge supports only one IP telephone. You cannot use multiple IP telephones through an Interaction SIP Bridge instance on a computer.
- You cannot install Interaction SIP Bridge and Interaction SIP Proxy on the same computer.

# Interaction SIP Bridge installation

This section contains the procedures for installing Interaction SIP Bridge on a computer in a remote local area network and contains the following topics:

- [Pre-installation tasks for Interaction SIP Bridge](#)
- [Install Interaction SIP Bridge through IC User Applications](#)
- [Deploy Interaction SIP Bridge](#)
- [Uninstall Interaction SIP Bridge](#)

## Pre-installation tasks for Interaction SIP Bridge

1. Log on to the computer in the remote network of the agent with an account that has administrative privileges on the local computer.  
For more information about required privileges for installing CIC client applications, see "Administrative privileges" in "Chapter 5: Client Workstation" of *PureConnect Installation and Configuration Guide*.
2. Ensure that the Internet connection is operational.
3. Start the virtual private network (VPN) client software program.
4. If necessary, provide the required information, such as authentication credentials, to the VPN software to connect to the corporate network.

## Install Interaction SIP Bridge through IC User Applications

1. Ensure that a Customer Interaction Center server has been installed and is accessible from the workstation on which you will install Interaction SIP Bridge.

**Tip:**

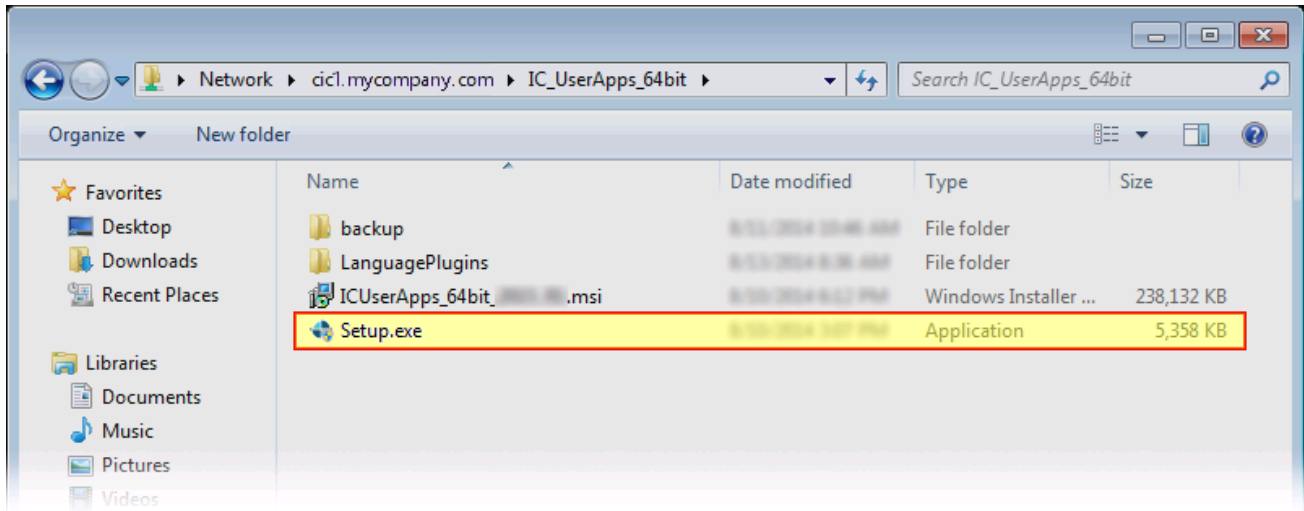
The remote workstation will likely need to establish a VPN connection to the network on which the Customer Interaction Center server resides. Other areas to consider for the connection to the Customer Interaction Center server are IPv4/IPv6 addressing, Network Address Translation (NAT), and firewall settings.

2. Ensure that the Windows user account through which you will install Interaction SIP Bridge can access the following shared resources on the Customer Interaction Center server:
  - `\\<CIC_server>\IC_WorkstationPreReqs`
  - `\\<CIC_server>\IC_UserApps_64bit`
  - `\\<CIC_server>\IC_UserApps_32bit`
3. On the remote workstation, use Windows Explorer to navigate to the appropriate installation resource:
  - `\\<CIC_server>\IC_UserApps_64bit`
  - `\\<CIC_server>\IC_UserApps_32bit`

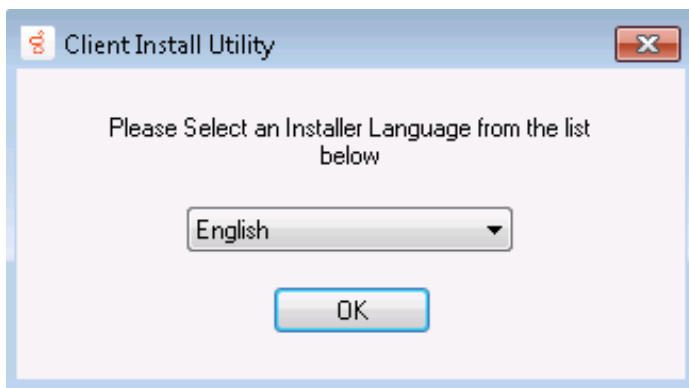
**Note:**

You can install the 64-bit version of the IC User Applications only on a 64-bit Windows operating system. You can install the 32-bit version of the IC User Applications on either a 32-bit or 64-bit Windows operating system.

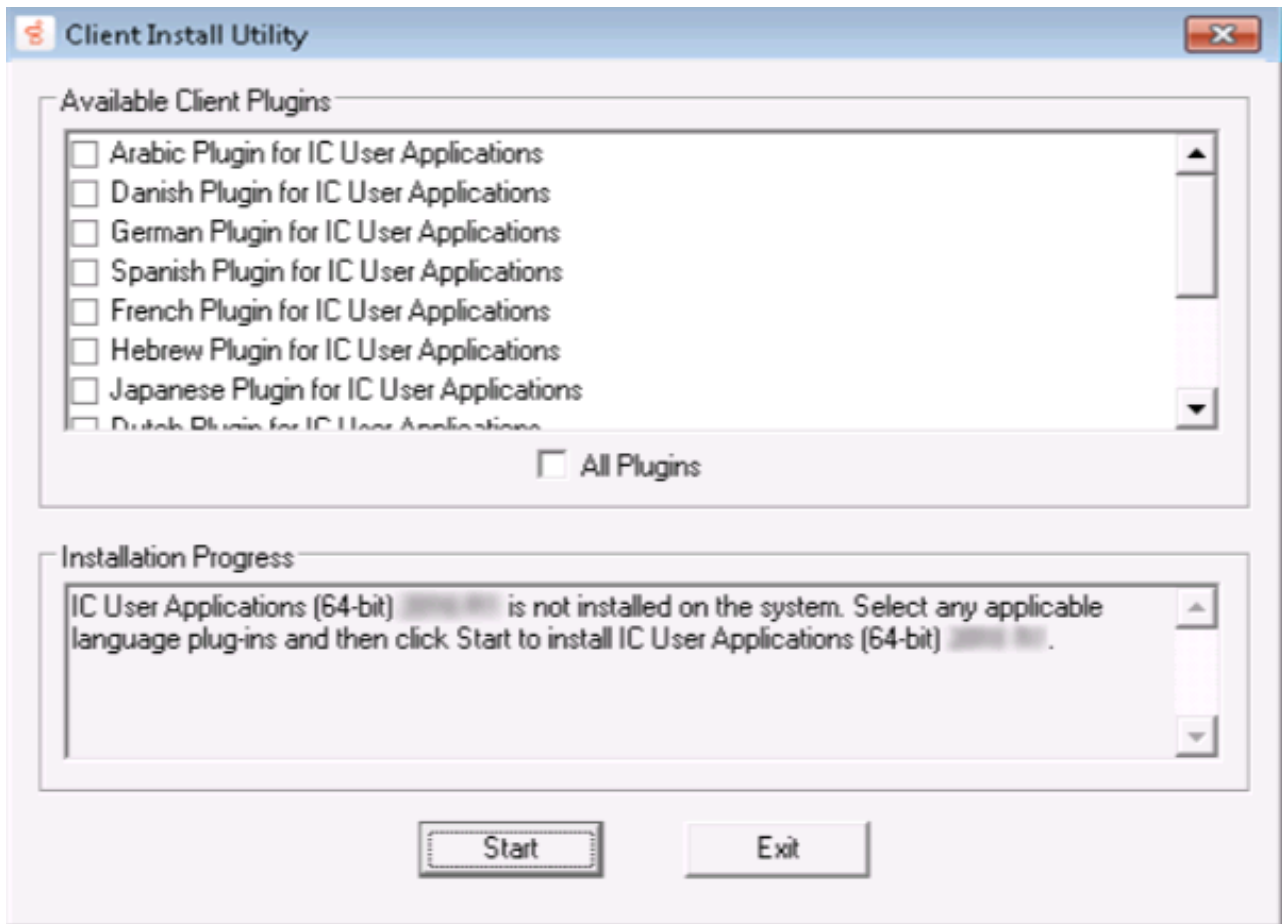
4. Run `setup.exe`.



The IC User Applications installation process starts and displays the first **Client Install Utility** dialog box.



5. In the first **Client Install Utility** dialog box, select the language that you want the installation program to use and select **OK**. The installation program displays the second **Client Install Utility** dialog box.



**Note:**

You do not need to select any plugins for installation in the second **Client Install Utility** dialog box.

6. Select the **Start** button.

The installation program checks the workstation for the necessary requirements.

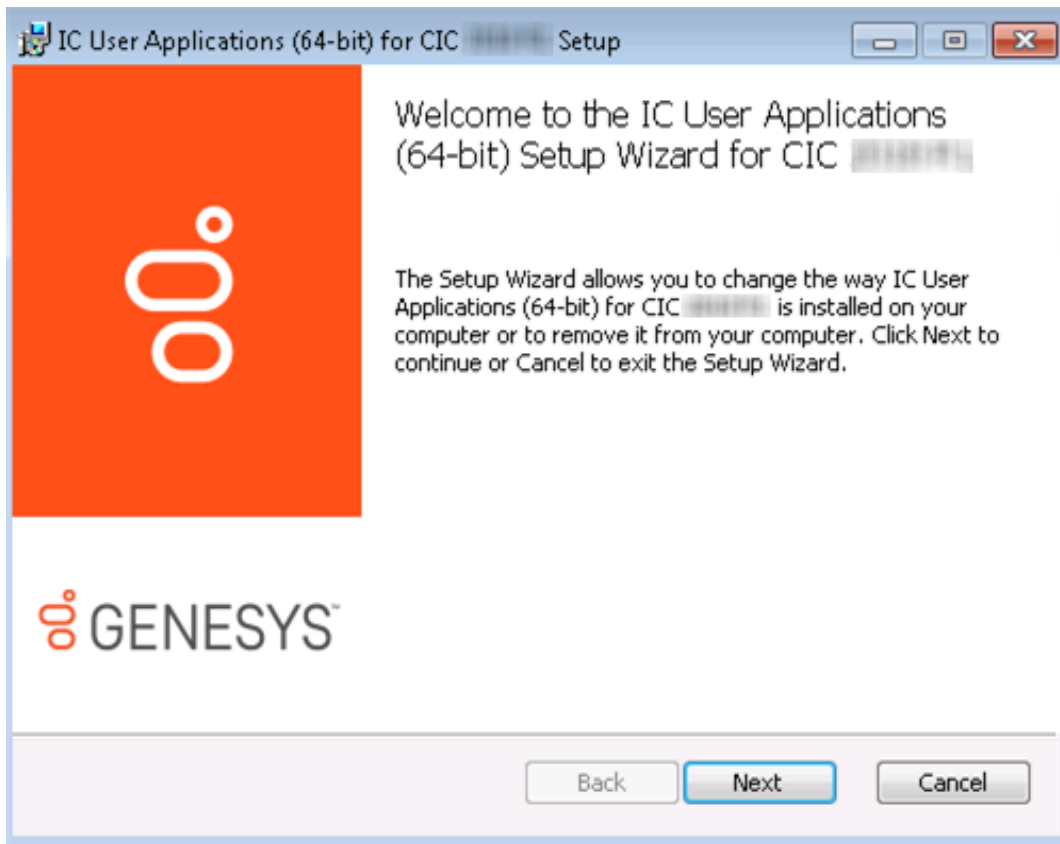
**Note:**

If the minimum requirements are not met, the installation program displays a dialog box that informs you of the items that you must install before you can install Interaction SIP Bridge. All of the required workstation software for IC User Applications is available in the following shared resource of the Customer Interaction Center server:

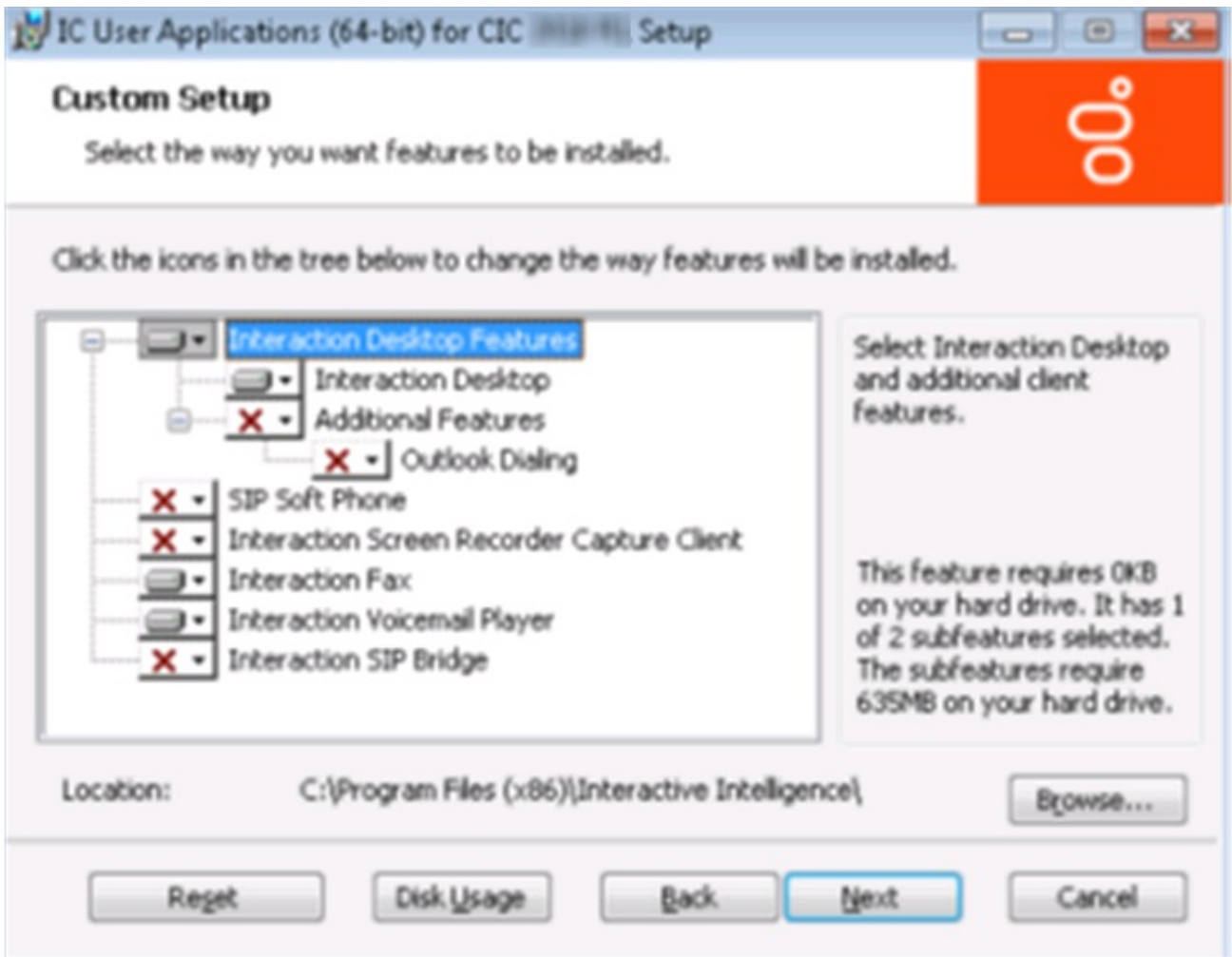
\\<CIC\_server>\IC\_WorkstationPreReqs

Stop the IC User Applications installation, install the required software on the workstation, and then restart the IC User Applications installation.

If the workstation meets the minimum requirements, the following dialog box appears:

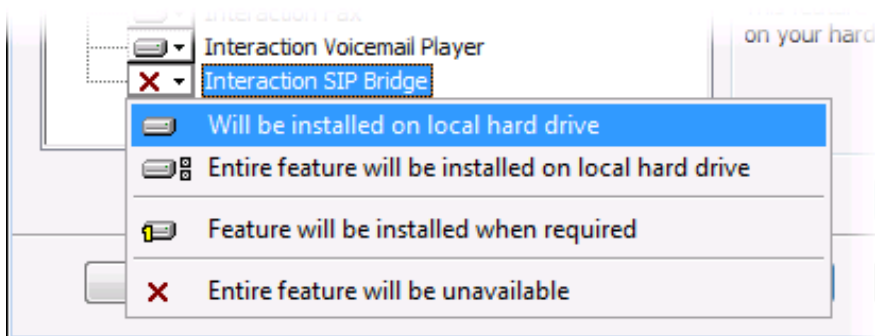


7. Select the **Next** button.
8. In the **Custom Setup** dialog box, configure which IC User Applications you want to install on the workstation.

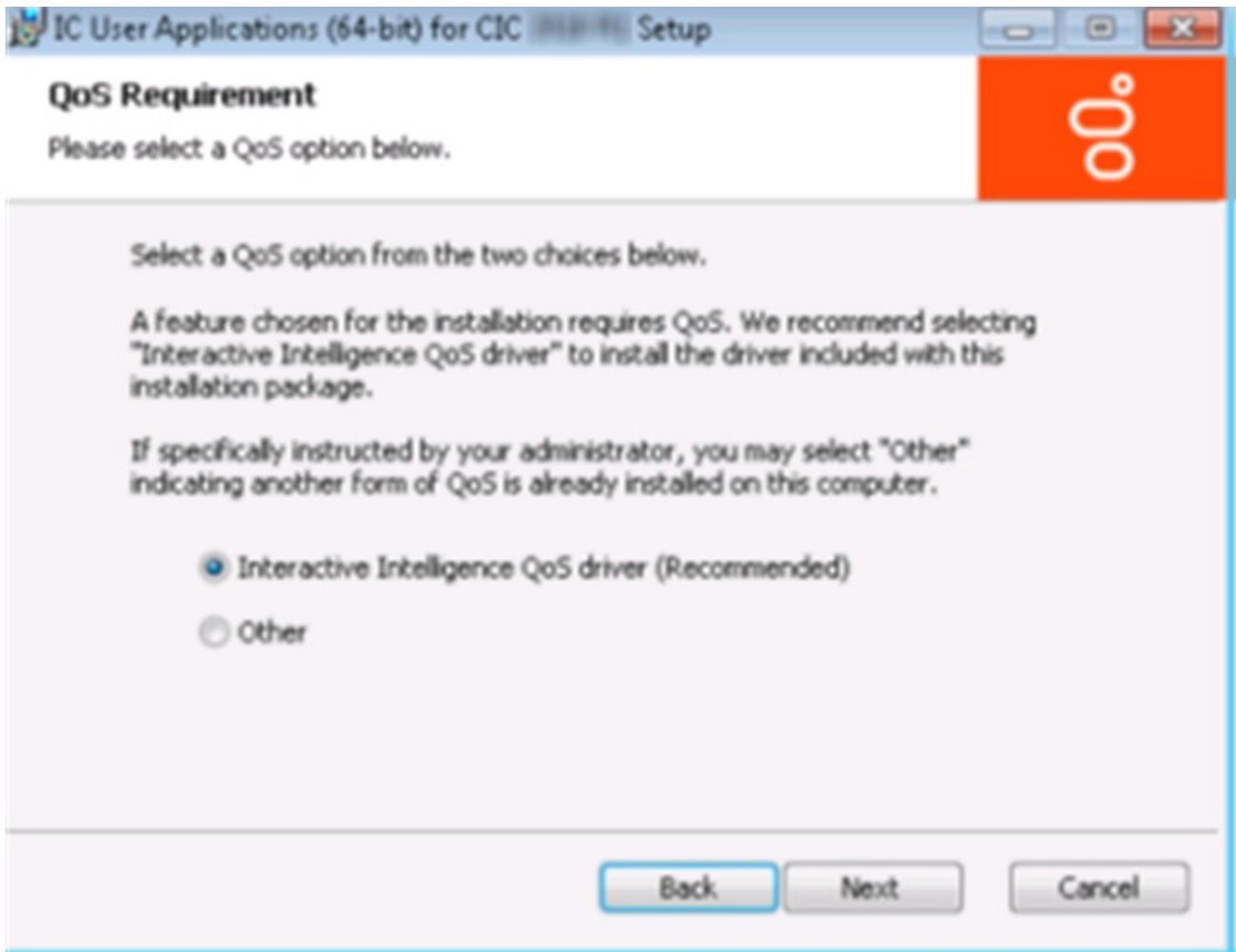


**Important!**

By default, Interaction SIP Bridge is not configured for installation. You must click the Interaction SIP Bridge item and select **Will be installed on local hard drive** from the resulting menu.



9. After you have configured the program to install Interaction SIP Bridge, select the **Next** button.
10. In the **QoS Replacement** dialog box, select whether you want Interaction SIP Bridge to use the PureConnect Quality of Service (QoS) driver and select the **Next** button.



Do not cancel the installation process if the status bar later appears to stall. This stall occurs during the installation of the PureConnect Quality of Service (QoS) driver and is normal.

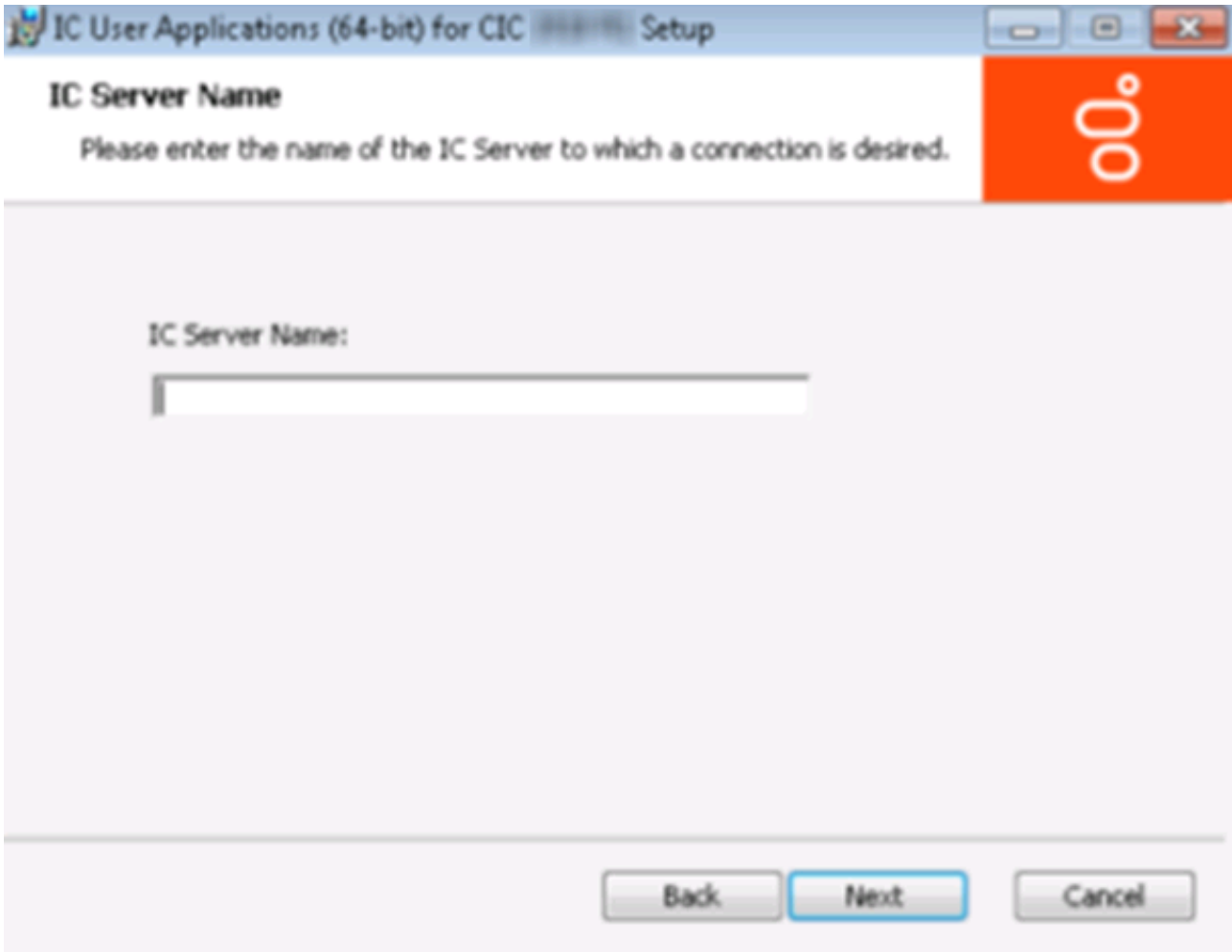
The installation program modifies existing network connections to include the PureConnect Quality of Service (QoS) driver, which Interaction SIP Bridge requires.

**Important!**

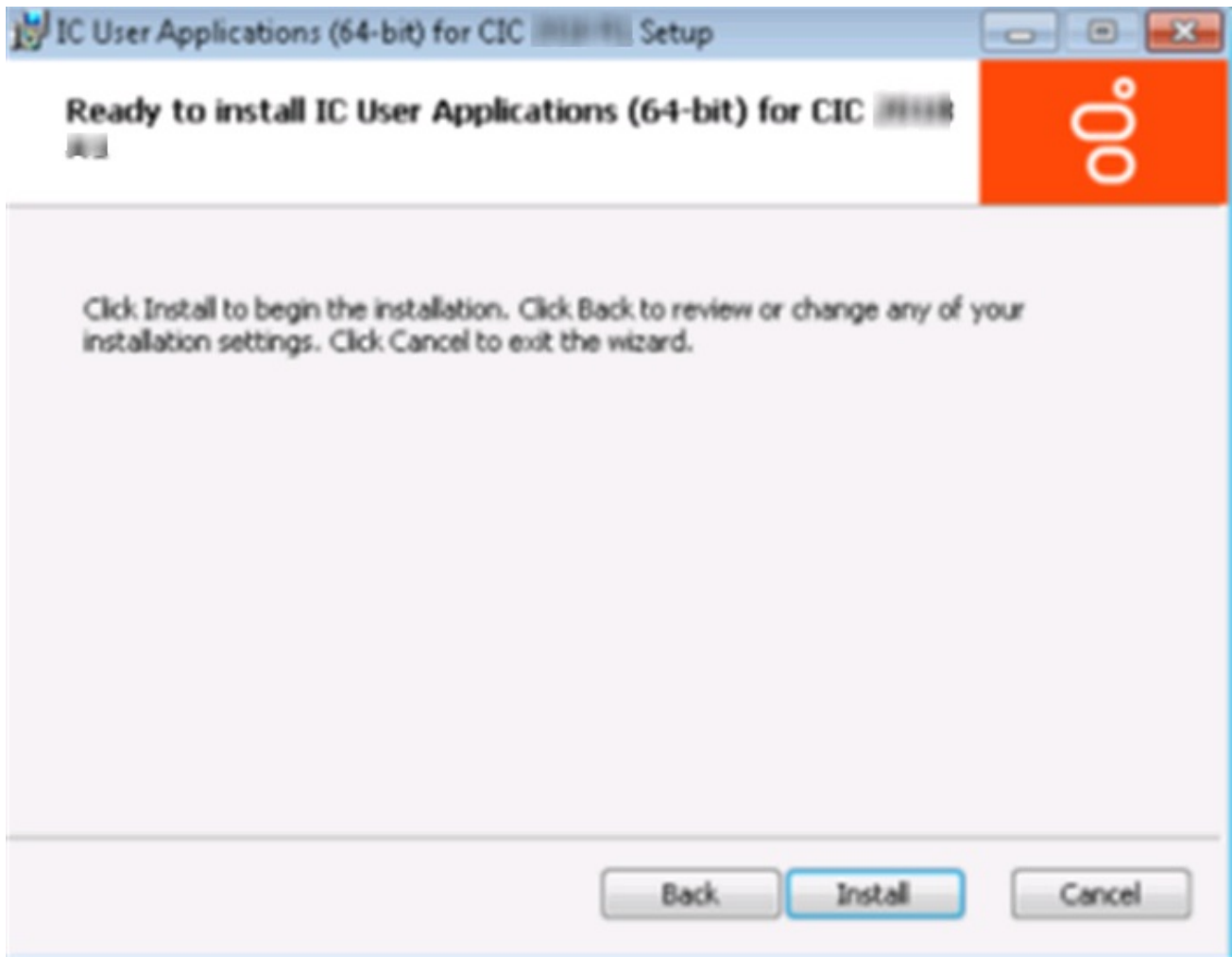
If your current VPN connection does not recover from the reset from the installation of the QoS driver, do the following steps:

1. Close the VPN client software program.
  2. Open a **Command Prompt** window and issue the following commands:
    - `ipconfig /release`
    - `ipconfig /renew`
  3. Close the **Command Prompt** window.
  4. Start your VPN client software.
  5. Log on to the VPN server with your credentials.
11. In the **IC Server Name** dialog box, enter the fully-qualified domain name (FQDN) or IP address of the Customer Interaction Center server to which Interaction SIP Bridge will connect and select the **Next** button.





12. In the **Ready to install IC User Applications for CIC** dialog box, select the **Install** button.



The installation program installs the selected applications. When the process completes, the following dialog box appears:



13. Select the **Finish** button.
14. In the **Client Install Utility** dialog box, select the **Exit** button.

**Note:**

If Windows displays the **Set Network Location** dialog box, select the type of network that is appropriate for the environment of the remote computer.

The installation program created an Interaction SIP Bridge icon on the desktop. You can use this icon to start Interaction SIP Bridge after you establish a virtual private network (VPN) connection to the network through which you can access the Customer Interaction Center server.

## Deploy Interaction SIP Bridge

If you want to deploy Interaction SIP Bridge, Genesys recommends that you use the Group Policy feature in recent versions of Microsoft Windows Server.

For more information about deploying Customer Interaction Center applications through the Group Policy feature, see *Group Policy Deployment for CIC Applications Technical Reference* in the PureConnect Documentation Library.

## Uninstall Interaction SIP Bridge

1. Open Windows Control Panel.
2. Select **Programs and Features**.  
Windows displays the **Programs and Features** window.
3. In the list on the right side of the window, select **IC User Applications**.
4. In the ribbon above the list, select **Uninstall**.  
Windows prompts you to confirm the uninstall action.
5. In the confirmation dialog box, select the **Yes** button.  
The uninstall process begins.
6. If you are prompted by a **User Account Control** dialog box, select the **Yes** button.  
All IC User Applications are removed from the workstation.

# Define Interaction SIP Bridge workstations and IP telephones in Interaction Administrator

This subject includes:

- [Interaction SIP Bridge configuration prerequisites](#)
- [Registration group considerations for Interaction SIP Bridge](#)
- [Define an Interaction SIP Bridge workstation](#)
- [Define a managed IP telephone for an Interaction SIP Bridge workstation](#)
- [Defining multiple managed IP telephones for use with Interaction SIP Bridge](#)

## Interaction SIP Bridge configuration prerequisites

Before you define Interaction SIP Bridge workstation entries and the associated IP telephones, do the following tasks:

- Determine a naming convention that results in a unique name for each Interaction SIP Bridge workstation.
- Acquire the MAC address of each IP telephone that you will associate with an Interaction SIP Bridge workstation.
- To ensure that the correct IP telephone is sent to the appropriate remote locations, determine a method to physically identify each IP telephone as being associated with a specific Interaction SIP Bridge workstation.
- Acquire and apply Interaction SIP Bridge licenses

Interaction SIP Bridge requires that you purchase Interaction SIP Bridge licenses. The Interaction SIP Bridge licenses are included in your Customer Interaction Center license, which you can view in Interaction Administrator by selecting **File > License Management**. The Interaction SIP Bridge license is **I3\_LICENSE\_SIP\_BRIDGE**. Ensure that you update your Customer Interaction Center license before you attempt to install and use Interaction SIP Bridge on remote computers.

## Registration group considerations for Interaction SIP Bridge

In Customer Interaction Center, registration groups define Customer Interaction Center network entities to which managed IP telephones can register for sending and receiving SIP messages. When you define an Interaction SIP Bridge workstation using Interaction Administrator, you select a registration to associate with it. The following list provides the Customer Interaction Center network entities that you can assign to a registration group:

- SIP lines
- A SIP registration server
- Interaction SIP Proxy servers

You can add multiple entities to the registration group for redundancy considerations. For example, if you define a registration group with a Customer Interaction Center SIP line as the first entity and an Interaction SIP Proxy server as the second entity, Interaction SIP Bridge will attempt to register to the second entity if the first is unavailable.

For some of these network entities, you can define a port number and a transport layer protocol within the registration group. For other network entities, such as SIP lines, you must configure the port number and transport layer protocol in the SIP line definition. Customer Interaction Center supports the following types of transport layer protocols:

- Telecommunications Protocol (TCP) This transport layer protocol requires the endpoint that receives the IP packets to provide confirmation when the IP packet arrives.
- User Datagram Protocol (UDP) With this transport layer protocol, IP packets are transmitted only once. The receiving endpoint does not send receipt confirmations.
- Transport Layer Security (TLS) TLS is an application layer protocol that encrypts IP packets. It typically uses TCP as the transport layer protocol.

The default Customer Interaction Center registration groups, <Default Registration Group> and <Default Secure Registration Group>, use the <Stations-UDP> and <Stations-TLS> default SIP lines.

For managed IP telephones that send communications within private networks that do not use WAN links or the Internet, UDP is a viable solution. Typically, private networks rarely have problems with dropped packets or other transmission problems. However, a managed IP telephone that you associate with an Interaction SIP Bridge workstation transmits communications through the LAN of the remote agent across a VPN connection over the Internet to the network connected to the Customer Interaction Center server. In this scenario, there is an increased possibility of dropped packets.

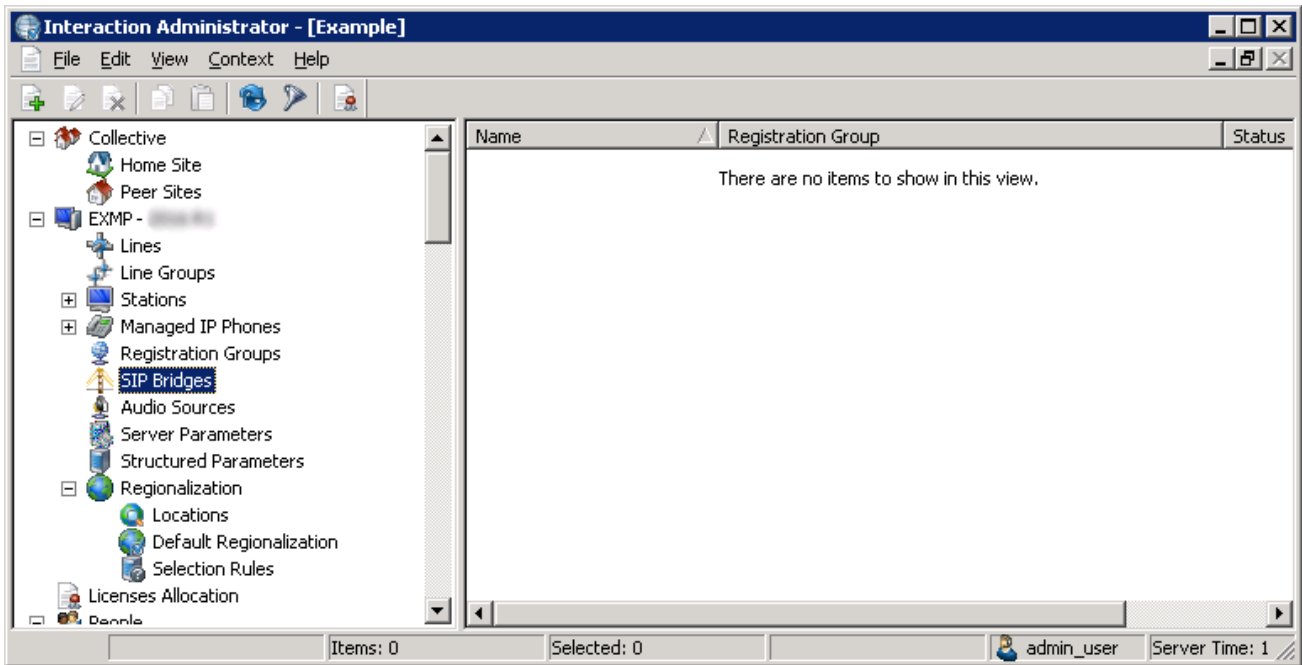
For Interaction SIP Bridge registration groups, consider either creating a new SIP line that uses the TCP transport layer protocol or using the default <Stations-TCP> SIP line. You can then assign that TCP-based SIP line to a registration group created specifically for Interaction SIP Bridge workstations.

For more information about registration groups and managed IP telephones, see Interaction Administrator Help.

## Define an Interaction SIP Bridge workstation

The following procedure provides the steps to define an Interaction SIP Bridge workstation on the Customer Interaction Center server through Interaction Administrator.

1. On a computer in the network that contains the Customer Interaction Center server, start Interaction Administrator.  
The **Interaction Administrator** window is displayed.

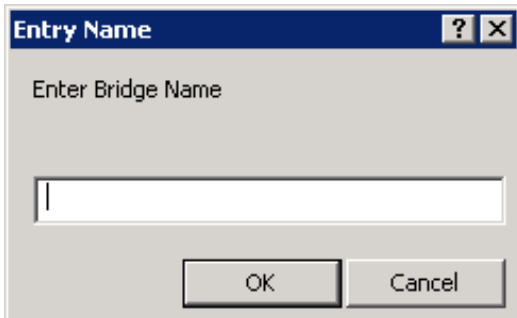


- In the left pane, select the **SIP Bridges** object.

**Note:**

Interaction Administrator displays the SIP Bridges object only if you purchased Interaction SIP Bridge licenses and applied them to your Customer Interaction Center license.

- In the right pane, right click an open area and select **New** from the resulting shortcut menu. The **Entry Name** dialog box is displayed.



- In the **Enter Bridge Name** box, enter a unique name for an Interaction SIP Bridge workstation.

**Tip:**

Create a name that is easily recognizable as belonging to a specific user or managed IP telephone.

- Select the **OK** button. The **SIP Bridges Configuration** dialog box is displayed.





Customer Interaction Center supports only Polycom IP telephones as managed IP telephones. For more information about managed IP telephones, see *CIC Managed IP Phones Administrator's Guide* in the PureConnect Documentation Library.

1. Start Interaction Administrator.
2. In the left pane of the **Interaction Administrator** window, select the **Manage IP Phones** container.
3. In the right pane, right click an open area and select **New** from the resulting shortcut menu.

The **New Managed IP Phone** dialog box is displayed.

4. In the following controls, enter the appropriate information:

Control	Usage
Name	Enter a unique name for this managed IP telephone. This name can be identical to the SIP Bridge entry that you created earlier.
Template	Select None.
Type	Select Workstation.
Manufacturer	Select Polycom.
Model	Select the model of the Polycom IP telephone that is connected to the LAN of the remote agent.

5. Select the **OK** button.

The **Managed IP Phone Configuration** dialog box is displayed.

**Managed IP Phone Configuration - JohnDoe** [?] [X]

General | Options | Information

Name:    Active

MAC Address:

Registration:  Registration Group  SIP Bridge

Location:

Audio Protocol:

Time Zone:

Station Appearances:

Key	Label	Station	Extension
1	JohnDoe	JohnDoe	
2	[Unassigned]		
3	[Unassigned]		

6. In the following controls, enter the appropriate information:

Control	Usage
Name	<p>Enter a unique name for this managed IP telephone.</p> <p><b>Tip:</b> Use a name that is easily identifiable, such as including the name of the user or the extension. You can name this managed IP telephone entry with the same name that you used for the SIP Bridge entry.</p>
MAC Address	<p>Enter the MAC address of the IP telephone.</p> <p>Typically, the IP telephone has a label that displays the MAC address. You can also view the MAC address by connecting the IP telephone to the network and using the menu system of the IP telephone.</p>
Registration	<p>Select the SIP Bridge option.</p> <p>The list box immediately changes to contain the existing SIP Bridge workstation entries and the Configure button is displayed.</p> <p>Use the Configure button to display the SIP Bridge Configuration dialog box, which enables you to modify the settings of the SIP Bridge entry that is selected in the list box.</p> <p>From the list box, you can also select the New item. The New item displays a series of dialog boxes that enable you to define a new SIP Bridge entry. For more information on the dialog boxes, see <a href="#">Define a managed IP telephone for an Interaction SIP Bridge workstation</a>.</p> <p><b>Note:</b> Interaction SIP Bridge does not support external registrations.</p>
Location	<p>Select a defined Location.</p> <p>Locations, as defined for Customer Interaction Center, control certain aspects of IP communications, such as available codecs, a specific Interaction Media Server selection rule, and other options. For more information about Locations, see the "Regionalization" topic in the Interaction Administrator Help.</p>
Audio Protocol	<p>Select the real-time media protocol that you want this managed IP telephone to use.</p>
Time Zone	<p>Select the time zone where the remote computer that hosts Interaction SIP Bridge resides.</p>
Station Appearances	<p>Use the Add, Edit, and Remove buttons to define what extensions (stations) are assigned to this IP telephone. For more information about station appearances, see the "Managed IP Phone Configuration General" topic in Interaction Administrator Help.</p> <p><b>Note:</b> Interaction SIP Bridge does not support shared station appearances.</p>

### Important!

At this time, Genesys does not recommend configuring emergency information for remote IP telephones as no location verification standard or appropriate PSAP routing mechanism for VoIP telephony exist. For more information about the problems with emergency telephone numbers for remote IP telephones, see [Emergency telephone numbers](#).

- When you finish configuring this managed IP telephone entry, select the **OK** button.

## Defining multiple managed IP telephones for use with Interaction SIP Bridge

You can create comma-separated values (CSV) files and import them into Interaction Administrator to create multiple managed IP telephone entries.

### Note:

At this time, the import feature nor the Managed IP Phone Assistant do not support the Interaction SIP Bridge settings in the **Managed IP Phone Configuration** dialog box. If you use a CSV file to create multiple managed IP telephones, you must then configure each managed IP telephone entry for the settings that are specific to Interaction SIP Bridge.

To import multiple managed IP telephone configurations in Interaction Administrator, do the following steps:

- In the left pane of the **Interaction Administrator** window, select the **Managed IP Phone** object.
- In the right pane, right click an open area.
- In the resulting shortcut menu, select the **Managed IP Phone Assistant** item.  
The **Managed IP Phone Assistant** wizard application is displayed.
- Follow the instructions on each page of the wizard.

# Configure Interaction SIP Bridge through web interface

Once you install Interaction SIP Bridge on the computer in the remote agent network, do the following steps to configure Interaction SIP Bridge:

1. Ensure that the Internet connection for the remote network is operational.
2. On the computer with Interaction SIP Bridge in the remote network, start the VPN client software.
3. Using the VPN client software, connect to the network that enables connectivity to the Customer Interaction Center server.
4. Start Interaction SIP Bridge by double-clicking the icon on the desktop or by selecting the Interaction SIP Bridge shortcut in the **Start** menu.

**Note:**

If the page is not displayed, right click the Interaction SIP Bridge icon in the System Tray and select **Configuration** in the resulting shortcut menu.

The **Configuration-Bridge** page is displayed.

The screenshot shows the 'Interaction SIP Bridge' configuration window. The title bar includes the application name and version. The main window is divided into a sidebar and a main content area. The sidebar contains icons for 'Bridge', 'Authentication', 'Protocols', 'Media', 'Diagnostics', and 'Administration'. The main content area is titled 'Configuration' and contains two main sections: 'SIP Bridge Provisioning' and 'Managed Phone Provisioning'. The 'SIP Bridge Provisioning' section has a status indicator 'VPN adapter disconnected' in red. It includes input fields for 'Bridge Name', 'IC User Name', 'IC Password', and 'IC Server' (containing 'cic-server.example.com'). There are dropdown menus for 'LAN Adapter' (selected as 'Local Area Connection - 123.123.123.123') and 'VPN Adapter'. A 'Listen Port' field contains '343'. A 'Connect' button is located below these fields. The 'Managed Phone Provisioning' section is checked and includes fields for 'DHCPINFORM Option Code' (160), 'HTTP / HTTPS Provision Proxy Ports' (8088 and 8089), and 'SIP Registration Protocol' (TCP). 'Apply' and 'Revert' buttons are at the bottom of this section.

5. In the **Bridge Name** box, enter the name for this Interaction SIP Bridge workstation.

**Tip:**

So that you can test functionality and proper configuration, Genesys recommends that you first define the Interaction SIP Bridge workstation through Interaction Administrator. For more information, see [Define an Interaction SIP Bridge workstation](#).

6. In the **IC User Name** box, enter the name of the user IC account that the remote user will use to authenticate with the Customer Interaction Center server.  
For more information about creating users through Interaction Administrator, see the "Users" topic in *Interaction Administrator Help*.
7. In the **IC Password** box, enter the password of the specified user account.
8. In the **IC Server** box, enter the IP address or host name of the Customer Interaction Center server or Customer Interaction Center servers in a switchover pair to which this Interaction SIP Bridge workstation will register. Separate multiple addresses with commas.

**Note:**

To register with a Customer Interaction Center server, you must define this Interaction SIP Bridge workstation through Interaction Administrator for that server.

To register with an Interaction SIP Proxy server, you must define this Interaction SIP Bridge workstation on the Customer Interaction Center server that manages the Interaction SIP Proxy server. The Interaction SIP Proxy server must then connect to the Customer Interaction Center server before it will accept registration requests from Interaction SIP Bridge.

**Note:**


If Interaction SIP Bridge is currently connected to a CIC server and you want to add a second CIC server, you must first select the **Disconnect** button before you can modify the **IC Server** box by adding a comma and the second server address.

9. In the **LAN Adapter** list box, select the network connection that Interaction SIP Bridge will use to send SIP messages to the Customer Interaction Center server.
10. In the **VPN Adapter** list box, select the network connection created during the installation of the VPN client software.
11. In the **Listen Port** box, enter the network port number which the Customer Interaction Center server will use to communicate system-related updates to Interaction SIP Bridge.  
The default value is 343.
12. If you are using a managed IP telephone with Interaction SIP Bridge, do the following steps:
  - a. Ensure that the **Managed Phone Provisioning** check box is enabled.
  - b. In the **DHCPINFORM Option Code** box, enter the DHCP Option number for which Interaction SIP Bridge will broadcast a response to DHCP INFORM provisioning requests. You must also configure the managed IP telephone with the same DHCP Option number.  
The default value is 160.
  - c. In the **HTTP / HTTPS Provision Proxy Ports** boxes, enter the network ports that Interaction SIP Bridge will use to listen for provisioning requests from a managed IP telephone.  
Interaction SIP Bridge relays configuration data from the Customer Interaction Center server and sends it to the managed IP telephone through the HTTP (the first box) and HTTPS (the second box) ports.  
The default values are 8088 (HTTP) and 8089 (HTTPS).
  - d. In the **SIP Registration Protocol** list box, select the transport layer protocol that Interaction SIP Bridge will use to send SIP messages to and from the managed IP telephone..

**Tip:**




Ensure that all firewalls between the Customer Interaction Center server and Interaction SIP Bridge do not block the ports that you specify in this procedure.

13. Select the **Apply** button.
14. Select the **Connect** button.  
Interaction SIP Bridge displays the **Untrusted Certificate** page.



# Interaction SIP Bridge


Version XXXXXXXXXX







[Status](#) [Config](#) [Logout](#)


---


## Configuration


 **Bridge**

 **Authentication**

 **Protocols**

 **Media**

 **Diagnostics**

 **Administration**

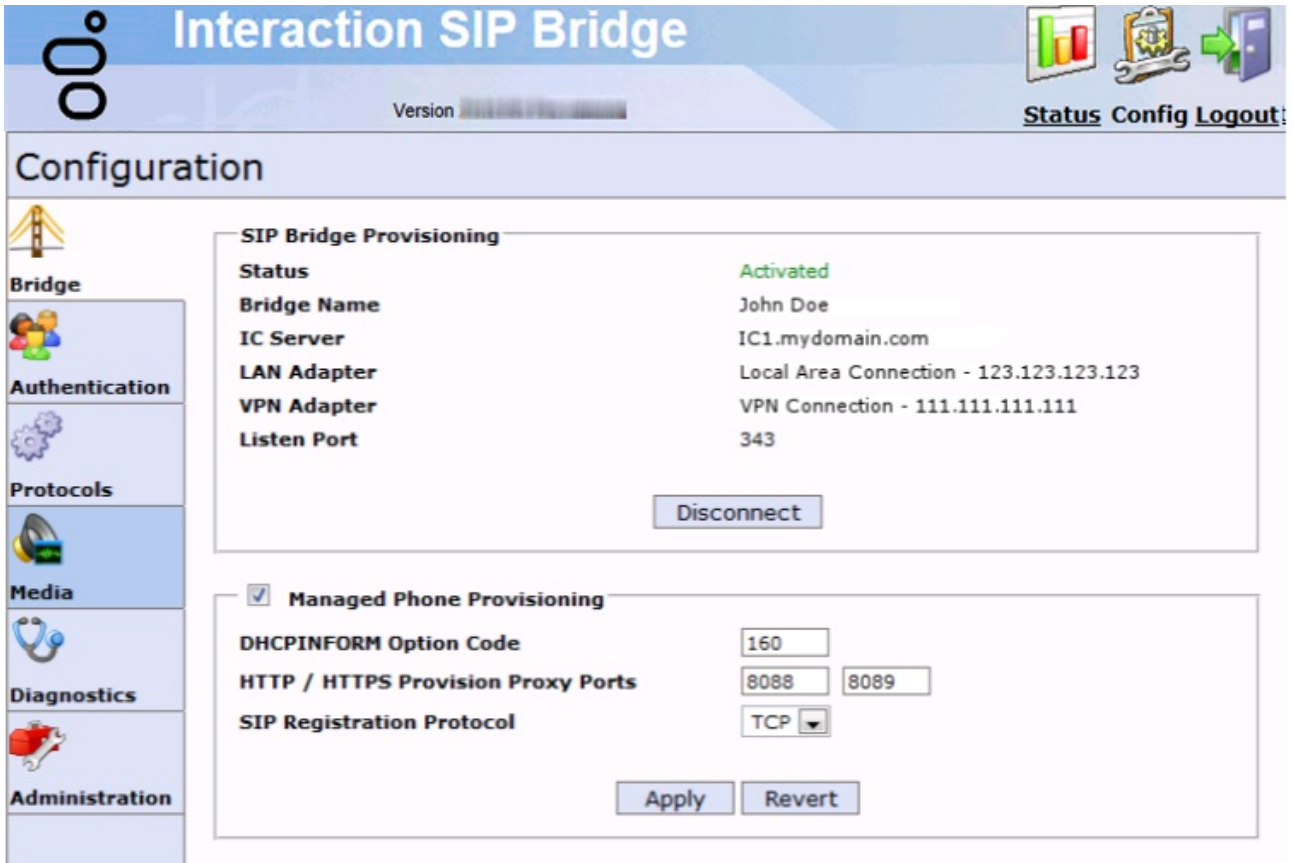
### Untrusted Certificate

The connection to https://123.123.123.123:8089 is not trusted. The server's certificate must be verified in order to continue. The full certificate chain in PEM format can be downloaded [here](#).

<b>Issued By:</b>	
<b>Organization:</b>	Servers
<b>Org. Unit:</b>	Server Group
<b>Common Name:</b>	IC1_Server Group
<b>Issued To:</b>	
<b>Organization:</b>	Servers
<b>Org. Unit:</b>	Server Lines (021058244116)
<b>Common Name:</b>	IC1
<b>X509 version</b>	v3
<b>Signature</b>	sha1WithRSAEncryption
<b>Serial Number</b>	1234567890
<b>Validity</b>	2011-10-18 02:58:06 GMT to 2031-10-19 02:58:06 GMT

15. Select the **Trust** button.

Interaction SIP Bridge connects to the Customer Interaction Center server and displays the **Bridge** page.



The screenshot shows the 'Interaction SIP Bridge' configuration interface. At the top, there is a logo on the left, the title 'Interaction SIP Bridge' in the center, and navigation links 'Status', 'Config', and 'Logout' on the right. Below the title bar is a 'Configuration' section with a left-hand navigation menu containing icons and labels for 'Bridge', 'Authentication', 'Protocols', 'Media', 'Diagnostics', and 'Administration'. The main content area is divided into two sections:

- SIP Bridge Provisioning:** This section shows the bridge is 'Activated'. It lists the following details:
  - Bridge Name: John Doe
  - IC Server: IC1.mydomain.com
  - LAN Adapter: Local Area Connection - 123.123.123.123
  - VPN Adapter: VPN Connection - 111.111.111.111
  - Listen Port: 343
 A 'Disconnect' button is located at the bottom of this section.
- Managed Phone Provisioning:** This section is checked with a checkbox. It includes:
  - DHCPINFORM Option Code: 160
  - HTTP / HTTPS Provision Proxy Ports: 8088 and 8089
  - SIP Registration Protocol: TCP (selected from a dropdown menu)
 'Apply' and 'Revert' buttons are at the bottom of this section.

Interaction SIP Bridge registers the IP telephone, relays the provisioning information (if using a managed IP telephone), and begins relaying SIP messages between the Customer Interaction Center server and the IP telephone.

**Note:**

Restart the managed IP telephone if it was operational while you did this procedure to ensure that it receives the most current configuration data from the Customer Interaction Center server through Interaction SIP Bridge.

# Configure additional IP telephone settings

To ensure that the IP telephone that you use with Interaction SIP Bridge functions properly, you must ensure that it is configured with the appropriate values, not only for communications with Interaction SIP Bridge but for your Customer Interaction Center network. In both the Customer Interaction Center suite of products and the IP telephone, many settings default to a value specified by industry standards. However, if your Customer Interaction Center network requires special network ports for one or more protocols, different Quality of Service (QoS) Differentiated Services Code Point (DSCP) markings, custom timeout periods, or other aspects that differ from default values, you must ensure that you configure the settings on the IP telephone with those same values.

For managed IP telephones, you use Interaction Administrator to configure many of these settings through the corresponding entry in the **Managed IP Phones** container. However, there are other settings for the IP telephone, such as the codec, that are not addressed by the managed IP telephone configuration.

## Important!

After you configure settings through the interface of the IP telephone, you must restart the IP telephone.

### Note:

VoIP telephony names for settings can vary among IP telephone brands and the products in the Customer Interaction Center network. For more information about VoIP telephone names for settings for a particular product, see the documentation for that specific product.

## Media settings

Protocols	Affected devices, products, and entities	Media settings
<ul style="list-style-type: none"><li>Real-time Transport Protocol (RTP)</li><li>Real-time Control Protocol (RTCP)</li><li>Secure Real-time Transport Protocol (SRTP)</li><li>Secure Real-time Transport Control Protocol (SRTCP)</li></ul>	<ul style="list-style-type: none"><li>IP telephones</li><li>Softphones</li><li>VoIP gateways, including Interaction Edge and Interaction Gateway</li><li>Interaction SIP Bridge</li><li>Interaction SIP Proxy</li><li>Interaction Media Server</li><li><b>Managed IP Phones</b> in Interaction Administrator</li><li><b>Lines</b> in Interaction Administrator</li><li><b>Locations</b> in Interaction Administrator</li></ul>	<ul style="list-style-type: none"><li>RTP/RTCP, SRTP/RTCP, or SRTCP/SRTCP protocol</li><li>Use QoS</li><li>QoS DSCP</li><li>Codec, such as G.711 or G.729</li><li>SRTP encryption cipher</li><li>Dual-Tone Multi-Frequency (DTMF) PureConnect products support the RFC 2833 specification for DTMF.</li></ul>



## Call control settings

Protocols	Affected devices, products, and entities	Call control settings
Session Initiation Protocol (SIP)	<ul style="list-style-type: none"> <li>IP telephones</li> <li>Softphones</li> <li>VoIP gateways, including Interaction Edge and Interaction Gateway</li> <li>Interaction SIP Bridge</li> <li>Interaction SIP Proxy</li> <li><b>Managed IP Phones</b> in Interaction Administrator</li> <li><b>Lines</b> in Interaction Administrator</li> </ul>	<ul style="list-style-type: none"> <li>Transport protocol TCP, UDP, or TLS (over TCP)</li> <li>Use QoS</li> <li>QoS DSCP</li> <li>Transport layer protocol (TCP, UDP) port number for the SIP protocol:</li> <li>SIP (Customer Interaction Center default value for &lt;Stations-TCP&gt; and &lt;Stations-UDP&gt;: 8060)</li> <li>SIP through TLS over TCP (Customer Interaction Center default value for &lt;Stations-TLS&gt;: 8061)</li> </ul> <div style="border: 1px solid black; padding: 5px; margin: 5px 0;"> <p><b>Note:</b> If you create a new SIP line in Interaction Administrator for Interaction SIP Bridge workstations, you must specify a different port number from 8060/8061. You must also configure the new port number value in both Interaction SIP Bridge and the IP telephone.</p> </div> <ul style="list-style-type: none"> <li>SIP timers</li> <li>T1</li> <li>T2</li> <li>TCPConnect (timeout)</li> <li>Retransmissions:</li> <li>MaxPacketRetry</li> <li>MaxInviteRetry</li> <li>RetryOnRespondCode</li> <li>Security features:</li> <li>EnableMutualTLS</li> <li>TLSCipherSuites</li> </ul>

## Authentication settings

Protocols	Affected devices, products, and entities	Authentication settings
Session Initiation Protocol (SIP)	<ul style="list-style-type: none"> <li>IP telephones</li> <li>Softphones</li> <li>VoIP gateways, including Interaction Edge and Interaction Gateway</li> <li>Interaction SIP Bridge</li> <li>Interaction SIP Proxy</li> <li><b>Managed IP Phones</b> in Interaction Administrator</li> <li><b>Lines</b> in Interaction Administrator</li> </ul>	<ul style="list-style-type: none"> <li>Use SIP authentication</li> <li>SIP user ID (registration ID)</li> <li>SIP authentication user name</li> <li>SIP authentication password</li> </ul>

## Registration settings

Protocols	Affected devices, products, and entities	Registration settings
Session Initiation Protocol (SIP)	<b>Managed IP Phones</b> in Interaction Administrator	Registration interval

# DHCP settings

Protocols	Affected devices, products, and entities	DHCP settings
Dynamic Host Control Protocol (DHCP)	<ul style="list-style-type: none"> <li>• IP telephones</li> <li>• Softphones</li> <li>• Ethernet switches</li> <li>• DHCP servers</li> <li>• <b>Lines</b> in Interaction Administrator</li> </ul>	<ul style="list-style-type: none"> <li>• Boot Server (default: Custom + Opt. 66)</li> <li>• BootSrv Opt (default: 160)</li> <li>• DHCP BootSrv Type (default: String)</li> <li>• ServerType (default: HTTP)</li> <li>• ServerAddr: (default: blank)</li> <li>• You can find all other default DHCP Option numbers at the <a href="#">IANA website</a>.</li> </ul>

# Use Interaction SIP Bridge

On a computer, Interaction SIP Bridge simply serves as a relay for IP-based calls between the associated IP telephone and the Customer Interaction Center network. After you initially configure Interaction SIP Bridge through its web interface, the only indication that it is running is an icon in the Notification Area of the Windows Task Bar.



## Start Interaction SIP Bridge

1. Ensure that the computer is connected to the Internet.
2. Start the VPN client software.
3. Supply your authentication credentials in the VPN client software and connect to the network that provides access to the Customer Interaction Center server.
4. Start Interaction SIP Bridge by one of the following methods:
  - Double click the **Interaction SIP Bridge** icon on the Desktop
  - Select the **Interaction SIP Bridge** item in the **Start** menu.

The Interaction SIP Bridge icon is displayed in the Notification Area of the Windows Task Bar.

## Interaction SIP Bridge operational indicators

The Interaction SIP Bridge icon in the Notification Area of the Windows Task Bar can display the following states to represent its operational state:

State	Icon	Description
Ready		This icon indicates that Interaction SIP Bridge is running, is connected to the Customer Interaction Center, and the associated IP telephone has registered with Interaction SIP Bridge.
Not Ready		This icon indicates that Interaction SIP Bridge is not ready for one or more of the following reasons: <ul style="list-style-type: none"><li>• Initial configuration of Interaction SIP Bridge has not been completed.</li><li>• Interaction SIP Bridge cannot establish a connection to the specified Customer Interaction Center server.</li><li>• The associated IP telephone has not yet registered with Interaction SIP Bridge.</li><li>• The SIP Bridge is not registered with any of the servers in its registration group.</li></ul>

# Interaction SIP Bridge web interface reference

This section provides reference information for the controls in the web interface of Interaction SIP Bridge.

- [Status - Bridge page](#)
- [Status - Sessions page](#)
- [Status - About page](#)
- [Configuration - Bridge page](#)
- [Configuration - Authentication page](#)
- [Configuration - Protocols page](#)
- [Configuration - Media page](#)
- [Configuration - Diagnostics page](#)
- [Configuration - Administration page](#)

## Status - Bridge page


The screenshot shows the 'Status' page of the Interaction SIP Bridge. The header includes the logo, title, version, and navigation links (Status, Config, Logout). The left sidebar contains 'Bridge', 'Sessions', and 'About'. The main content area is divided into three sections: 'Network Adapters' (LAN Adapter: Local Area Connection 2 - 192.168.1.102; VPN Adapter: Local Area Connection - 123.123.123.123), 'IC Servers' (cic-server.example.com, Last update at 2015-07-17 17:00:40), and 'SIP Registrations' (Lab1, From: 192.168.1.101:5060, Expires at 2015-07-18 12:56:11; To: 123.123.123.123:8060, Expires at 2015-07-19 09:58:11). A 'Refresh' button is located at the bottom of the SIP Registrations section.

Control	Purpose
<b>Network Adapters - LAN Adapter</b>	This character string displays the Windows network connection that Interaction SIP Bridge is using for IP communication on the local area network (LAN).
<b>Network Adapters - VPN Adapter</b>	This character string displays the Windows network connection that Interaction SIP Bridge is using for IP communications on the virtual private network (VPN).
<b>IC Servers</b>	This box displays the Customer Interaction Center (CIC) server or Interaction SIP Proxy server to which Interaction SIP Bridge is registered.
<b>SIP Registrations</b>	This box displays the SIP-based IP telephone that is registered with this Interaction SIP Bridge workstation.
<b>Refresh button</b>	Select this button to display the current data for all controls on this page.

## Status - Sessions page




Control	Purpose
<b>Duration</b>	This column displays the amount of time that a session has been active. The table sorts active sessions based on the elapsed time with the longest session listed first.
<b>To</b>	This column displays the Display Name portion of the dialed SIP address. Interaction SIP Bridge extracts this information directly from the SIP To header of the originating SIP INVITE message. If the Display Name is not available, this column displays the User portion.
<b>From</b>	This column displays the Display Name of the calling party. Interaction SIP Bridge extracts this information directly from the SIP From header of the originating SIP INVITE message.
<b>Call ID</b>	This column displays a truncated version of the unique SIP call identifier that the initiating party assigns to this call. Interaction SIP Bridge extracts this information from the SIP Call-ID header of the originating SIP INVITE message. You can click a Call ID to display detailed information for the associated session.
<b>Filter box</b>	Enter a string of characters for which you want to display only the sessions that contain the same string.
<b>Filter button</b>	Select this button to display only the session that contain the string of characters in the <b>Filter</b> box.
<b>Refresh button</b>	Select this button to display the current data for all current sessions. <div style="border: 1px solid black; padding: 5px; margin-top: 10px;"> <p><b>Note:</b> If you select the <b>Refresh</b> button, any current filters are removed.</p> </div>

## Status - About page




# Interaction SIP Bridge

Version 10.0.0.0

Status [Config](#) [Logout](#)

## Status



Bridge


**Product Version** 10.0.0.0

**File Version** 10.0.0.0 built on 10/10/2018

**Machine Name** Golf

**License**

**Running Since** 2018-02-08 11:35:03 (47m 12s)



Sessions


**Product Version** 10.0.0.0

**File Version** 10.0.0.0 built on 10/10/2018


**Machine Name** Golf

**License**

**Running Since** 2018-02-08 11:35:03 (47m 12s)



About



Copyright ©1997-2011 [Genesys](#). All Rights Reserved.  
This program is protected by US and international copyright laws.

Control	Purpose
<b>Product Version</b>	This number represents the major and minor version numbers of this current installation of Interaction SIP Bridge.
<b>File Version</b>	This string of characters consists of four period-separated fields: the major and minor version numbers the product code, and the patch level.
<b>Machine Name</b>	This string of characters is the name of the computer on which Interaction SIP Bridge is running.
<b>License</b>	This string of characters displays the type of Interaction SIP Bridge license that this installation uses. If the license has an expiration date, it is also displayed. Interaction SIP Bridge licenses are a component of the CIC server license.
<b>Running Since</b>	This string of characters displays the date and time when Interaction SIP Bridge was last started and the elapsed time of this session.
<b>Refresh button</b>	Select this button to refresh the data displayed on this page.

## Configuration - Bridge page

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## Configuration



Bridge



Authentication



Protocols



Media



Diagnostics



Administration

### SIP Bridge Provisioning

**Status** Activated

**Bridge Name** JohnDoe SIP Bridge

**IC Server** cic-server.example.com

**LAN Adapter** Local Area Connection 2 - 192.168.1.102

**VPN Adapter** Local Area Connection - 123.123.123.123

**Listen Port** 343

Disconnect

**Managed Phone Provisioning**

**DHCPINFORM Option Code**

**HTTP / HTTPS Provision Proxy Ports**

**SIP Registration Protocol**

Apply

Revert

Control	Usage
<b>Status</b>	This character string displays the current operational state of Interaction SIP Bridge.
<b>Bridge Name</b>	This character string displays the name that you assigned to this Interaction SIP Bridge workstation in both Interaction Administrator and during the initial configuration of Interaction SIP Bridge.
<b>IC Server</b>	This character string displays the name of the Customer Interaction Center (CIC) server to which this Interaction SIP Bridge is connected.
<b>LAN Adapter</b>	This character string displays the Windows network connection that Interaction SIP Bridge is using for IP communication on the local area network (LAN).
<b>VPN Adapter</b>	This character string displays the Windows network connection that Interaction SIP Bridge is using for IP communications on the virtual private network (VPN).
<b>Listen Port</b>	This character string displays the network port number to which the Customer Interaction Center server uses to communicate system-related updates to Interaction SIP Bridge.  The default value is 343.
<b>Disconnect button</b>	Select this button to disconnect Interaction SIP Bridge from the Customer Interaction Center server to which it is currently connected.
<b>Managed Phone Provisioning check box</b>	<p>Enable this check box to use Interaction SIP Bridge with a managed IP telephone as defined through Interaction Administrator.</p> <div data-bbox="370 814 1498 909" style="border: 1px solid black; padding: 5px;"> <p><b>Note:</b> Interaction SIP Bridge currently supports only managed IP telephones.</p> </div> <p>For information on IP telephones that Customer Interaction Center and Interaction SIP Bridge supports as managed IP telephones, see <a href="http://testlab.genesys.com">http://testlab.genesys.com</a>.</p>
<b>DHCPINFORM Option Code</b>	<p>Enter the DHCP Option number that the IP telephone uses in a DHCPINFORM query for attaining the address of a provisioning server.</p> <p>The default value is <b>160</b>.</p>
<b>HTTP/HTTPS Provision Proxy Ports</b>	<p>Enter the port numbers that Interaction SIP Bridge uses for the HTTP and HTTPS protocols for transferring firmware and configuration files for the IP telephone.</p> <p>The default values are <b>8088</b> and <b>8089</b>.</p>
<b>SIP Registration Protocol</b>	<p>Select the transport layer protocol that the IP telephone uses to relay SIP messages.</p> <p>The default value is <b>TCP</b>.</p>
<b>Apply button</b>	Select this button to save any changes.
<b>Revert button</b>	Select this button to discard any changes and display the previous values.



## Configuration - Authentication page

Important!

This page configures authentication settings for use with an unmanaged IP telephone, also known as a station in Interaction Administrator. Interaction SIP Bridge does not currently support this feature.

The screenshot shows the 'Configuration - Authentication' page in the Interaction SIP Bridge interface. The page has a blue header with the logo and title 'Interaction SIP Bridge'. Below the header, there are navigation links: 'Status', 'Config', and 'Logout'. The main content area is titled 'Configuration' and contains a table with one entry, 'HQLab1'. The table has columns for 'User name' and 'Password'. The 'User name' is '203cde6952744edaacdf0a5b8...' and the 'Password' is '\*\*\*\*\*'. There are 'Edit' and 'Del' buttons next to each field. Below the table, there is a 'Page:' indicator showing '1' and '[ 1 total item ]'. There is also a 'Filter:' input field with 'Filter' and 'Reset' buttons. On the left side, there is a sidebar with navigation icons and labels: 'Bridge', 'Authentication', 'Protocols', 'Media', 'Diagnostics', and 'Administration'.

Interaction SIP Bridge

Version 2.0.0.0

Status Config Logout

### Configuration

Bridge

Authentication

Protocols

Media

Diagnostics

Administration

	User name	Password	
HQLab1	203cde6952744edaacdf0a5b8...	*****	Edit Del

Page: 1 [ 1 total item ]

Filter:  Filter Reset

## Configuration - Protocols page



## Configuration

-   
**Bridge**
-   
**Authentication**
-   
**Protocols**
-   
**Media**
-   
**Diagnostics**
-   
**Administration**

**UDP**

Port Number	<input type="text" value="8060"/>
Timer T1 (ms)	<input type="text" value="500"/>
Timer T2 (ms)	<input type="text" value="1000"/>
Max SIP INVITE Retries	<input type="text" value="3"/>
Max Packet Retries	<input type="text" value="4"/>

**TCP**

Port Number	<input type="text" value="8060"/>
Connect Timeout (sec)	<input type="text" value="2"/>

**TLS**

Port Number	<input type="text" value="8061"/>
Mutual Authentication	<input type="text" value="No"/>
Allowed TLS Cipher suites	<input checked="" type="checkbox"/> TLS_RSA_WITH_AES_256_CBC_SHA <input checked="" type="checkbox"/> TLS_RSA_WITH_AES_128_CBC_SHA <input type="checkbox"/> TLS_RSA_WITH_3DES_EDE_CBC_SHA <input type="checkbox"/> TLS_RSA_WITH_DES_CBC_SHA <input type="checkbox"/> TLS_RSA_WITH_RC4_128_SHA <input type="checkbox"/> TLS_RSA_EXPORT_WITH_DES40_CBC_SHA
Certificate Installed	<b>No</b>
Trusted Authorities Installed	0

Control	Usage
<b>UDP check box</b>	Enable this check box to if you to give Interaction SIP Bridge the ability to send SIP messages using the UDP transport layer protocol.
<b>Port Number</b>	Enter the port number on which Interaction SIP Bridge will listen for SIP messages using the UDP transport layer protocol. The default value is <b>8060</b> .
<b>Timer T1 (ms)</b>	Enter the number of milliseconds that represents the initial incremental delay before Interaction SIP Bridge considers a transmission for which it receives no acknowledgement canceled and then terminates the pending transaction.This is also known as the Round Trip Time (RTT) estimate. The default value is <b>500</b> .
<b>Timer T2 (ms)</b>	Enter the maximum number of milliseconds that that Interaction SIP Bridge will wait between packet retransmissions for non-INVITE transactions. The default value is <b>1000</b> .
<b>Max SIP INVITE Retries</b>	Enter the maximum number of attempts that Interaction SIP Bridge will send a SIP INVITE request before it considers the destination to be unreachable. The default value is <b>3</b> .
<b>Max Packet Retries</b>	Enter the maximum number of attempts that Interaction SIP Bridge will send a non-INVITE SIP request before it considers the destination to be unreachable. The default value is <b>4</b> .
<b>TCP check box</b>	Enable this check box to if you to give Interaction SIP Bridge the ability to send SIP messages using the TCP transport layer protocol.
<b>Port Number</b>	Enter the port number on which Interaction SIP Bridge will listen for SIP messages using the TCP transport layer protocol. The default value is <b>8060</b> .
<b>Connect Timeout (sec)</b>	Enter the maximum number of seconds that Interaction SIP Bridge will wait for a TCP connect request to complete before it abandons the attempt. The default value is <b>2</b> .
<b>TLS check box</b>	Enable this check box to if you to give Interaction SIP Bridge the ability to send SIP messages using the TLS application layer protocol (authenticated and, optionally, encrypted) on the TCP transport layer protocol.
<b>Port Number</b>	Enter the port number on which Interaction SIP Bridge will listen for SIP messages using TLS on TCP.
<b>Mutual Authentication</b>	Select <b>Yes</b> from this list box if you want Interaction SIP Bridge to require an endpoint to present an authenticated certificate before it completes establishing the TLS connection. The default value is <b>No</b> .
<b>Allowed TLS Cipher suites</b>	If you want to use encryption for SIP messages using the TLS protocol, enable the cipher suites that Interaction SIP Bridge will accept and use for transmission and receipt of SIP messages.
<b>Certificate Installed</b>	This string of characters displays if a certificate has been installed for Interaction SIP Bridge. Interaction SIP Bridge must have a certificate before it can use TLS for SIP messages. Interaction SIP Bridge generates this certificate when it registers with a server, such as Customer Interaction Center or Interaction SIP Proxy.You define these registration servers on the <b>Configuration - Bridge</b> page.
<b>Trusted Authorities Installed</b>	This number indicates the total trusted certificate authorities defined on this Interaction SIP Bridge workstation.These authorities are created for each registration server (Customer Interaction Center, Interaction SIP Proxy) that you define on the <b>Configuration Bridge</b> page.

## Configuration - Media page

This page enables you to configure customized settings for the media protocols.

**Interaction SIP Bridge**  
Version 2018 R3 Pre-release  
[Status](#) [Config](#) [Logout](#)

### Configuration

**Media Properties**

Name	Value	Delete
Select name of property to add	<input type="text"/>	<input type="button" value="Add"/>

### Add a new property value to Interaction SIP Bridge

To add a new property and value, do the following steps:

1. In the **Select name of property to add** list box, select the property that you want to add.
2. To the right of the list box, select the **Add** button.

The property and the default value are displayed in the table above the list box.

**Note:**

Items that are displayed in red require you to restart Interaction SIP Bridge before the changes take effect.

3. In the value control for the newly-added property, enter an appropriate value.
4. Select the **Apply** button.

---

## Interface controls

Control	Usage
<b>Name</b>	This column displays the name of an added property.
<b>Value</b>	This column displays the current setting of the associated property.
<b>Delete</b> check box	Select this check box to remove an existing property in the list. You must then select the <b>Apply</b> button to remove the property.
<b>Select name of property to add</b>	In this list, select the media protocol property that you want to customize.
<b>Add</b> button	Select this button to add the selected property from the adjacent list box.
<b>Apply</b> button	Select this button to enable the added property and value combinations.
<b>Revert</b> button	Select this button to discard all added properties and values that you added since the last use of the <b>Apply</b> button.

## Media protocol properties

Property	Description and settings
<b>RtpQosTaggingEnabled</b>	This property enables the QoS feature for RTP communications on Interaction SIP Bridge. This feature is enabled by default. You do not need to add this property and set it to <b>true</b> .
<b>RtpQosDscpValue</b>	This property sets the QoS DSCP marking that Interaction SIP Bridge writes to each RTP packet. By default, this value is <b>2E (46, 101110) EF</b> . You do not need to add this property and set it to this value.
<b>RtcpQosTaggingEnabled</b>	This property enables the QoS feature for RTCP communications on Interaction SIP Bridge. This feature is enabled by default. You do not need to add this property and set it to <b>true</b> .
<b>RtcpQosDscpValue</b>	This property sets the QoS DSCP marking that Interaction SIP Bridge writes to each RTCP packet. By default, this value is <b>2E (46, 101110) EF</b> . You do not need to add this property and set it to this value.
<b>RtpPortRange</b>	<p>Enter the range of ports that Interaction SIP Bridge monitors for incoming RTP communications.</p> <p>Select a port range that does not conflict with ports in use by any other applications on the network. Genesys recommends a port range of 16384 to 32767 (default values).</p> <p>Interaction SIP Bridge requires four ports for a single telephone call.</p>
<b>RtpDiagnosticCaptureEnabled</b>	<p>Interaction SIP Bridge can record all RTP/RTCP packets in communications that it relays and save that information in a file with additional metadata. This is known as a <i>diagnostic captures</i>, which someone can later analyze in the process of troubleshooting problems.</p> <p>Interaction SIP Bridge stores diagnostic captures in the log file directory of the computer in the following format:</p> <p><i>CallID_ResourceID_RandomDigits.hpaacap</i></p> <p>For more information about log files, see <a href="#">Interaction SIP Bridge log files</a>.</p> <p>If you are using SRTP for secure communications, Interaction SIP Bridge stores the cryptographic keys for decrypting the audio data within the capture file as metadata. To limit disclosure of these keys, Interaction SIP Bridge can optionally encrypt the capture file when it is written. Interaction SIP Bridge also adds a keyed authentication tag to ensure integrity. You set these security options through the <b>RtpDiagnosticCaptureCipherSuite</b> property.</p> <p><b>Important!</b></p> <p>Enable this property only if you are actively troubleshooting a communication problem. Do not create diagnostic captures on a continual basis as the resulting files consume storage space. Genesys recommends that you enable this property only when directed by a PureConnect Customer Care representative.</p>
<b>RtpDiagnosticCaptureCipherSuite</b>	<p>This property sets the encryption algorithm that Interaction SIP Bridge uses for encrypting diagnostic captures. It supports the following values:</p> <ul style="list-style-type: none"> <li><b>NULL_NULL</b> - This is the default value. Interaction SIP Bridge does not encrypt or authenticate diagnostic captures.</li> </ul> <p><b>NULL_HMACSHA1</b> - This value indicates that Interaction SIP Bridge does not encrypt diagnostic captures but provides an authentication mechanism.</p> <ul style="list-style-type: none"> <li><b>AES128_HMACSHA1</b> - Interaction SIP Bridge encrypts diagnostic captures with a 128-bit cipher and provides an authentication mechanism.</li> <li><b>AES256_HMACSHA1</b> - Interaction SIP Bridge encrypts diagnostic captures with a 256-bit cipher and provides an authentication mechanism.</li> </ul>

## Configuration - Diagnostics page



## Configuration

- **Bridge**
- **Authentication**
- **Protocols**
- **Media**
- **Diagnostics**
- **Administration**

**General**

**Configuration file** [I3SIPBridgeConfig.xml](#)

**Syslog receiver**

**Diagnostic Logs**

**Location** [C:/Users/GOLF\\_U~1/AppData/Local/Temp/inin\\_tracing](#)

**Tracing Topics**

**Retention** 2 days

**Compression**

Control	Usage
<b>Configuration file</b>	This hyperlink displays the current configuration file for Interaction SIP Bridge. Use this link to download and inspect the configuration file or to provide it to a Genesys Customer Care representative.
<b>Syslog receiver</b>	<p>Syslog is a standard (RFC 5424) method of recording operational events and messages for a computer or system and is a feature of Interaction SIP Bridge. This feature generates a <code>syslog.ininlog</code> file.</p> <p>To enable this feature and to have Interaction SIP Bridge immediately start recording data to this log file, select <b>Enable</b> from the <b>Syslog receiver</b> list box and select the <b>Apply</b> button.</p> <p>The default value is <b>Disable</b>.</p> <p><b>Note:</b> The syslog receiver supports only the UDP protocol and monitors for communications on port 514.</p>
<b>Apply button</b>	Select this button to enable the currently selected mode in the <b>Syslog receiver</b> list.
<b>Location</b>	This hyperlink displays the directory where Interaction SIP Bridge is currently storing log files.
<b>Tracing Topics - View button</b>	<p>Select this button to display all Interaction SIP Bridge tracing topics and their current levels of data collection.</p> <p>Tracing topics are the functional areas of Interaction SIP Bridge for which data can be written to log files.</p> <p>The default level for all tracing topics is <b>Status</b>.</p> <p><b>Caution!</b> Genesys recommends that you do not change topic tracing levels as doing so can cause severe performance degradation. You should only change topic tracing levels at the direction of a PureConnect Customer Care representative.</p>
<b>Retention</b>	<p>This number displays the number of days that Interaction SIP Bridge will store trace log files. Interaction SIP Bridge deletes trace log files that surpass this time limit.</p> <p>You can configure the number of days by changing the <code>ININ_TRACE_LOG_RETENTION</code> Windows environment variable. If you change this value, you must restart Windows for the change to take effect.</p> <p>The default value is <b>7</b>.</p>
<b>Compression</b>	<p>Select <b>Yes</b> from this list box if you want Interaction SIP Bridge to compress trace log files.</p> <p>The default value is <b>Yes</b>.</p>
Log compression start time box	In the box to the right of the <b>Compression</b> list box, enter the time (in 24-hour format) that you want Interaction SIP Bridge to compress trace log files.
<b>Apply button</b>	Select this button if you have changed the status of the <b>Compression</b> feature or the time when the compression process occurs.

## Configuration - Administration page

By default, Interaction SIP Bridge does not have a user name and password. This enables you to configure Interaction SIP Bridge with the necessary information that it requires to communicate with the IP telephone and the Customer Interaction Center server. If you have configured Interaction SIP Bridge and want to prevent either the user of the workstation or any other person from modifying the configuration, you can set a new user name and password. Afterwards, any user that attempts to view the web interface of Interaction SIP Bridge is prompted for the Interaction SIP Bridge user name and password.





## Configuration

-  **Bridge**
-  **Authentication**
-  **Protocols**
-  **Media**
-  **Diagnostics**
-  **Administration**

**Change Login Credentials**

**Old User Name**

**Old Password**

**New User Name**

**New Password**

**Confirm New Password**

Control	Usage
<b>Old User Name</b>	Enter the previous Interaction SIP Bridge user name. The default is <i>&lt;blank&gt;</i> .
<b>Old Password</b>	Enter the previous Interaction SIP Bridge password for the specified user name. The default is <i>&lt;blank&gt;</i> .
<b>New User Name</b>	Enter the new user name that you want to use for allowing access to the Interaction SIP Bridge configuration interface. <b>Note:</b> This user name is not related to any other user name, such as the IP telephone or any entity in Customer Interaction Center.
<b>New Password</b>	Enter the password for the new user name.
<b>Confirm New Password</b>	Re-enter the password for the new user name.
<b>Apply button</b>	Select this button to overwrite the previous Interaction SIP Bridge user name and password with the new values. <b>Important!</b> Once you select the <b>Apply</b> button, you cannot use the <b>Revert</b> button to restore the old user name and password.
<b>Revert button</b>	Select this button to reverse any changes that you made since you last selected the <b>Apply</b> button.

# Emergency telephone numbers

Public telephone networks use emergency telephone numbers that provide the name, telephone number, and a location or address to a local Public Safety Answering Point (PSAP). Some examples of emergency telephone numbers are Enhanced 911 (E911) in the United States, 112/E112 in the European Union, and 999 in multiple nations.

Conventional telephony systems, such as the Public Switch Telephone Network (PSTN) and cellular telephone networks provide an address of record or an approximate location in relation to a cellular tower. In some cases of cellular telephony, Global Positioning System (GPS) location information is also sent. Currently, VoIP telephony does not support the verification of a location for emergency calls.

Some standards exist that support emergency calls through VoIP. These standards require an administrator to provide information when the VoIP device is defined in a system, such as managed IP telephones in Customer Interaction Center, or when an agent registers a VoIP telephony device to a call distribution server. Although no official international standards regarding the verification of a location for emergency calls through VoIP have yet been established, many have been proposed or are in development. Some third-party companies provide emergency call routing and location information through VoIP to the appropriate PSAP for enterprise systems and remote workers.

In Customer Interaction Center, you can enter emergency information when you define a managed IP telephone; however, if a remote agent uses that IP telephone to dial an emergency telephone number, Customer Interaction Center—depending on the configured dial plan—routes the call through a gateway that is most likely not in the same location as the remote agent or does not route the call at all. If an emergency call is allowed through the dial plan, the gateway routes the emergency call to the PSTN, which then relays the call to the PSAP that is local to that gateway. The result is that a remote agent could be connected to a PSAP that does not cover the area in which the remote agent made the call.

For remote agents, including those who use Interaction SIP Bridge, Genesys recommends one of the following solutions for supporting emergency calls through VoIP:

- Remote agents must use PSTN or cellular telephony devices to make emergency calls.
- Use a third-party E911 service provider to provide emergency call routing to a PSAP that services the location based on previously-submitted location information of the managed IP telephone.

**Note:**

CIC supports E911 call routing through a third-party E911 service provider if an emergency call is made through a CIC client, such as Interaction Connect. For more information about E911, see [E911 Technical Reference](#) in the [PureConnect Documentation Library](#).

# Troubleshooting

- [Interaction SIP Bridge log files](#)
- [Cannot connect to CIC server](#)
- [Interaction SIP Bridge registers but the IP telephone does not](#)
- [Poor audio quality](#)
- [Calls fail after first attempt but succeed on second attempt](#)
- [License failure](#)
- [IP telephone does not display correct time and date](#)
- [IP telephone is not receiving firmware updates](#)

## Interaction SIP Bridge log files

By default Interaction SIP Bridge stores log data in the following file:

```
%LOCALAPPDATA%\inin_tracing\interactionsipbridge<n>.ininlog
```

%LOCALAPPDATA% is a variable that identifies the log file directory that Interaction SIP Bridge created during installation. By default, this variable equates to the following directory path:

```
C:\Users\<<UserName>\AppData\Local
```

If you experience recurring problems with Interaction SIP Bridge, contact PureConnect Customer Care. The PureConnect Customer Care representative will ask for the log file to determine the problem. You may be asked to change the tracing levels through the [Configuration - Diagnostics page](#). Do not change tracing levels unless directed to do so by a PureConnect Customer Care representative.

If you want to change the name and location of the log file, you can modify the `ININ_TRACE_ROOT` environment variable of the Windows operating system.

## Cannot connect to CIC server

If Interaction SIP Bridge cannot register with the Customer Interaction Center server, consider the following actions:

- Ensure that the Internet connection of the remote computer is functional.
- Ensure that the VPN client software is running and that it has an authorized connection to the network through which Interaction SIP Bridge connects to the Customer Interaction Center server.
- Ensure that you entered the address of the Customer Interaction Center server correctly on the **Configuration Bridge** page of the Interaction SIP Bridge web interface. If you entered a host name and cannot connect, add the domain to the address, such as `CIC1.mydomain.com`.
- Ensure that you created an entry for this Interaction SIP Bridge workstation in the **SIP Bridge** container of Interaction Administrator.
- Use the Windows **ping** command to verify that the Customer Interaction Center server is operational and reachable.
- If the Internet connection of the remote computer running Interaction SIP Bridge resets and the Internet Service Provider issues a new external IP address, you must re-establish the VPN connection.
- If you use a switchover pair of Customer Interaction Center servers and the primary server fails, you must have configured Interaction SIP Bridge to register with the failover server or a dedicated Interaction SIP Proxy server that services the Customer Interaction Center server. For more information about the Customer Interaction Center switchover system, see *CIC Automated Switchover System Technical Reference* in the PureConnect Documentation Library.

## Interaction SIP Bridge registers but the IP telephone does not

This problem occurs if the VPN connection contains routes for the same network subnet. Because of this routing conflict, Windows cannot determine which adapter—VPN or LAN—to use to reach the IP address of the IP telephone. For example, both the home LAN and the VPN may use the IP address 192.168.0.0 with a subnet mask of 255.255.255.0.

Change either the network address and subnet mask of the home LAN or the corporate network so that subnets do not overlap.

For more information, see the following knowledgebase article:

<https://my.inin.com/products/pages/kb-details.aspx?entryid=q139967215602844>

## Poor audio quality

If you experience poor audio quality when using Interaction SIP Bridge, consider the following possible causes:

---

### Bandwidth limitations

- Ensure that the upload speed of the Internet connection is sufficient.  
Most home-based Internet connections have an upload bandwidth limit. In some situations, this limit may be as restrictive in comparison to the download bandwidth. If you generate a large amount of network traffic over your Internet connection, this traffic can impede real-time communications.
  - Ensure that an appropriate codec is being used.  
The G.711 codec provides an 8 kHz sampling frequency and uses 64 kbps of bandwidth. Other codecs, such as G.726 (32 kbps), G.729 (8 kbps), and G.723.1 (6.3 kbps) use much less bandwidth at the expense of smaller sampling frequencies. If you are having audio quality issues for a specific codec, selecting a codec that uses less bandwidth may be a solution.
  - Ensure that the LAN containing the remote computer with Interaction SIP Bridge has sufficient bandwidth available. Other systems on the LAN could transfer large amounts of data that could cause delays at the switch or wireless access point.
- 

### Quality of Service (QoS)

- Quality of Service (QoS) must be enabled for the IP telephone, Interaction SIP Bridge (enabled by default), the SIP line in Interaction Administrator, and Interaction Media Server.
  - Ensure that the RTP QoS DSCP markings are the same for the IP telephone, Interaction SIP Bridge, and Interaction Media Server
  - Ensure that the SIP QoS DSCP markings are the same for the IP telephone, Interaction SIP Bridge, and the SIP line in Interaction Administrator.
- 

### High network latency

Latency measures the amount of time for an IP packet to reach a network node or endpoint and for the original transmitting device to receive an acknowledgement. Latency values above 150 milliseconds can result in noticeable audio problems in real-time communications. Latency values above 300 milliseconds result in severe audio problems.

Use the Windows `tracert target_name` command to view latency values for each node in a network connection to a specific endpoint.

---

### Dropped packets

If IP packets containing real-time communications are being dropped somewhere along the network connection, those communications are absent when the receiving endpoint reassembles the data from the IP packets. If you are using UDP as the transport layer protocol in real-time communications, use TCP instead.

---

### Insufficient CPU resources on the computer

Ensure that the remote computer on which Interaction SIP Bridge is functioning has sufficient CPU resources to process real-time communications. Also consider the following items:

- For computers with multiple core CPUs, network interface card (NIC) drivers that do not support Receive Side Scaling (RSS) process all network communications through a single CPU core. If the computer is processing a large amount of network communications and other CPU-intensive tasks, the single CPU core could reach 100% usage, which would then limit the processing speed of network communications.
  - Use Windows Task Manager to ensure that no active CPU-intensive processes are set with a higher priority than the `InteractionSIPBridge.exe` process, which has a default priority of **Above Normal**.
- 

### Calls fail after first attempt but succeed on second attempt

This problem can be caused by the timeout configuration of the dynamic Address Resolution Protocol (ARP) feature in your Ethernet switch. If the ARP timeout value for the port to which the IP telephone is connected on the switch expires, the first call will return a *fast busy* audio signal. To correct this problem, configure the ARP timeout on the port for a longer period of time.

---

## License failure

If you have previously connected Interaction SIP Bridge to the Customer Interaction Center server, a change in the IP address of the Customer Interaction Center server results in a failure of license verification. This failure occurs because the computer that hosts Interaction SIP Bridge retains DNS data in its local cache.

To remove the cached DNS data, do the following steps:

1. On the Windows computer that hosts Interaction SIP Bridge, open a command prompt window as an administrator.
2. Enter the following command and press the **Enter** key.

```
ipconfig /flushdns
```

## IP telephone does not display correct time and date

Interaction SIP Bridge does not support domain-based time synchronization for the IP telephone. As a result, the time and date displayed on your IP telephone may be incorrect.

**Note:**

This procedure changes the SNTP server for all devices in the location.

To solve this problem, do the following steps:

1. Start Interaction Administrator.
2. In the left pane of the **Interaction Administrator** window, select the **Locations** object in the **Regionalization** container.
3. In the right pane, double click the location entry that the managed IP telephone uses.  
The **Locations Configuration** dialog box is displayed.
4. In the **Locations Configuration** dialog box, select the **SNTP Server: Other** option.
5. In the **Other** box, enter the URL address of an Internet-based NTP server.
6. Select the **OK** button to close the **Locations Configuration** dialog box.

## IP telephone is not receiving firmware updates

Ensure that the IP telephone is not statically configured to contact a provisioning system and is configured to use DHCP (option 160 is the default) to discover the provision proxy hosted by Interaction SIP Bridge.

# Change log

Date	Changes
07-September-2012	Initial Release
06-August-2013	Removed reference to obsolete product
02-October-2013	Added admonishments for external registrations and shared stations appearances
18-August-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, references to Interactive Intelligence Product Information site URLs, and copyright and trademark information.
22-January-2015	<ul style="list-style-type: none"> <li>• Added an admonishment to disconnect the active connection to a CIC server if you want to add a second CIC server for registration</li> <li>• Added the "Interaction SIP Bridge registers but the IP telephone does not" article to the "Troubleshooting" section</li> </ul>
13-April-2015	<ul style="list-style-type: none"> <li>• Updated "Copyright and Trademark Information" page</li> <li>• Modified document formatting for corporate information standards</li> </ul>
23-October-2015	<ul style="list-style-type: none"> <li>• Updated "Copyright and Trademark Information" page</li> <li>• Updated corporate branding in all affected images</li> </ul>
04-December-2015	Added "Restrictions" section to "Interaction SIP Bridge requirements".
15-February-2016	<ul style="list-style-type: none"> <li>• Updated diagrams to new corporate standards</li> <li>• Added diagram of example agent remote network configurations that are commonly used with Interaction SIP Bridge</li> <li>• Modified content on agent remote network configuration for clarification</li> </ul>
01-July-2016	<ul style="list-style-type: none"> <li>• IC-136460 - SIP Bridge does not support SRV registrations</li> <li>• Removed content stating support of DNS SRV registration groups for Interaction SIP Bridge</li> <li>• Updated "Copyright and trademark information" page</li> <li>• Minor formatting changes</li> </ul>
28-September-2016	Edited <a href="#">Emergency telephone numbers</a> for the E911 feature of CIC 2017 R1.
30-January-2018	Conversion to HTML.
08-February-2018	Rebranded images and terminology.