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GENESYS™

Configuration of CIC Phone Features for Polycom Phones

Technical Reference

Abstract

This document applies to non-managed phones. It describes the manual configuration of Polycom IP phones where an administrator changes the Polycom phone configuration files. It details the configuration of Shared Line Appearances (SLAs), Zone Paging, Call Park, and Group Call Pickup. These features require configuration settings in Interaction Administrator and Polycom configuration files.

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

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Overview

This document addresses the manual configuration of Customer Interaction Center (CIC) phone features for Polycom phones. It includes configuration and setup instructions for the following features:

- Shared Line Appearances
- Zone Paging
- Call Park
- Group Call Pickup

CIC client

Customer Interaction Center (CIC) supports two interaction management client applications. This documentation uses the term CIC client to refer to either Interaction Connect or Interaction Desktop.

SIP Auto Provisioning

CIC includes the SIP Auto Provisioning feature which enables you to configure supported Polycom IP phones or SIP Soft Phones automatically. Interaction Administrator includes the Managed IP Phones container and other enhancements that support auto-provisioning.

Managed Phones vs. Non-Managed Phones

The managed phone approach is the preferred method for configuring, setting up, and managing Polycom IP phones or SIP Soft Phones. You can use auto-provisioning and still have the flexibility to create custom configuration files as needed. For more information, see *IC Managed IP Phones Administrator's Guide*.

This technical reference addresses the manual configuration of Polycom IP phones which occurs when an administrator directly modifies the Polycom phone configuration files. It details configuration changes required in the Polycom configuration files to support CIC phone features. It also describes the necessary Interaction Administrator configuration settings for these non-managed phones.

Adopt one approach or the other for the configuration and setup of your IP phones. For the Shared Line Appearances feature, you cannot mix managed and non-managed phone types.

The Call Park and Zone Page features are automatically available for managed phones and require no additional changes to the individual phone configuration files. However, it is also necessary to configure the appropriate station groups in Interaction Administrator to support the Call Park and Zone Page features. See *Interaction Administrator Help* for details on station group configuration. See also Interaction Administrator configuration settings for Zone Page.

Requirements

Polycom firmware requirements for each model are listed on the **SIP IP Phones** information page available at the PureConnect Testlab website, http://testlab.genesys.com. Also, consult the Polycom documentation for instructions on determining the firmware version number for a particular phone.

Polycom Configuration Files

IC phone features require changes to the Polycom configuration files. The SIP configuration file (`sip.cfg`) contains SIP protocol and core configuration settings that typically affect an entire installation. This file is supplied in the Polycom firmware.

The IP Phone Configurator creates the `xIC.cfg` file which overrides attribute values in `sip.cfg`. We recommend that you do not modify the `sip.cfg` file. If you store all site-wide configuration values in the `xIC.cfg` file, and then you can install Polycom firmware upgrades without replacing previous configuration changes.

IP Phone Configurator

IP Phone Configurator is a tool that speeds up the initial configuration of large numbers of Polycom phones. See **Download the IP Phone Configurator utility** under **Polycom Related Files** on the **SIP-Related Download Files** page on the PureConnect Product & Support Information website: https://my.inin.com/products/sip-platforms/Pages/SIP-Related-Download-Files.aspx.

IP Phone Configurator creates these files:

`xIC.cfg`

> This site-wide settings file provides overrides to attributes in the `sip.cfg`.
>
> **Note:**
> If you are not in the United States, or any other country that uses the same DST scheme, also adjust some site-specific settings in `xIC.cfg`, such as GMT offset and daylight savings time attributes.

<group name>`.cfg`

> This optional file provides overrides to attributes in the individual phone configuration files, `phone name-` *< Ethernet address>* `.cfg`.

<Ethernet address>`.cfg`

> This file is unique for each phone. This bootstrap file tells the phone which configuration files to load.
>
> **Note:**
> The *Ethernet address* part of the file name is the **MAC address** of the phone. *< Ethernet address>*`.cfg` is the term used in the Polycom documentation.

phone name-<Ethernet address>`.cfg`

> This file is unique for each phone. It contains phone configuration settings for one phone.
>
> The *phone name* part of this file name is the individual phone name you specify in this tool. If you import data to Interaction Administrator, it is also the station name for the phone.

<any name>`.csv`

> This phone configuration data export file can be used to import station data to Interaction Administrator.

<any name>`.ipc`

> Use this IP Phone Configurator data file to save and retrieve phone and phone group data. It can be used to organize phone configuration data by site, company, vendor, or other identifier.

Manual configuration

Polycom configuration files are in XML format. After you create these files, it is easier to modify them if you use an XML editor.

> **Note:**
> We recommend that you do not modify the `sip.cfg` file.

Interaction Administrator Configuration Settings

In addition to changes to Polycom configuration files, some of the CIC phone features require certain configuration settings in Interaction Administrator.

This document contains examples of the appropriate settings in Interaction Administrator. See *Interaction Administrator Help* for additional details.

SNTP Server

The CIC installation process sets the SNTP (Simple Network Time Protocol) service to run on the CIC server. Depending on where CIC is installed, additional SNTP configuration can be required.

> **Note:**
> You can also configure SNTP settings by means of DHCP. For more information, see "Configure the time server (Polycom and IIS)" in *CIC Managed IP Phones Administrator's Guide*. These settings are valid for both managed and non-managed phones.

- If CIC is *installed on a domain controller* Windows Server 2008, no additional SNTP service configuration is necessary.
- If CIC is *installed on a non-domain controller* Windows Server 2008, it is possible the NTPServer is not enabled. The system is an **NTP/SNTP client**, but not a server. Two additional configuration steps are necessary for the SNTP service.
  1. Change a registry entry on the CIC server:
     a. Locate `HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\W32Time\TimeProviders\NtpServer` and set the value of **Enabled** to **1**.
     b. Open a command window, and type:
        `net stop w32time && net start w32time`
        > **Note:**
        > The NTP/SNTP server-side functionality is not active on non-domain controllers by default.
  2. Supply the address of the SNTP server in the `xIC.cfg` file.
- If CIC is *installed on a non-domain controller* Windows Server 2008 system in a domain without a reliable NTP source, Polycom phones do not use this NTP server. A reliable NTP source can be external or direct. Two additional configuration steps are necessary for the SNTP service.
  1. Make two changes in registry entries on the CIC server:
     a. Locate `HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\W32Time\Config` and set the value of **AnnounceFlags** to **5**.
     b. Locate `HKEY_LOCAL_MACHINE\SYSTEM\CurrentControlSet\Services\W32Time\TimeProviders\NtpServer` and set the value of **Enabled** to **1**.
     c. Open a command window, and type:
        `net stop w32time && net start w32time`
     > **Note:**
     > These registry changes makes the time services a trusted authority using the local clock. A system that uses an external NTP server is preferred, but the default configuration is to try to use an external NTP server. If that is not working, these parameters treat the computer clock as a trusted source for computers connecting to this SNTP server.
  2. Supply the address of the SNTP server in the `xIC.cfg` file.

DHCP Server

Polycom phones also support dynamic host configuration protocol (DHCP). When set up, DHCP permits Plug and Play TCP/IP network setup.

Basic network settings can be derived from DHCP or typed manually using the LCD-based user interface on the phone. Polycom recommends using DHCP where possible to eliminate repetitive manual data entry. You can change phone network configuration settings by means of a main menu and two submenus: **DHCP Menu** and **Server Menu**.

Parameters obtained from a DHCP server can override manually typed networking parameters. These parameters include the TFTP server (boot server) address and the SNTP server address. For more information, see *Polycom® Administrator Guide* for a list.

TFTP Server

CIC provides a built-in TFTP server. When a Polycom phone is switched on, it downloads its configuration files from the TFTP root directory. By default, Polycom phones also write log files to this TFTP directory (if the TFTP server is configured to accept write operations).

> **Note:**
> The TFTP server does not directly facilitate the creation or modification of Polycom phone configuration files. The TFTP root directory is available only by means of the network (if you set it up as a shared folder) or by direct access to the CIC server. The TFTP server is not used for managed phones.

TFTP Server Subdirectories

By default, the IP phone files are stored in the main TFTP Server directory. Or you can use separate subdirectories on the TFTP Server for phone logs, contact directories, and override files. Specify these subdirectory names in the Polycom master configuration file (`<Ethernet address>.cfg`) to make the phones write the files to the appropriate subdirectories.

These types of files can be stored in subdirectories on the TFTP Server:

Phone logs

Both startup and application event log files are maintained for each phone. These event log files are stored in the flash file system on the phone and are periodically uploaded to the TFTP server. The event log files are `<Ethernet address>-boot.log` and `<Ethernet address>-app.log`.

Contact directories

Each phone maintains a local contact directory. The person using the phone can change the directory contents at will. Changes are stored in the flash file system on the phone and are backed up to the boot server copy of `<Ethernet address>-directory.xml` (if configured). Only updated contact directories are stored in this subdirectory. When the phone starts up, the boot server copy of the directory, if present, replaces the local copy.

Overrides

The `<Ethernet address>-phone.cfg` files are also referred to as boot server override files. Configuration overrides made at the phone are written to this subdirectory. Storing the updated phone configuration files in a specific subdirectory helps to separate them from the files you often modify which are stored in the root directory.

TFTP Server Interface

You can use a web-based interface for the CIC TFTP server to:

- Monitor the TFTP client file transfer activity.
- Specify upload file types.
- Configure the upload subdirectories.

View the TFTP Server Configuration

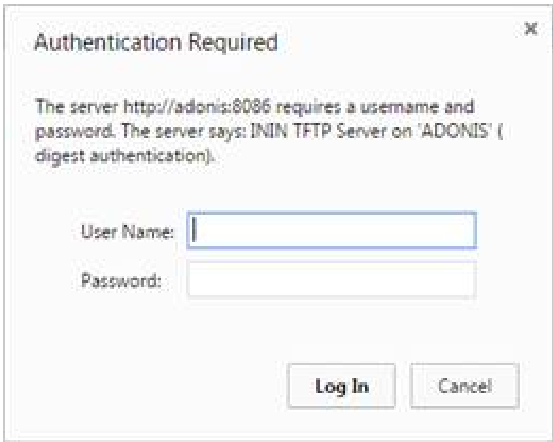
TFTP server configuration is performed in its web-based configuration interface.

To display the TFTP server browser-based configuration pages:

1. Point your browser to the TFTP server and its port number.
   Example:
   http://spyhunter:8086

   > **Note:**
   > The default TFTP server port number is 8086, but this setting can be changed in the registry. See `HKEY_LOCAL_MACHINE\SOFTWARE\Interactive Intelligence\TFTPServer\Config Port`.

2. In the **Enter Network Password** dialog box, type the TFTP server **User Name** and **Password** and click **Log In**.

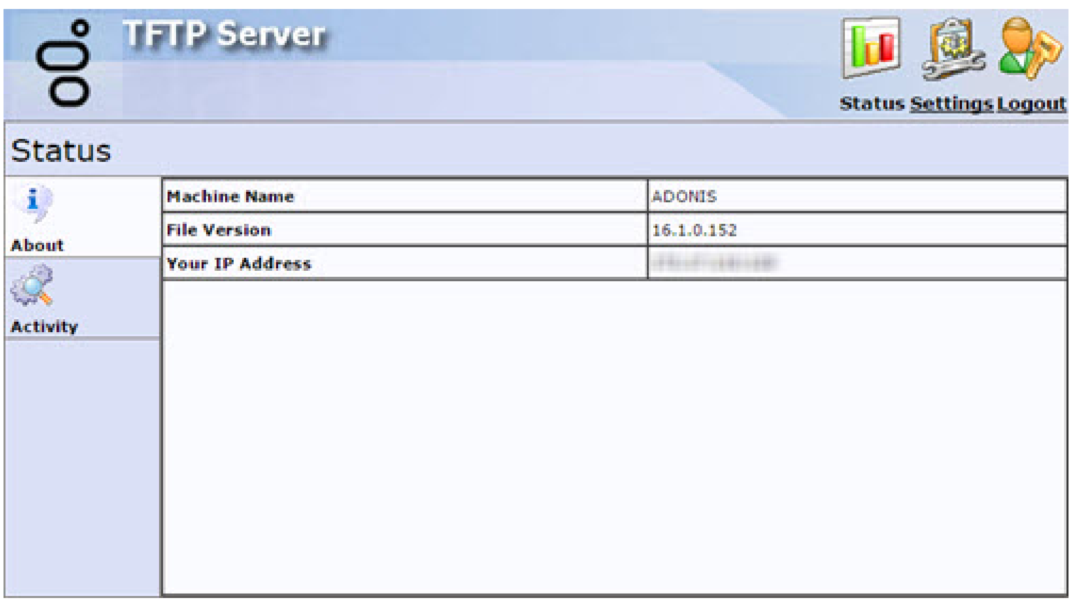


> **Note:**
> The TFTP server User Name and Password are defined in the initial installation. Unless otherwise specified, the default user name is `admin` (all lowercase) and the default password is `1234`.

> **Tip:**
> See Changing the TFTP Server Login Credentials for information on changing this user name and password.

Result:
The **TFTP Server About** configuration page appears. It contains basic TFTP server information.

> **Note:**
> You can also display this page at any time by selecting **Status** from the menu bar and clicking the **About** icon on the left side of the **Status** page.



Machine Name

The TFTP server *computer name*.

File Version

PureConnect Customer Care representatives sometimes ask for this TFTP server files version number.

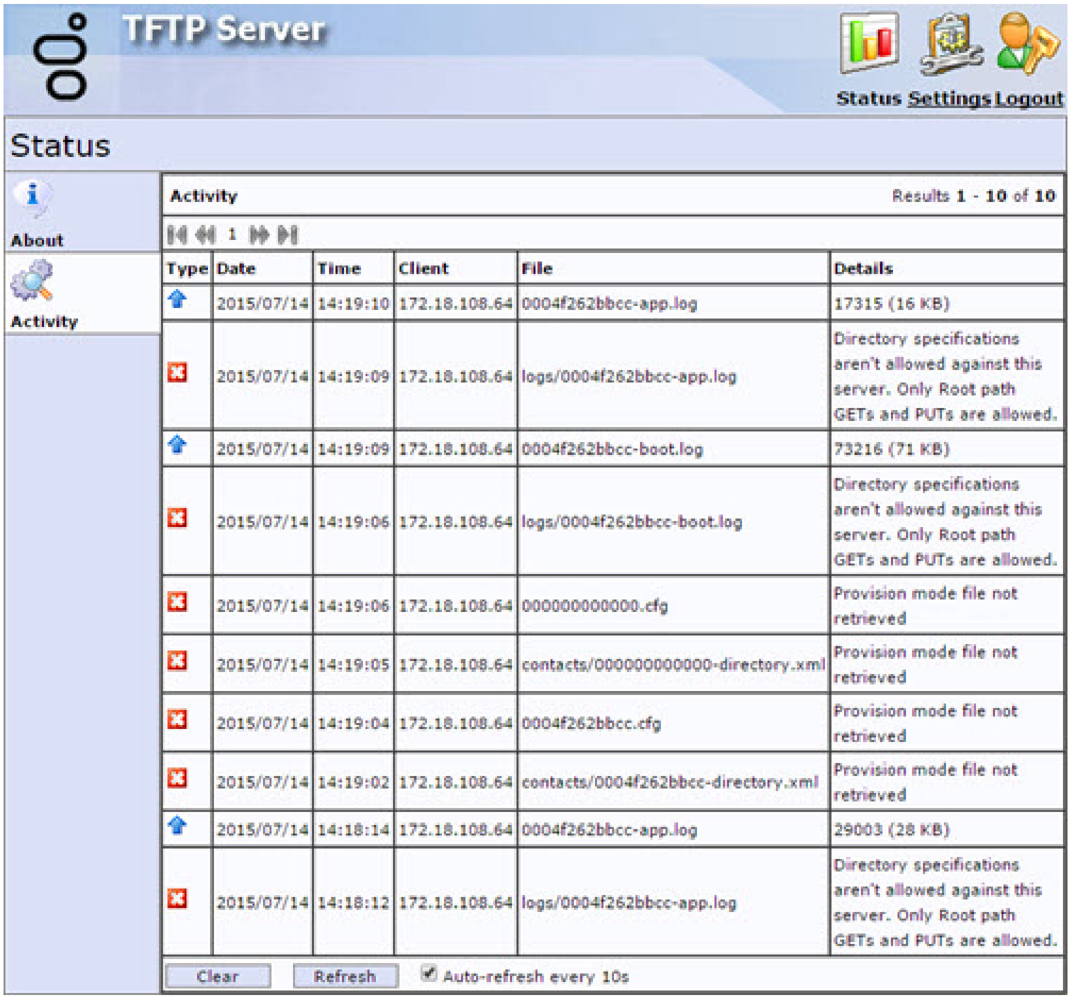
Your IP Address

The IP address of the computer where you are running the browser.

Monitoring TFTP Server Activities

You can monitor and review the client file transfer activities on the TFTP server. Select **Status** from the menu bar and click the **Activity** icon on the left side of the **Status** page.

Result:
The **TFTP Server Activity** page appears.



The following columns appear in the **Activity** view:

Type

These icons identify the direction of the file transfer, a file transfer failure, or a message.

	File successfully downloaded from phone to TFTP server.
	File successfully uploaded from TFTP server to phone.
	File transfer failure.
	IP Address is not in the allowed range.
	Phone has returned an error during the file transfer. The message from the phone appears in the **Details** column.

Date and **Time**

Date and time of the file transfer activity. Activities are presented in reverse chronological order, with the most recent activity first on the list.

Client

The IP address for the telephone.

File

The name of the file that was transferred.

Details

This column displays information related to the activity. It displays the transfer size for uploads and downloads. If a File Transfer Error occurs, it displays the message.

> **Note:**
> Transfer size is not always the same as file size. If a phone knows that it does not need the entire file, it is common for it to quit downloading a large file. For example, the phone can determine the file version from the header (first set of bytes) of `sip.id` and if it has the current version, it stops the download process.

Use these controls to work with the Activity view:

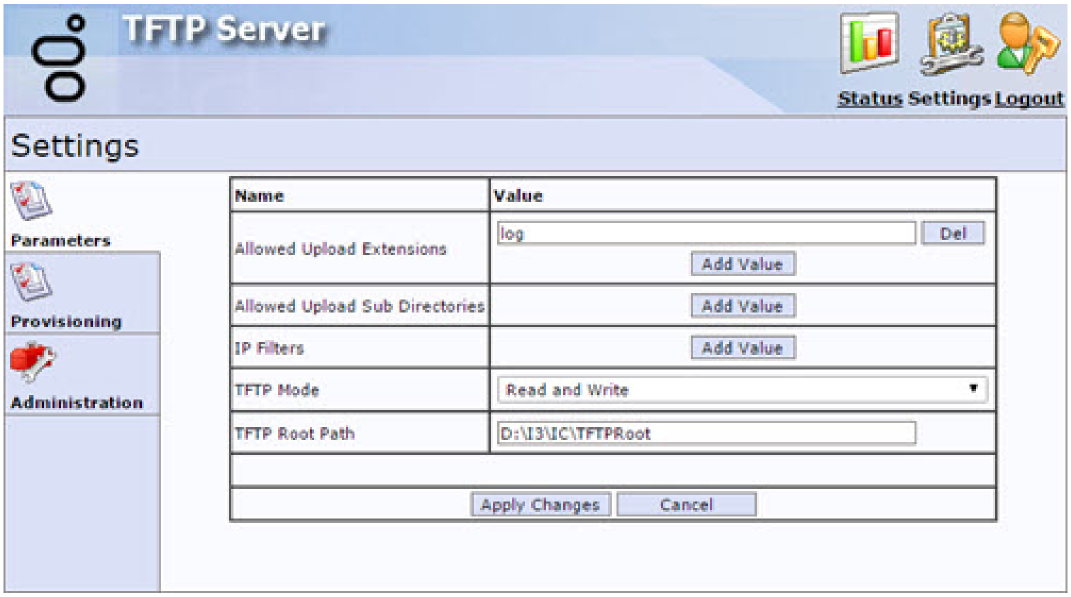
	Display first page of **Activity** view.
	Display previous page of **Activity** view.
1 **2**	Click a page hyperlink to explore the **Activity** view page by page.
	Display next page of **Activity** view.
	Display last page of **Activity** view.
Clear	Clear all activity history from the **Activity** view.
Refresh	Refresh the **Activity** view with the latest file transfer information.
Auto-refresh every 10s	Select this check box to refresh the **Activity** view automatically every 10 seconds. **Note:** This option is available only on the first page. It updates the display so you always see the 15 most recent file transfer activities. **Tip:** You can control the number of items displayed in each page by adding `size=X` to the URL string, for example `http://spyhunter:8086/status/activity?size=30`). You can use **Auto-refresh** to see *X* number of the most recent file transfer activities.

Configuring the TFTP Server

You can configure the TFTP server from the web interface.

1. Select **Settings** from the menu bar and click the **Parameters** icon on the left side of the **Settings** page.

   Result:
   The **TFTP Server Parameters** page appears.



2. In **Allowed Upload Extensions**, do one of the following:
   - Click **Add Value** to specify an additional allowed extension for files that can be uploaded to the TFTP server.
   - Click **Del** on the appropriate line to delete an allowed extension.

   Default value: log.

> **Note:**
> Specify an allowed extension without any prefix or period. For example, specify `*.log` as `log` to allow only `*.log` files to be uploaded. If **Allowed Upload Extensions** values are not set or are blank, the TFTP server accepts any file name when the server is configured for write or read/write mode.

3. In **Allowed Upload Sub Directories**, do one of the following:
   - Click **Add Value** to specify the name of a subdirectory in the TFTP server root directory that can accept files uploaded from the phones.
   - Click **Del** on the appropriate line to delete the name of an allowed subdirectory.

   Default value: blank

> **Note:**
> Directory names can contain only upper or lowercase letters, numbers, and underscores. Spaces and punctuation are not allowed.

> **Warning:**
> Physically create the appropriate subdirectories on the TFTP server and identify them here. Phone logs, configuration files, or contact directories cannot be saved until the subdirectories exist. Allowed subdirectory names must match the ones used in the *`<Ethernet address>`*`.cfg` files for each phones. See [Modifying the Phone Configuration Files].

4. In **IP Filters**, do one of the following:
   - Click **Add Value** to specify a range of IP addresses which have access to the TFTP server. Type the address range in the format `X.X.X.X-X.X.X.X` in the two boxes provided.
   - Click **Del** on the appropriate line to delete an allowed IP address range.

   Default values: blank

> **Note:**
> Addresses outside the ranges provided are not allowed access. You can specify multiple address ranges.

> **Warning:**
> If no values are specified, access is not limited.

5. In **TFTP Mode**, select the mode the TFTP server runs in. The possible settings are:

Read	The TFTP server sends files, but does not accept TFTP uploads. This setting is the most secure mode.
Read and Write	The TFTP server allows upload and download of files.
Write	The server accepts TFTP uploads, but does not allow phones to download files. This mode would almost never be used for supporting IP phones.

6. In TFTP Root Path, specify the directory the TFTP server uses as its home or root directory.
   Default value: `C:\I3\IC\TFTPRoot`
   The CIC install creates this registry value and sets it to *[i3 install disk][i3 install directory]*`\TFTPRoot` (example: `D:\i3\ic\TFTPRoot`) and creates the directory automatically.
7. Do one of the following:
   - Click the **Apply Changes** button to save changes to the TFTP server configuration.

     > **Note:**
     > These changes are effective immediately.

   - Click the **Cancel** button to discard changes made to the TFTP server configuration.

Modifying the Phone Configuration Files

To direct the phone logs, contact directories, and overrides files to the appropriate TFTP server subdirectories, also modify each *<Ethernet address>*`.cfg` file.

> **Tip:**
> You can specify the name of the log directory in the Site configuration settings in the CIC IP Phone Configurator. This log directory name defaults to all the *<Ethernet address>*`.cfg` files you create with this tool. The log file directory name specified in the **TFTP Server Parameters** must match the `LOG_FILE_DIRECTORY` value in the *<Ethernet address>*`.cfg` for the Polycom phones.

Using an XML editor, supply the appropriate directory names in these attributes:

- `LOG_FILE_DIRECTORY`
- `OVERRIDES_DIRECTORY`
- `CONTACTS_DIRECTORY`

Changing the TFTP Server Login Credentials

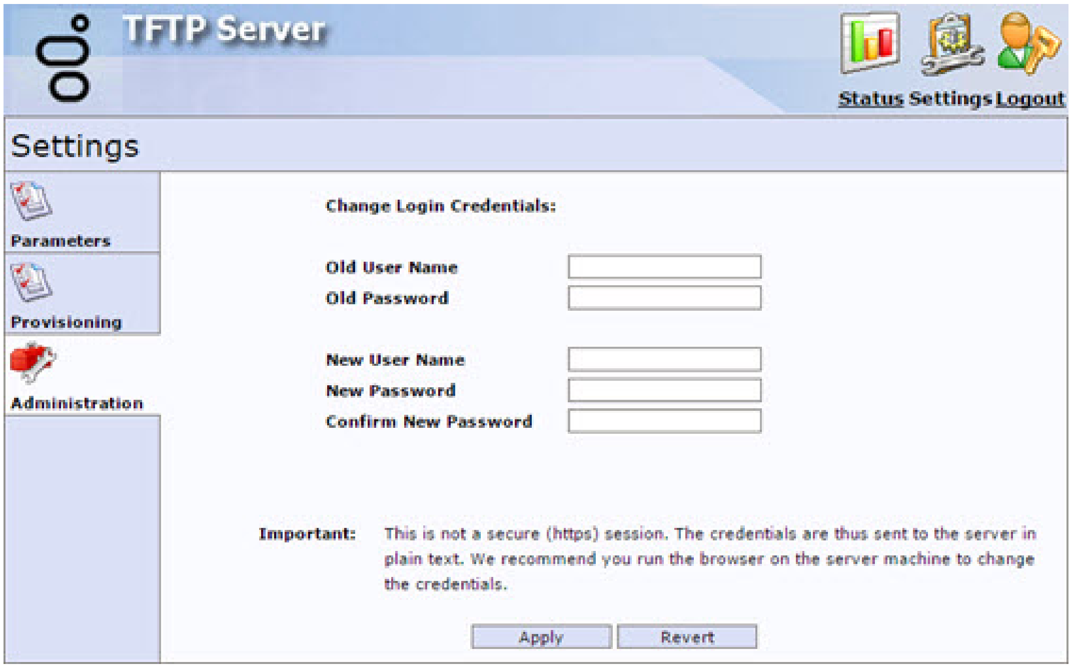
You can change the TFTP server user name and password at any time.

1. Log on to the TFTP server configuration interface by running a browser on the TFTP server computer.

   > **Note:**
   > You can log on to the TFTP server from another computer, but we recommend that you run the browser on the server computer when changing the logon credentials. See Configuring the TFTP Server for instructions for displaying the TFTP Server configuration pages.

2. Select **Settings** from the menu bar and click the Administration icon on the left side of the **Settings** page.

   Result:
   The **Change Login Credentials** page appears.



3. In the **Old User Name** and **Old Password** boxes, type the user name and password for the user currently logged on to the TFTP Web Configuration pages.

   > **Tip:**
   > The CIC server install sets the default user name to `admin` (all lowercase) and the default password to `1234`.

4. In the **New User Name** and **New Password** boxes, type a new user name and password.
5. In the **Confirm New Password** box, retype the new password.
6. Do one of the following:
   - Click **Apply** to save the new user name and password.
   - Click **Revert** to discard user name and password changes.

Forgot Your User Name and Password?

If you forget the TFTP server configuration logon user name and password, you can reset them to the default values of `admin` and `1234` by deleting the following registry entries. And then you can use the default values to log on and reset the logon credentials.

```
HKEY_LOCAL_MACHINE\SOFTWARE\Interactive Intelligence\TFTPServer\Config Username
```

```
HKEY_LOCAL_MACHINE\SOFTWARE\Interactive Intelligence\TFTPServer\Config Password
```

Installation and basic configuration

The CIC server install automatically installs the necessary TFTP server component files. You can use the Setup Assistant to change the TFTP server **Service Startup Type** from **Manual** to **Automatic**. See *IC Installation and Configuration Guide* for additional details.

Note:
If you change the TFTP server **Service Startup Type**, restart the service.

Shared Line Appearances

Users can employ Shared Line Appearances (SLAs) to use their phones to manage calls for other users. They can answer and make calls as if they were using an CIC station belonging to another user or using a group extension.

An SLA is associated with a line key on the Polycom phone. Users can have both shared and private lines associated with different line keys on the same phone.

> **Note:**
> Ensure all the phones that use the Shared Line Appearances feature are either managed or non-managed. You cannot mix the two types.

Terminology

Some of the terms used in this document for configuration and setup of SLAs include:

Line Appearance

An appearance for a telephone line, a single-user circuit on a telephone system. A phone can have more than one line appearance for the same line or shared line.

Shared Line Appearance

A line appearance that appears on multiple phones.

Call Appearance

An active call on a line appearance or shared line appearance

Multiple Call Appearances

More than one call on one line appearance or on a shared line appearance. Only one of the calls can be active at one time; the other calls on the same line appearance are on hold.

Primary Station

The station to which the line appearance is assigned.

Secondary Station

The station on which an SLA appears. Secondary stations must have their own private (non-shared) line in addition to the SLA.

Primary User

The user whose station is the **Primary Station**.

Secondary User

The user whose station is the **Secondary Station**.

Operation

An SLA occurs when a single CIC station (or *line*) appears on multiple phones. There is a single logical entity for the shared *line*, even though it can appear on several distinct physical devices (phones). A device using the shared line appearance does not have a unique identity or addressability.

SLAs are similar to extension phones in your home. When a call comes in, all of the extensions ring and you can answer the call from any extension. When you call from an extension, nobody else can call on the same line until you are finished. However, there is one important difference. The line belongs to a specific station in IC. This station is the *primary* and all the other stations on which this line appears are *secondary*. This distinction becomes important when CIC user status for the user associated with the primary station is set to **Available**, **Forward**, or one of the *not available* statuses.

Here are some features of the user experience with shared line appearances:

- Shared line appearances make the same line available on more than one phone.
- Shared line appearances can be associated with more than one line key on a phone.
- A specific icon on the phone display indicates a Shared Line Appearance. This icon is typically a telephone that is half black and half gray.
- In most cases, incoming calls to a shared line appearance cause all of the phones with the shared line appearance to ring.

  > **Note:**
  > If the status for the CIC user of the primary station is one of the *not available* statuses, incoming calls do not ring on the primary station. Incoming calls ring on secondary stations when the primary user is not available. These calls ring regardless of the secondary user's CIC status unless the secondary user presses the **Do Not Disturb** button on the phone.

- If the CIC status for the primary station user for the SLA is set to **Available, Forward**, none of the phones with the SLA ring for alerting calls. The call is sent to the forwarded number for the primary station user. If the forwarded number does not answer the call, it goes to voice mail for the primary station user.
- Any user who is party to an SLA can answer an alerting call or a held call on an SLA. Only one user is connected to the call if multiple users attempt to pick up the call simultaneously.
- Any call made on an SLA is reported as if the primary station user called.
- If an SLA is in use on one phone, it cannot be used on another phone at the same time. If two users attempt to use an SLA simultaneously, the system permits one to succeed and indicates to the other that the line is in use.
- The phone display indicates when an SLA is alerting, off-hook, on hold, or connected.
- Calls on SLAs can be transferred, put on hold, or added to a conference.
- If no one answers an alerting call on an SLA, the call goes to voice mail for the primary station user.

Configuration

You configure SLAs in both the Polycom phone configuration files and in Interaction Administrator.

The following sections describe the parameters and values you must set for SLAs. The examples describe a situation where an assistant is able to answer and place calls on the line belonging to the manager through the use of shared line appearances. The assistant also has three private line appearances which are not shared with the phone for the manager.

These values are used in the following examples:

	Manager	Assistant
Station type	Primary	Secondary
Private line appearances	0	3
Shared line appearances	3	3
IA[1] Station Name	hal1	hal3
IA Station Extension	7001	7003
IA Nb.[2] Calls	3	3
IA Identification Addr.	70011[3]	

[1] IA stands for Interaction Administrator.

[2] Nb. is an abbreviation for number.

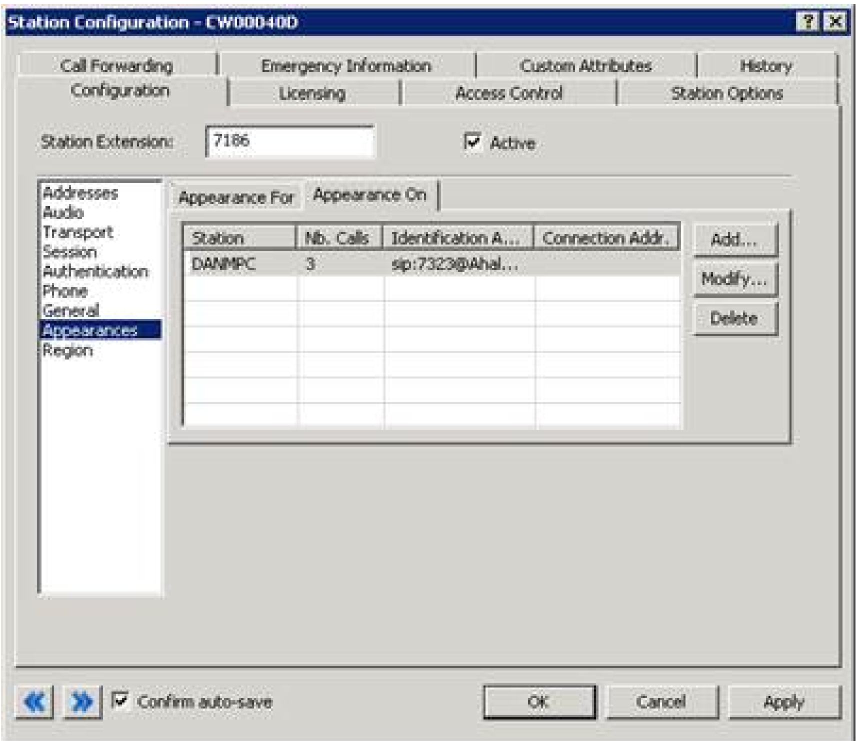
[3] This example uses 70011 as the IA Identification Address on the first secondary station. You could number any other secondary stations in sequence. For example, you could use 70012 for the next secondary station.

Interaction Administrator primary station configuration

When you configure the primary station, you *specify the secondary stations on which the primary station appears*. In this example, you list the station for the assistant in the **Appearance On** tab of the station configuration for the manager. This setting indicates that the station for the manager appears *on* the phone belong to the assistant as a shared line appearance.

Identification Addr.:
This address is the SIP ID number for the primary phone and a sequential number. In this example, it is `70011` (the SIP ID number for the primary station and `1`). For example, if multiple secondary stations monitor the same primary station, use `70011` for the first monitoring station and `70012` for the second monitoring station. Then number additional monitoring stations in sequence. This address must match the `reg.x.address` used for the shared line appearance in the Polycom phone configuration file for the secondary station. See Polycom configuration settings for SLAs topic.

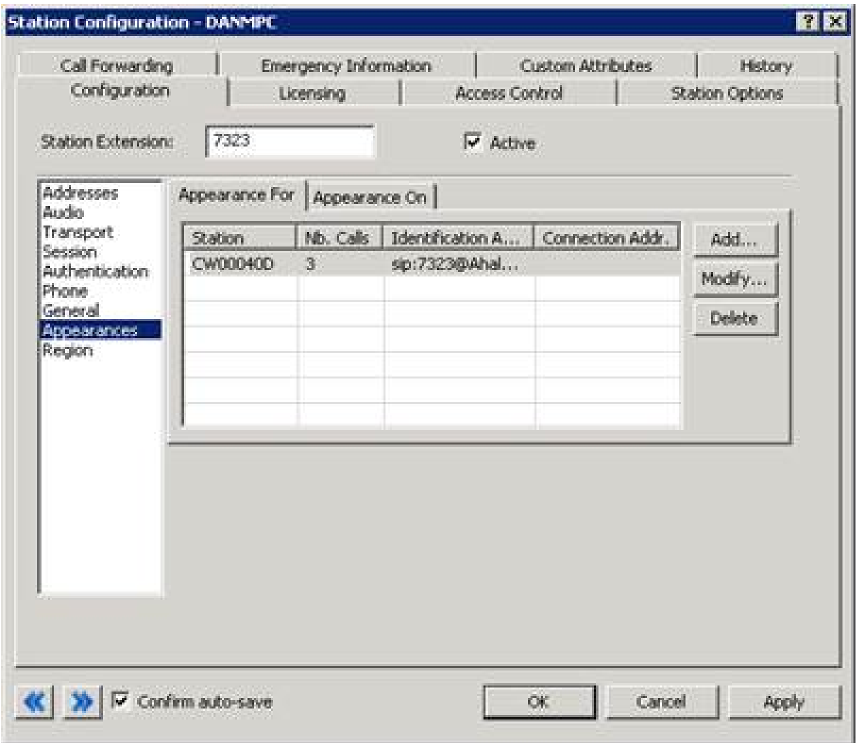
See the *Interaction Administrator Help* topic: "SIP Station Appearances" for additional information about configuring secondary line appearances.



Interaction Administrator secondary station configuration

When you configure the secondary station, you *specify the stations which appear as secondary line appearances on the secondary station*. In this example, you list the station for the manager in the **Appearance For** tab of the station configuration for the assistant. These secondary line appearances *appear for* other stations.

See *Interaction Administrator Help* topic: "SIP Station Appearances" for additional information about configuring secondary line appearances.



Polycom configuration settings for SLAs

All of the SLA settings are contained in the Polycom phone configuration files. Set the following attributes in the Registration (`<reg/>`) section of the *<Ethernet address>*`-phone.cfg` files for the stations involved in shared line appearances.

Attribute	Notes
`reg.x.address`	This address corresponds to this registration (*userPart* or *userPart*`@polycom.com`). On the primary station, this number matches the user portion of the **Identification Address of the SIP station** in Interaction Administrator. For a shared line appearance on a secondary station, this number matches the user portion of the **Identification Address of the Shared Line Appearance** configured in Interaction Administrator. In both cases, this number must be unique across all SIP stations and shared line appearances.
`reg.x.label`	This text label appears on the phone display next to the associated line key. It can be the same as `reg.x.address`. On secondary stations, set this label to the `reg.x.address` of the primary station. This setting makes the label on the secondary line appearance the same as the one on the primary station.
`reg.x.type`	Set this attribute to `private` for line appearances that appear only on this phone. Set this attribute to `shared` for shared line appearances.
`reg.x.thirdpartyname`	Set this attribute to a string in the same format as `reg.x.address`. On the primary station, this attribute matches the `reg.x.address` for the primary station. On the secondary station, this attribute also matches the `reg.x.address` for the primary station.
`reg.x.lineKeys`	This attribute defines the number of line appearances on this phone which are associated with this address (the number of line keys on the phone associated with this registration.) Set this attribute from `1` to the maximum number of line keys on the phone.
`reg.x.callsPerlineKey`	This attribute defines the number of concurrent calls or conferences that can be active or on hold for each line key on the phone. Set this attribute to blank on the individual phones. Use the **Call Appearances** setting in Interaction Administrator to specify this value. See Call Appearances.

Call Appearances

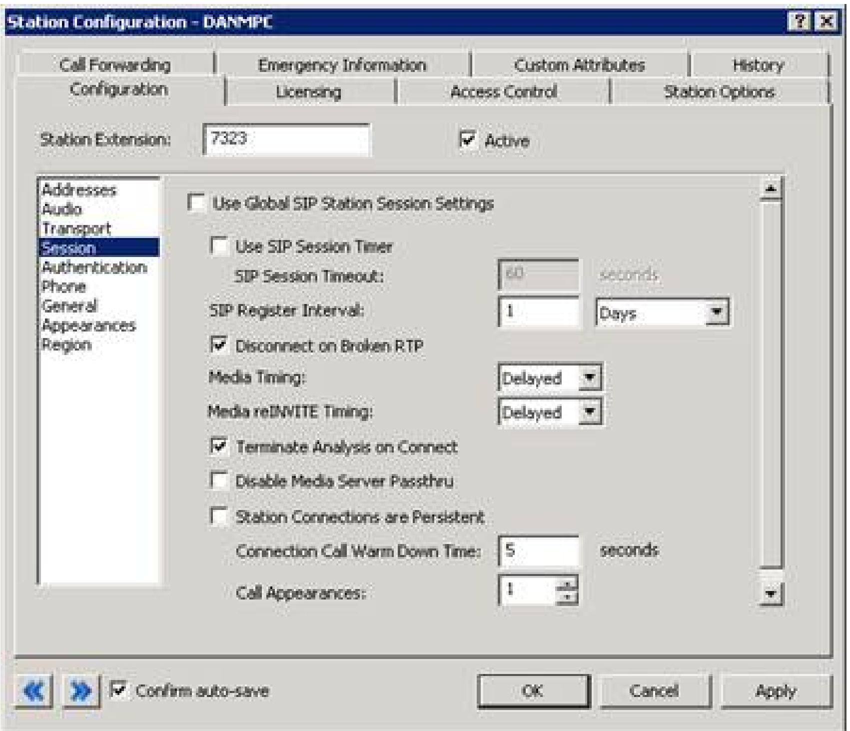
Polycom SoundPoint® IP phones support multiple calls for each line key. You can define the number of concurrent calls or conferences that can be active or on hold for each line key on the phone.

> **Note:**
> Users can employ the Hold feature to pause activity on one call and switch to another call on the same line. The maximum number of calls for each line key varies by phone model. Consult the most recent version of *Polycom Administrator Guide SoundPoint®/SoundStation® IP SIP* for details.

The following instructions describe how to set the number of call appearances for each line key for a particular phone.

To set the number of call appearances:

1. In Interaction Administrator, in the **Stations** container, select the station that corresponds to the phone you are configuring.
2. Right-click on the station name and select **Properties**.
   Result: The **Station Configuration** dialog box appears.
3. Select the **Configuration** tab and then select **Session**.
4. Clear the **Use Global SIP Station Session Settings** check box.

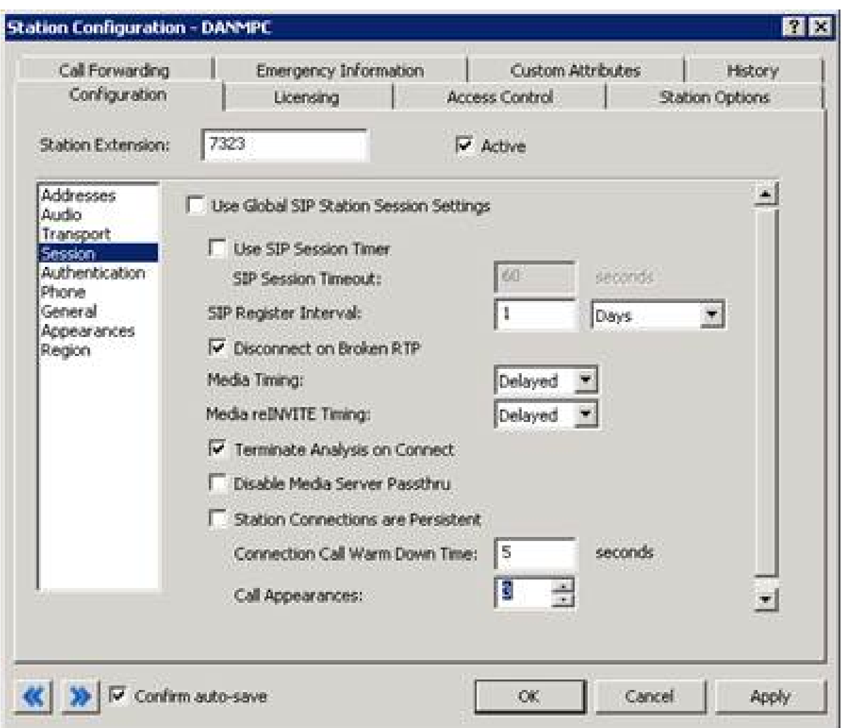


> **Note:**
> This selection prevents the phone from inheriting the SIP Session settings defined at the Station Space level. And then you can set unique values for this station.

5. Scroll down.

   In **Call Appearances**, set the number of appearances for each line key.

> **Note:**
> The maximum number of calls for each line key varies by phone model. Consult the most recent version of *Polycom Administrator Guide SoundPoint®/SoundStation® IP SIP* for details.



6. Click **Apply** to save the settings. Click **OK** to close the **Station Configuration** dialog box.

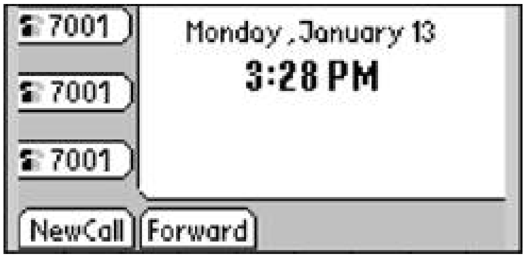
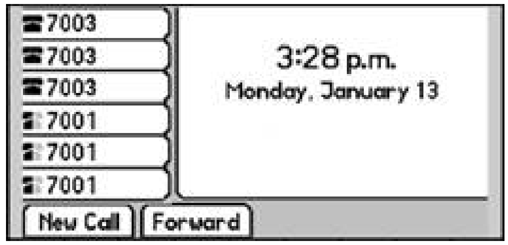
Polycom SLA settings examples

In the following examples, the primary station configuration has one shared line that appears on three line keys (three shared line appearances). Configuration of the secondary station describes three line appearances of the private line for the assistant in the `reg.1` attributes and three shared line appearances in the `reg.2` attributes.

Primary station	Secondary station
reg.1.displayName="7001" **reg.1.address="7001"** **reg.1.label="7001"** **reg.1.type="shared"** **reg.1.thirdPartyName="7001"** reg.1.auth.userId="" reg.1.auth.password="" reg.1.server.1.address="172.18.24.58" reg.1.server.1.port="5060" reg.1.server.1.transport="UDPonly" reg.1.server.1.expires="" reg.1.server.1.register="" reg.1.server.1.retryTimeOut="" reg.1.server.1.retryMaxCount="" reg.1.server.1.expires.lineSeize="" reg.1.acd-login-logout="0" reg.1.acd-agent-available="0" reg.1.ringType="6" **reg.1.lineKeys="3"** reg.1.callsPerLineKey=""	reg.1.displayName="7003" **reg.1.address="7003"** **reg.1.label="7003"** **reg.1.type="private"** **reg.1.thirdPartyName=""** reg.1.auth.userId="" reg.1.auth.password="" reg.1.server.1.address="172.18.24.58" reg.1.server.1.port="5060" reg.1.server.1.transport="UDPonly" reg.1.server.1.expires="" reg.1.server.1.register="" reg.1.server.1.retryTimeOut="" reg.1.server.1.retryMaxCount="" reg.1.server.1.expires.lineSeize="" reg.1.acd-login-logout="0" reg.1.acd-agent-available="0" reg.1.ringType="2" **reg.1.lineKeys="3"** reg.1.callsPerLineKey="1" reg.2.displayName="70011" **reg.2.address="70011"** **reg.2.label="7001"** **reg.2.type="shared"** **reg.2.thirdPartyName="7001"** reg.2.auth.userId="" reg.2.auth.password="" reg.2.server.1.address="172.18.24.58" reg.2.server.1.port="5060" reg.2.server.1.transport="UDPonly" reg.2.server.1.expires="" reg.2.server.1.register="" reg.2.server.1.retryTimeOut="" reg.2.server.1.retryMaxCount="" reg.2.server.1.expires.lineSeize="" reg.2.acd-login-logout="0" reg.2.acd-agent-available="0" reg.2.ringType="2" **reg.2.lineKeys="3"** reg.2.callsPerLineKey=""

Polycom display examples

This section presents examples of the Polycom phone displays that would be the result of the preceding settings examples. The appearances would vary depending on the phone model.

Primary station	Secondary station
	
This illustration shows the three shared lines as they would appear on a Polycom SoundPoint IP 500 SIP display.	This illustration shows the three private lines and three shared lines as they would appear on a Polycom SoundPoint IP 600 SIP display.

Call Park

Users can employ Call Park to place the current call on hold in a specific orbit. An orbit can hold one call. The user who puts a call in orbit assigns the orbit a number. And then any user can pick up that call from another station.

While on a connected call, a user can press the call park soft key and type an orbit number. If that orbit is vacant, the system holds the call and removes it from the queue for the user. To pick up the call, a user presses the call park soft key and presses the orbit number.

Using other IVR digit sequences, users can also play a list of all the calls in orbit or hear details about a selected call in a specific orbit.

> **Note:**
> The Call Park feature is automatically available for managed phones.

Interaction Administrator configuration settings for Call Park

No Interaction Administrator configuration settings are required for Call Park.

Polycom configuration settings for Call Park

Make *Call Park* and *Call Retrieval* available by setting the `enabled` attribute for the `call-park` feature to `1` in the **Feature** (`<feature/>`) section of the `xIC.cfg` file.

```
<feature
…
feature.11.name="call-park" feature.11.enabled="1"
```

> **Note:**
> In preceding example, Call Park is feature "11". Feature numbers vary by release. See *Polycom Administrator Guide SoundPoint®/SoundStation® IP SIP* for more details.

Group Call Pickup

Group Call Pickup allows a user to answer an alerting call on any phone in their station group. The user does not need to know the extension number of the ringing phone, but can press the Pickup and Group soft keys.

> **Note:**
> A phone can belong to more than one station group. If calls are ringing on multiple station groups, a user picks up the *oldest* station group call from a station belonging to multiple groups.

Interaction Administrator configuration settings for Group Call Pickup

Group call pickup applies to phones (stations) in the same station group. You can use existing station groups or create new ones for group call pickup. See *Interaction Administrator Help* for details on station group configuration.

Polycom configuration settings for Group Call Pickup

Make Group Call Pickup available by setting the `enabled` attribute for the `group-call-pickup` feature to `1` in the **Feature** (`<feature/>`) section of the `xIC.cfg` file.

```
<feature
…
feature.12.name="group-call-pickup" feature.12.enabled="1"
```

Zone Page

Make Zone page allows a user to make a live, one-way broadcast to a small, selected group of phones.

A user dials a short sequence of numbers (`*901` + zone number) to start a zone page. After hearing a beep, the user speaks into the handset to begin the page. The zone page recipients hear it through the speakers on their phones and do not have to pick up a handset.

> **Note:**
> The Zone Page feature is automatically available for managed phones. However, to use these extensions as zone numbers, you need to configure CIC station group extensions, workgroup extensions, user extensions, or station extensions in Interaction Administrator.

Other zone page features include:

- Zone numbers are CIC station group extensions, workgroup extensions, user extensions, or station extensions.
- A zone page cannot interrupt another zone page. If a phone is playing a page and someone attempts to page a group that includes that phone, the phone does not interrupt the first page for the second page.
- If an intended recipient is already using the phone at the start of a zone page, the zone page does not play on the speaker for that phone.

Limitations

Zone page is supported only on Polycom phones.

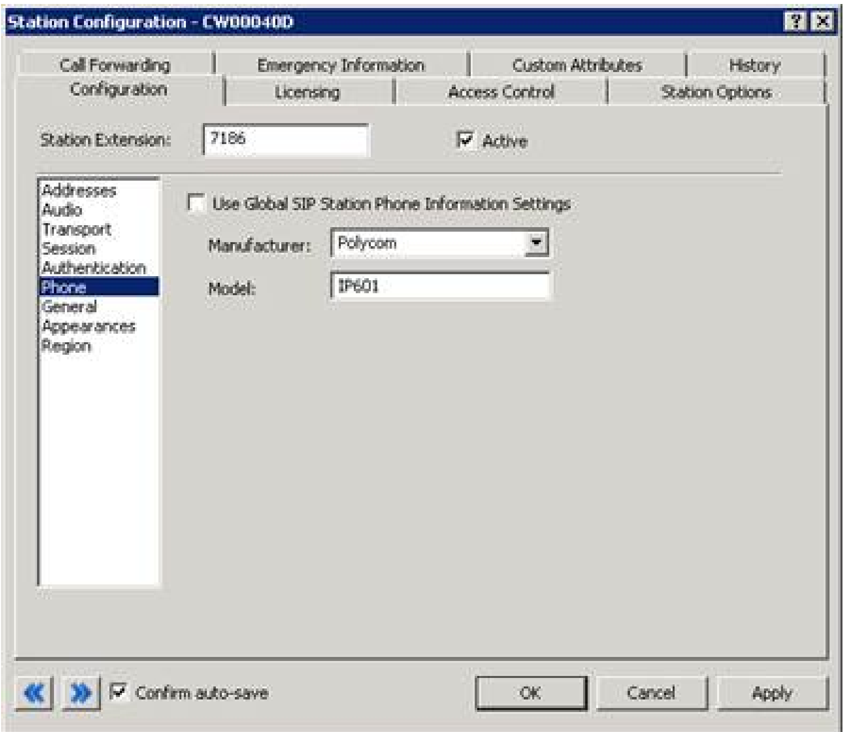
At sites with poor network performance, a significant delay can occur before some stations receive the page. In an environment where the listener can hear more than one phone at a time, a noticeable echo between the broadcasting phones can occur.

Zone paging causes the phones to go "off hook" before accepting the page. The more phones there are in a zone, the greater the delay before the page is broadcast. Zone paging is not intended for live paging to an entire organization. It is intended for paging to a small geographical zone in the office or to a small group.

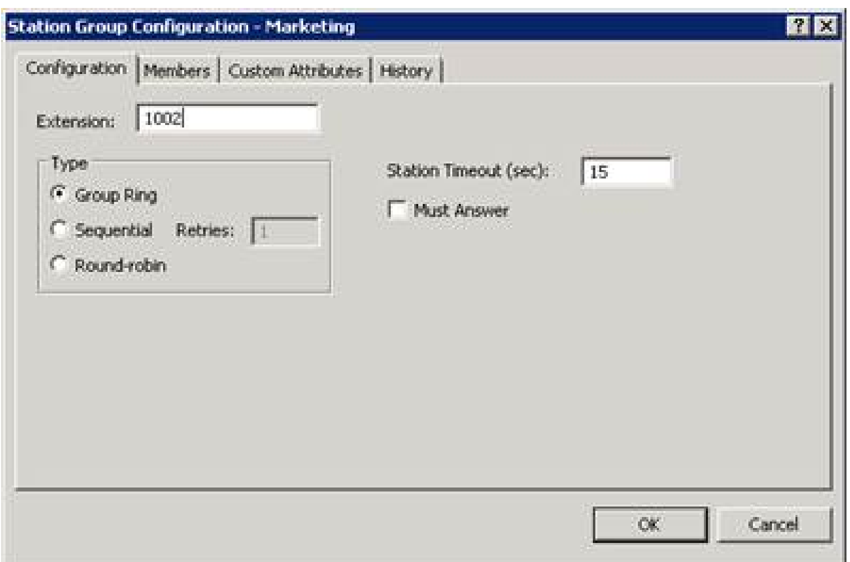
Interaction Administrator configuration settings for Zone Page

On the **Phone** page in the **Configuration** tab of **Station Configuration**, set **Manufacturer** to **Polycom** and **Model** to the appropriate Polycom model number. This configuration makes auto answer work with zone page and dialing through the CIC client.

Setting **Manufacturer** to **Polycom** instructs CIC to include an `Alert-Info` field in the SIP header. The value of `Alert-Info` is set to `http://localhost/AutoAnswer`.



Because zone numbers are CIC station group extensions, workgroup extensions, user extensions, or station extensions, you do not need to create special zone numbers in order to use zone paging. For example, you can create station groups that correspond to physical locations (zones) in your office and add or delete members. See the Interaction Administrator help for details on maintaining the appropriate extensions. Be sure to specify an extension for any group that you intend to page.



Polycom configuration settings for Zone Page

To receive a zone page, the phones must go *off hook* automatically. Configure this setting in the **Alert Information** (`<alertInfo/>`) section of the `xIC.cfg` file. The `alertInfo` value is a string that is compared against the `Alert-Info` value in the SIP message header. If there is a match, the behavior described in the ring class occurs. Ring class 3 auto-answers the incoming call.

In the `xIC.cfg` file, set the `alertInfo` value to `http://localhost/AutoAnswer` and its class to `3`.

```
<voIPProt
<SIP
<alertInfo VoIpProt.SIP.alertInfo.1.value="<http://localhost/AutoAnswer>" voIpProt.SIP.alertInfo.1.class="3"
/>
```

*90x Dialing Options

CIC provides SIP business features that are used by pressing *90 and a feature number on the telephone dialing pad. These features include:

*901	Page	Page a zone, station extension, station group extension, user extension, or workgroup extension. Press the complete sequence to bypass the prompt; for example, *901 1234 where 1234 is the zone number. The phone beeps to prompt you to start talking.
*902	Park a call	Park the active call in a specific orbit which places it on hold and removes it from your station. Press the complete sequence to bypass the prompt; for example, use *902 11 where 11 is the orbit number.
*903	Pick up a parked call	Pick up a parked call from any station. Press the complete sequence to bypass the prompt, for example, use *903 11 where 11 is the orbit number.
*904	List parked calls	Play a list of parked calls. Or you can press *904 *<orbit number>* to hear the orbit number, who the call is from, and how long the call has been on hold.
*905	Pick up a group call	Answer a call ringing on any extension in your group.

However, if you press the NewCall softkey on a Polycom phone and then start dialing, the phone immediately connects the call. This connection occurs after you press * and before you can press the 9 and the rest of the string.

The phone administrator can modify the Digit Map in the Dial Plan map so that Polycom phones pause after NewCall is pressed and attempt to match any *90x string. This pause is configured in the xIC.cfg file.

To configure Polycom phones to allow users to type *90x options:

Add the following string to the dialplan.digitmap attribute in the xIC.cfg file:

```
|*T|*905|*90[1-4]x.T
```

*T	The phone accepts * after the digitmap timeout.
*905	The phone accepts *905 immediately.
*90[1-4]x.T	The phone accepts *901 - *904 and an optional set of digits.

Example:

Based on the default sip.cfg, the modified <dialplan> element would look like:

```
<dialplan


dialplan.impossibleMatchHandling="0"
dialplan.removeEndOfDial="1">
<digitmap dialplan.digitmap="[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT|*T|*905[1-4]x.T"



dialplan.digitmap.timeOut="3"/>
```

For more information, see section 4.6.1.2.1 "Digit Map <digitmap/>" in *Polycom Administrator Guide*.

Change Log

<table>
<tr><th>Date</th><th>Changes</th></tr>
<tr><td>July 31, 2014</td><td>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.</td></tr>
<tr><td>March 31, 2015</td><td>Updated document to reflect the changes required for the transition from Interaction Client .NET Edition to Interaction Desktop. This includes a new CIC client section. Updated the Copyright and Trademarks page.</td></tr>
<tr><td>June 25, 2015</td><td>Rebranding changes, new logo and font colors. Updated Copyright and Trademark Information page.</td></tr>
<tr><td>July 28, 2015</td><td>
IC-131357 Make rebranding changes in technical references
In the CIC client section:
Removed all references to Interaction Client .NET Edition. (This is no longer distributed.)
Removed from boilerplate: Starting with CIC 2015 R3, Interaction Desktop replaces Interaction Client .NET Edition as the primary CIC client.
In IP Phone Configurator section, changed IP Phone Configuration documentation URL,
FROM: It is available under Polycom Related Files on the SIP-Related Download Files page on the Interactive Intelligence Support website: https://my.inin.com/products/sip-platforms/Pages/SIP-Related-Download-Files.aspx#polycomrelatedfiles.
TO: See Download the IP Phone Configurator utility under Polycom Related Files on the SIP-Related Download Files page on the Interactive Intelligence Product & Support Information website: https://my.inin.com/products/sip-platforms/Pages/SIP-Related-Download-Files.aspx.
In the Configuring the TFTP Server section,
In step 2, changed OK to Log In.
Updated screen capture of login dialog box.
Updated screen capture of and of result.
In the Monitoring TFTP Server Activities section,
Updated screen capture of Server Activity Page.
In the Configuring the TFTP Server section
Updated screen capture of Server Parameters page.
In the Changing the TFTP Server Login Credentials section
Updated screen capture of Change Login Credentials page.
Verified IA screen captures are accurate in:
Interaction Administrator primary station configuration
Interaction Administrator secondary station configuration
Call Appearances
Interaction Administrator configuration settings for Zone Page
</td></tr>
<tr><td>April 7, 2017</td><td>Removed reference to Interaction Client Web Edition. Updated copyright page.</td></tr>
<tr><td>September 21, 2017</td><td>Rebranded to Genesys.</td></tr>
<tr><td>March 19, 2018</td><td>Updated document to new format.</td></tr>
</table>

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