1.1 Polycom SoundPoint IP30x, IP320/330, IP430, IP450, IP50x, IP550/560, IP60x, IP650, IP670, IP4000, IP6000, IP7000

IP 301   IP 501   IP 601

IP 650   IP 550/560   IP 330/320

IP 601 + Exp   IP 6000   IP 4000

IP 670 + Exp   IP 7000

1.1.1 Important Notes

- The IP 560 supports Gigabit Ethernet for both the phone uplink as well as the PC port. This includes in-line power for the phone.

- **_WARNING:_** The IP-601, IP-650, IP-670, IP-550/560, IP430, and IP450 power supplies are **_NOT_** compatible with any of the previous models. Prior to the these models, power supplies could be interchanged between units. If a power supply from any other unit is used on the models listed above, it will reboot un-predictably. However, if a power supply from the

601, 550, 430, 450, or 650 is used on any other model, permanent damage could result to the unit.

- Check the SIP 3rd Party Validation Website for current validation status. The SIP 3rd Party Component Validation Website can be viewed at: http://testlab.inin.com
- BootROMs 3.x and newer are signed. Once the 3.x BootROM is installed, the phone cannot be downgraded to the 2.x series BootROM.
- As of firmware v1.5.x, there is no longer an 'ipmid.cfg' this has been merged with the 'sip.cfg'
- As with any Polycom firmware upgrade, it is **not** recommended to use pre-existing configuration files. From version to version the XML tags in the config files change. If older config files are used with newer firmware versions the results will be unpredictable. I3 Support will insist that appropriate config files be used with any given version of Polycom firmware.
- Managed Phone Provisioning is the recommended/tested configuration option for IC 3.0 or above. FTP is recommended for other IC versions.
- Due to the limited configuration options, configuration via the web interface is not recommended or supported.
- SFTP is not supported by Polycom as of the current firmware release, even though it is a selectable menu item.
- Do not mix configuration options (FTP, web interface, or the phone menus). It is very easy for one interface to over ride another and limit the ability to fully configure the phone until a complete reset of the phone and configuration files is performed.

1.1.2   Vendor Documentation

Updated documentation can be found on Polycom's website:
http://www.polycom.com/

1.1.3   Validated Firmware Versions

Unless otherwise noted all phones were validated on the following Firmware and BootROMs.

Firmware: 3.1.2.0392
BootROM: 4.1.2.0037

IP 301 Firmware: 2.1.0.2695 - BootROM version: 3.2.3.0002
IP 300 Firmware: 1.6.7.0098   - BootROM version: 3.2.1
IP 500 Firmware: 1.6.7.0098   - BootROM version: 3.2.1
IP 600 Firmware: 1.6.7.0098   - BootROM version: 3.2.1

Not all combinations of BootROM and firmware versions are validated. The current testing period was performed with the 4.0.0.0423 BootROM. It is recommended to use firmware and Bootrom combinations listed above.

1.1.4 Install

A functional file distribution mechanism will be needed to distribute firmware and configuration files to your phones. You can choose from options like FTP, TFTP, and HTTP/HTTPS. Once a functional FTP server has been setup, credentials to log into the FTP server will be required. The phone will use this account to connect to the FTP server and download BootROM, Firmware, and configuration files.

TFTP is the recommended server for small to midsize implementations. In order for the Polycom phones to function as designed, the ability to write configuration and log files back to the boot server is required. Since TFTP provides no facility for authentication, this opens up the possibility that a malicious user could manipulate configuration files on a standard TFTP server running in a read/write mode. For this reason, version 2.4 of the CIC and EIC products include a custom TFTP service on the server that has more advanced options to enhance the security of TFTP. The ININ/Vonexus TFTP server adds facilities like file extension filters and IP filters to create a secure, yet simple way to support your IP phone base.

A newly supported option available with firmware 2.2.1 is the ability to provision the phones via HTTP and or HTTPS. Interaction Center 3.0 introduces the Managed Phone concept, allowing auto provisioning of the phone from Interaction Administrator. This is not part of any release prior to 3.0. The preferred provisioning method continues to be FTP for releases prior to 3.0.

Note: When provisioning managed phones that are not yet in the list, please use the following profiles. All the phones should be in the list with SU3.

- IP6000 and IP7000 should use the IP4000 managed phone template.
- IP670 should use the IP650 managed phone template.

1.1.5 Configuration

Line Appearances: The line appearance is equivalent to a station in the Interaction Administrator.

SoundPoint IP4000 – 1 line appearance
SoundPoint IP6000 – 1 line appearance
SoundPoint IP7000 – 1 line appearance
SoundPoint IP30x – 2 line appearances
SoundPoint IP430 – 2 line appearances

SoundPoint IP450 – 3 line appearances

SoundPoint IP50x – 3 line appearances

SoundPoint IP550 – 4 line appearances

SoundPoint IP560 – 4 line appearances

SoundPoint IP600 – 6 line appearances

SoundPoint IP601 – 6/20/34/48 line appearances (6 standard, 14 on each side-car with a maximum of three side-cars).

SoundPoint IP650 – 6/20/34/48 line appearances (6 standard, 14 on each side-car with a maximum of three side-cars).

SoundPoint IP670 – 6/20/34/48 line appearances (6 standard, 14 on each side-car with a maximum of three side-cars).

Call Appearances: All line appearances share up to 4 calls.

Using *persistent* connections allows the phone to handle more call appearances than the phone is physically capable. This is done by using the *persistent* connections and the Interaction Center Client.   To manage more calls than the phone is capable (for instance an operator want to handle up to 20 simultaneous calls), check the *Persistent* checkbox in the Station configuration in Interaction Administrator.   The Interaction Client can be used to manage a large number of calls while the phone will be the audio device for the calls.   The phone will show one call (from the Interaction Center) while the Interaction Client will be used to manipulate the calls.

A new feature introduced with the 1.5.x firmware is the ability to have a single line appearance span multiple buttons on the phone. This allows for a more traditional call behavior, similar to that found in many PBX's. Calls can be picked up, selected via the line appearance buttons. And if a call is active on one line, a second call will "roll-down" to the next configured line appearance. Details on how to configure this option are detailed in the Polycom Administrator Guide. In Interaction Administrator set the number of call appearances on the station to match the number of line appearances a particular station will span.

1.1.5.1   Configuration files

The Polycom SoundPoint SIP software package consists of a binary image file (sip.ld), a binary BootROM file (bootrom.ld), and three XML based config files (*macaddress.*cfg, phone*x*.cfg and sip.cfg).

The XML-formatted configuration files consist of master configuration (*macaddress.*cfg) files and phone specific (phone*x*.cfg) files. Master configuration files identify the executable image as well as an ordered list of phone specific configuration files to be applied to the phone.

In the event that a phone's master config file cannot be found, it will attempt to download a generic master config file '000000000000.cfg' if it

exists. The generic master config file contains references to 'phone1.cfg' so this will also be applied to the phone. Here is an example of the default master config file:

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File-->
<!-- Edit and rename this file to <Ethernet-address>.cfg for each phone.-->
<!-- $Revision: 1.13 $  $Date: 2004/11/26 23:30:44 $ -->
<APPLICATION APP_FILE_PATH="sip.ld" CONFIG_FILES="phone1.cfg, sip.cfg"
MISC_FILES="" LOG_FILE_DIRECTORY=""/>
```

Master configuration files contain three XML attributes:

APP_FILE_PATH   the path name of the application executable

CONFIG_FILES   a comma-separated list of configuration files

MISC_FILES   a comma-separated list of other required file

Note:

The order of the configuration files listed in CONFIG_FILES is significant. The files are processed in the order listed (from left to right). The same parameters may be included in more than one file. The parameter found first in the list of files will be the one that is effective.

1.1.5.2   Configuration steps for use with Interaction Center

(un-Managed 3.0 stations or pre-3.0 integrations)

The following steps lay out the minimum steps required to use the Polycom phones as managed SIP stations with the Interaction Center.

1. Set up the FTP server and copy all the template XML files and application files that came with the Polycom phone into the root directory of the FTP server. These files include,

   sip.ld, 000000000000.cfg, phone1.cfg, sip.cfg, bootrom.ld, bootrom.ver

2. Create per-phone configuration files.

   • Create per-phone phone*XXXX*.cfg and *<Ethernet address>*.cfg files by using the phone1.cfg and 000000000000.cfg files as templates.

   • Edit the contents phone*XXXX*.cfg as follows,

      reg.1.displayname="*Name displayed in the SIP Messaging*"

      reg.1.address="*User portion of the station's SIP address*"

      reg.1.label="*The button label on the phone*"

      reg.1.server.1.address="*IP address of Interaction Center*"

   • Edit the CONFIG_FILES in the *<Ethernet address>*.cfg file so that it references the appropriate phone*XXXX*.cfg file, that is, replace the reference to phone1.cfg with phone*XXXX*.cfg.

3. Configure the phones to use the FTP server as its boot server.

      a. Reboot the phone and press the Setup button that appears during the boot screen countdown. The default password to enter the Setup menu is "456".

b. Use the arrow key to navigate to the Server Menu… option. Press the Select button to enter the Server setup.

c. Use the arrow keys and Edit options to enter the IP address of the FTP server in the Server Address: field and enter the username and password in the FTP User: and FTP Password: fields to match those setup on the FTP server.

1.1.5.3   Voicemail Retrieval

There are two ways to configure a Polycom IP phone for voicemail retrieval:

1. **Program the Message Center (Messages button)** Programming the Message Center Connect option is a 3 step process for retrieving voicemails but it's initiated by pressing the Messages button on the phone. To retrieve voicemail, the user presses the Messages button, then selects the Message Center… option followed by the Connect soft-key.

Programming the Message Center is accomplished by setting the following values in the phone specific configuration file.

```
msg.mwi.1.callBackMode="contact"
msg.mwi.1.callBack="7777"
```

Note: 7777 is for this example only and the actual entry should contain the value of the IP Message Button parameter.

To access this setting, the user must,

a. Press the Messages button on the phone.

b. Select 1. Message Center …

c. Select the Connect soft-key

2. **Program a speed dial button**. Programming a speed dial button allows the user to dial the voicemail pilot number by pressing a single button, but it also requires an unused line appearance.

Programming the speed dial button is a simple process. The Polycom phones will look for an XML file called *MACAddress*-directory.xml on the ftp server when it boots. This file contains a list of phone directory entries and speed dial entries. The Polycom distribution comes with a template file called 000000000000-directory~.xml. Add the following entry to create a Messages speed dial button on the Polycom phones.

```
<?xml version="1.0" standalone="yes" ?>
<directory>
  <item_list>
    <item>
      <fn> Messages</fn>
      <ct> 7777</ct>
      <sd>1</sd>
      <rt>1</rt>
```

```
                </item>
            </item_list>
        </directory>
```
Note: 7777 is for this example only and the actual entry should contain the value of the IP Message Button parameter.

1.1.5.4   Auto-answer

The Polycom SoundPoint line supports auto-answer based on call type. For instance, if a call is placed from the Interaction Client, the connection call to the phone will be auto-answered. The same holds true for an ACD assigned interaction (assuming the ACD Options are setup properly) or a connection call as a result of opening a voicemail with the Voxform. The connection call will also be auto-answered if the pickup button is clicked in the Interaction Client, or if the pickup option is clicked in the "toast" popup with the .NET Interaction Client.

In order to enable this functionality, find the following line in the 'sip.cfg' file:

<alertInfo voIpProt.SIP.alertInfo.1.value="" voIpProt.SIP.alertInfo.1.class=""/>

And replace it with this:

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

Next open up the station configuration for the Polycom phone and on the General/Configuration page, enter "Polycom" (not case sensitive) in the Manufacturer field. Make sure and save changes to the station.

After the above steps have been completed, reboot the phone to reload the modified config files. Exit and restart the Interaction Client attached to that phone. Verify operation of auto-answer in the desired scenarios.

1.1.5.5   Zone Paging (xIC 2.4 and beyond)

The Zone paging option requires that the Auto-Answer functionality described in section 8.3.5.5 be enabled. Once auto-answer has been enabled, all phones in a particular zone will be put into a station group in Interaction Administrator.

From any phone on the system dial "*901" and press a line key. Note that you cannot press a line key then dial the "*901" as the Polycom dial plan will interfere with the dial string.

Once you dial the "*901" you'll be ask for the extension of the party you wish to page. You can enter a single station extension here or the extension of the station group.

When the call is places all phones will be paged and placed off-hook to all the caller to announce the "page."

1.1.5.6   Shared Line Appearances (xIC 2.4 and 3.0 un-managed Phones)

This functionality will allow two physical phones to share the same line appearance (station in IA). Along with the ability to share the line appearance the phones will also indicate the status of line in question.

As a call is delivered to the shared line, both stations (A&B) will ring to indicate an incoming call. As one phone in the pair answers the call (A), the LED on the line of the idle phone (B) will glow red (IP50x phones will show horizontally moving phone icon on the LCD display), showing the line is in use. If the user of the idle phone (B) presses the line key, dial tone will delivered and it will be possible to place a call from the line, even though it indicates busy.

If the call is placed on hold (A) the LED will flash red on both phones (A&B) (IP50x phones will show phone icon with the handset turned upside down on the LCD display). The call can then be picked up from the idle phone (B). Once this operation is performed, the call will be connected to that phone and the LED on the first station (A) will glow red indicating a busy status (IP50x phones will show horizontally moving phone icon on the LCD display).

Configuration of the phones is similar to the standard setup with the exception of a couple of changes made to the reg.x.server.x line in the phone specific config file. Here's and example from a primary phone:

reg.3.displayName="ip600-1" reg.3.address="8401" reg.3.label="8401" reg.3.type="shared" reg.3.thirdPartyName="8401" reg.3.auth.userId="" reg.3.auth.password="" reg.3.server.1.address="10.10.220.30" reg.3.server.1.port="" reg.3.server.1.transport="DNSnaptr" reg.3.server.2.transport="DNSnaptr" reg.3.server.1.expires="" reg.3.server.1.register="" reg.3.server.1.retryTimeOut="" reg.3.server.1.retryMaxCount="" reg.3.server.1.expires.lineSeize="" reg.3.acd-login-logout="0" reg.3.acd-agent-available="0" reg.3.ringType="2" reg.3.lineKeys="" reg.3.callsPerLineKey=""

Notice the type is set to shared.

Here's the line config from the other phone that is sharing this particular line appearance:

reg.3.displayName="ip601-1" reg.3.address="84011" reg.3.label="8401" reg.3.type="shared" reg.3.thirdPartyName="8401" reg.3.auth.userId="" reg.3.auth.password="" reg.3.server.1.address="" reg.3.server.1.port="" reg.3.server.1.transport="DNSnaptr" reg.3.server.3.transport="DNSnaptr" reg.3.server.1.expires="" reg.3.server.1.register="" reg.3.server.1.retryTimeOut="" reg.3.server.1.retryMaxCount="" reg.3.server.1.expires.lineSeize="" reg.3.acd-login-logout="0" reg.3.acd-agent-available="0" reg.3.ringType="3" reg.3.lineKeys="" reg.3.callsPerLineKey=""

This is all the configuration required on the phone end. During testing a numbering scheme of xxxx1 was used to note the shared lines. So if the primary extension was 1234, then the first shared appearance would be 12341, and the second would be 12342, etc. This is not a requirement, but something similar should be considered to reduce confusion when troubleshooting.

Configuration in Interaction Administrator for shared lines is covered in the 2.4 documentation.

1.1.5.7   Line Roll-Down (Polycom firmware 1.5.x and beyond)

Post 1.5.x firmware, the Polycom phones have the ability for inbound calls to "ring down," the line keys. This behavior is very similar to legacy PBX operations where each user had a roll-over line. If the primary line is busy, a second incoming call would "roll-down," to the next line key. This behavior can be configured for as many line keys as the Polycom phone has.

Configuration for this feature is on a per line basis and can be found at the end of each line definition in the phone specific config file. Two settings are required:

> 1). reg.1.lineKeys="3"
>
>> This tells phone phone how many line appearances each line definition should take. In this case, once the phone has been booted up, this line will occupy the first three line keys on the phone.
>
> 2). reg.1.callsPerLineKey="1"
>
>> This tells the phone to allow one call per line. This can be increased but will not mimic legacy PBX behavior. As the next call comes in, it will ring the second line key, and then the third, up until the maximum is reached as specified in the above parameter.

The only requirement for this functionality in Interaction Administrator is to configure the maximum call appearances on the specific station to match the maximum number of line keys that you'd defined.

As calls are delivered you will be able to move between them using the line keys.

1.1.5.8   Forwarding and Do Not Disturb Buttons Status Sync

This functionality is being integrated to work with the client. When DND is set through the phone, the client will update the user status to DND. This behaves similarly if the status is set via the client. Neither DND or Forward functionality is currently in 3.0 SU3. Until it is released, these buttons are not supported with the use of managed phones.

1.1.5.9   Conferencing Under TLS

Due to resource limitations on the older phones, conferencing will not work under 501 or 601 models using TLS.

Another issue that can surface is that some of the phone models will not continue a conference if the conference is initiated from the phone, and the originator

disconnects. Especially if a conference into a conference is made. If this type if behavior is to be executed, it is recommended to use the client to perform the conference. This issue was not observed under non-TLS scenarios.