



# Technical Bulletin 43565

## Using Polycom® SoundPoint® IP and Polycom® SoundStation® IP Phones with Asterisk

### Introduction

This document provides introductory information on how to use Polycom® SoundPoint® IP phones and Polycom® SoundStation® IP conference phones with the Asterisk PBX. It offers answers to frequently asked questions along with feature configuration examples for both Asterisk and Polycom phones.

The examples in this document follow the best practices outlined in *Configuration File Management on SoundPoint IP Phones*, available at <http://www.polycom.com/voicedocumentation>. For more information, see the following documents:

- *Administrator's Guide for the Polycom SoundPoint/SoundStation IP Family*, available at <http://www.polycom.com/voicedocumentation>
- The Asterisk wiki at <http://www.voip-info.org>

This document assumes you have an installed and functioning Asterisk server. If you do not have a functioning Asterisk server, see one of the following support pages:

- For Asterisk Business Edition: <http://www.digium.com>
- For general Asterisk information: <http://www.voip-info.org/wiki-Asterisk>

In this document, the terms SoundPoint IP phone, SoundStation IP phone, and Asterisk/Asterisk server refer to the following:

- SoundPoint IP phone—Polycom SoundPoint IP 301, 320, 330, 430, 450, 501, 550, 560, 601, 650, or 670 desktop phone
- SoundStation IP phone—Polycom SoundStation IP 4000, 6000, or 7000 conference phone
- Asterisk or Asterisk server—a functioning Asterisk PBX installation

The examples in this document were tested with Polycom SIP application release 3.1.3 with bootROM 4.1.0 (both available from <http://www.polycom.com/voicedocumentation>) with Asterisk version 1.6.0.1. If there is an issue with a configuration that is specific to a particular software release, the issue is noted.

The Asterisk community is the source of much of the material in this document.

The topics in this document include:

- [Creating a Basic Configuration](#)
- [Using the Messages Key](#)
- [Adding Line Busy Lamps](#)
- [Enabling Auto Answer for Intercom](#)
- [Paging a Group of Extensions](#)
- [Using the Call Park Feature](#)
- [Using the Transfer Key](#)
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## Creating a Basic Configuration

The configuration file fragments in the following figure show the basic settings required to configure both Asterisk and the SoundPoint IP or SoundStation IP phones to successfully interoperate.

Note that these settings are minimal configurations and only show changes from default factory settings on the phone. If you are not sure of the current settings and their impact on the phone in your configuration, restore the phone to the factory default settings (see [Removing Configuration Overrides](#)).

**To create a basic configuration:**

1. For each phone to be connected, define the phone extension number in the Asterisk **sip.conf** file, as shown in the example below.

```
.  
.br/>[8036]  
username=8036username  
secret=8036password  
mailbox=8036@default  
type=peer  
host=dynamic  
dtmfmode=rfc2833  
context=polycom  
canreinvite=yes  
callerid="Polycom Demo" <8036>
```

***Fragment from "sip.conf"***

In the above example, the extension number 8036 is defined. This extension:

- Is defined by the number inside square brackets [8036]
- Has a SIP registration username of 8036username
- Has a SIP registration password of 8036password
- Uses the Asterisk mailbox number 8036 in the default context
- Is an Asterisk type `friend`, meaning it is both a user and a peer
- Is a `dynamic` host according to Asterisk
- Sends DTMF tones as special RTP packets according to RFC2833
- Exists in the Asterisk context `polycom`
- Allows RTP traffic to bypass the Asterisk server
- Sends a caller ID with text as `Polycom Demo` and a number 8036

2. Define the dial plan in the Asterisk **extensions.conf** file for the context declared above, as shown in the example below.

```
.  
.br/>[polycom]  
exten => 8036,1,Dial(SIP/8036,25)  
exten => 8036,2,VoiceMail,u8036  
exten => 8036,102,VoiceMail,b8036
```

***Fragment from “extensions.conf”***

In the above example, the extension number 8036 was added to the dial plan.

When someone dials this extension from within the context `polycom`:

- Asterisk will first try to ring the extension 8036 using the SIP channel, waiting up to 25 seconds for the extension to answer.
  - If the extension is unanswered, Asterisk will direct it to mailbox 8036.
  - If the extension is busy, Asterisk will direct it to mailbox 8036.
3. Once the Asterisk configuration is complete, configure the SoundPoint IP or SoundStation IP phone. You can do this in one of three ways:
    - Via a centralized provisioning (or boot) server using configuration files. This method offers the most flexibility. This is the recommended method of configuring Polycom phones and the method that is used throughout this document.
    - Via the phone’s built-in web server using a web browser. Not all options are available using this method, but you can use this method to configure the phone to register with Asterisk.
    - Via the phone’s menu system (using the phone’s buttons). Not all options are available using this method, but you can use this method to configure the phone to register with Asterisk.

4. If you are configuring your SoundPoint IP or SoundStation IP phone to be provisioned from a boot server, locate the files provided by Polycom in the firmware distribution. There are four configuration files, three systems files, and one language customization directory provided with firmware release SIP 3.1 or later:

- **000000000000.cfg**
- **000000000000-directory~.cfg**
- **phone1.cfg**
- **sip.cfg**
- **sip.ld**
- **sip.ver**
- **SoundPointIPWelcome.wav**
- **SoundPointIPLocalisation** (a sub directory)

5. Using FTP, place all the files listed in step 4 into a directory that is accessible from your phone. Then, do the following:

- a. Copy the default **000000000000.cfg** file to a file named **<mac>.cfg** where **<mac>** is the 12 character MAC address of the phone (which is also the phone's serial number). This MAC address is printed on a sticker attached to the underside of the phone.
- b. Open the **<mac>.cfg** file with a text editor, and then locate the line starting with the text `APPLICATION APP_FILE_PATH`. On this line, change the `CONFIG_FILES` key as shown in the example below.

```
.  
.  
...CONFIG_FILES="<extension>.cfg, sip.cfg"...
```

*Fragment from "<mac>.cfg"*

- c. Copy the default **phone1.cfg** file to a file named **<extension>.cfg** where **<extension>** is the phone's extension number.
- d. Open the **<extension>.cfg** file with a text editor, and then locate the line starting with the text `<phone1>`. On the following line is the registration information the phone requires to register with Asterisk. Change the keys shown below to match your configuration.

In the following example:

- The extension is 8036
- The registration username is 8036username
- The registration password is 8036password

```
.  
.br/>... reg.1.displayName="8036" reg.1.address="8036" reg.1.label="8036"  
reg.1.auth.userId="8036username"  
reg.1.auth.password="8036password" ...  
reg.1.server.1.address="<your_asterisk_IP_address> ...  
.br/>.
```

**Fragment from “<extension>.cfg”**

If your phone supports more than one line (excluding the SoundStation IP phones) you can also define the registration parameters for the other lines by changing the keys starting with `reg.2.displayName`, `reg.2.address`, etc.

- e. Confirm the phone has power and IP connectivity with the Asterisk server. Then, reboot the phone using the key combination or by a power cycle.
- f. Press the **SETUP** soft key. (Note: The SETUP soft key disappears when the Welcome screen displays.)
- g. Enter the phone administrator password. (The default is 456.)
- h. Enable or disable the phone's **DHCP Client**.

If you enable the phone's DHCP client, an IP address and other phone parameters will be supplied by your DHCP server. If you disable the phone's DHCP client, you must specify the IP address, network mask, and default gateway (use the Asterisk server's IP address if this is an isolated network). Note that this is a minimum configuration. At this stage you can also specify the DNS Server IP address(es), NTP Server IP address, GMT Offset, Domain Name, and CDP settings (see [Enabling CDP](#)).

- i. Navigate to **Server Menu**, and then press the **Select** key.
- j. Set the **Server Type** to **FTP**. For other options, see the *Administrator's Guide for the Polycom SoundPoint/SoundStation IP Family*, available at <http://www.polycom.com/voicedocumentation>.
- k. Set the **Server Address** to the IP address of your FTP server where the previously described configuration files are located.
- l. Set the **Server User** to a username with the correct permissions to GET files from your FTP server. By default, the Server User is PlcmSplp.

- m. Set the **Server Password** for the Server User. By default, the Server Password is PlcmSplp.
- n. Set the **Provisioning Method** to **Default**. For other options, see the *Administrator's Guide for the Polycom SoundPoint/SoundStation IP Family*, available at <http://www.polycom.com/voicedocumentation>.
- o. Press the **Exit** key and select **Save & Reboot**.

The phone will now reboot, access your FTP server, load the configuration files, and register with your Asterisk server.

## Using the Messages Key

If you have configured voicemail for an extension, you can configure the Messages key on the phone for one touch access to the voicemail system.

### To configure the Messages key:

1. In `<extension>.cfg`, add the lines shown below.  
(Note: you can change these lines in `sip.cfg` to update all of the phones).

```
.  
.   
.   
up.oneTouchVoiceMail="1 "  
msg.bypassInstantMessage="1 "  
msg.mwi.1.subscribe=" "  
msg.mwi.1.callBackMode="registration "  
msg.mwi.1.callBack=" "  
.   
. 
```

**Fragment from “<extension>.cfg”**

```
.   
.   
exten => 217, hint, SIP/217  
exten => 217,n(vm2),Dial(SIP/217,25,Tt)  
exten => 217,n,VoiceMail(217| u)  
.   
. 
```

**Fragment from “extensions.conf”**

2. In the Asterisk **extensions.conf** file, for each extension, add the logic shown below.

```
.
.
exten => 217, hint, SIP/217
exten => 217, 1, NoOp(SIP/mypeer ${CALLERID(num)} ext ${EXTEN})
exten => 217, n, GotoIf("${CALLERID(num)}" = "${EXTEN}")?vm:vm2)
exten => 217, n(vm), VoicemailMain(${EXTEN}@default)
exten => 217, n(vm2), Dial(SIP/217,25,Tt)
exten => 217, n, VoiceMail(217| u)
```

**Fragment from “extensions.conf”**

If you press the **Messages** key, the phone will dial itself (the first registration), and Asterisk will check if the dialed extension matches the caller ID. If it does, the call will be sent to the voicemail system, passing the phone’s extension as a parameter.

## Adding Line Busy Lamps

Some PBXs have a feature that allows one phone station to indicate the status of many lines using LEDs or other indicators on the phone. This feature, also known as Busy Lamp Feature, is common on key systems.

SoundPoint IP phones can simulate this feature using the phone’s presence features and the Asterisk hint option.

The following examples show three phones (extensions 8003, 8027, and 8036). Each extension will have the other two lines configured using presence so that a user of that phone can view the status of the other lines.

## To add line busy lamps:

1. Define the extensions in the Asterisk **sip.conf** file, as shown below.

```
.  
. [br/>[br/>8003]  
username=8003username  
secret=8003password  
mailbox=8003@default  
type=friend  
host=dynamic  
dtmfmode=rfc2833  
context=polycom  
canreinvite=yes  
callerid="Polycom Demo 1" <8003>  
  
[8027]  
username=8027username  
secret=8027password  
mailbox=8027@default  
type=friend  
host=dynamic  
dtmfmode=rfc2833  
context=polycom  
canreinvite=yes  
callerid="Polycom Demo 2" <8027>  
  
[8036]  
username=8036username  
secret=8036password  
mailbox=8036@default  
type=friend  
host=dynamic  
dtmfmode=rfc2833  
context=polycom  
canreinvite=yes  
callerid="Polycom Demo 3" <8036>
```

***Fragment from "sip.conf"***

2. Enable the presence feature on the SoundPoint IP phone.

Open the phone's **sip.cfg** file with a text editor. Then, locate the line starting with the text `<feature`, and then enable the presence feature by setting the key value for that feature to **1**, as shown below.

```
.  
.  
<feature feature.1.name="presence" feature.1.enabled="1" ...  
.  
.
```

**Fragment from "sip.cfg"**

3. Add the extensions you want to monitor to the phone's directory, stored in the **<mac>-directory.xml** file, as shown below.

```
<directory>  
  <item_list>  
    <item>  
      <ln>Demo 1</ln>  
      <fn></fn>  
      <ct>8003</ct>  
      <sd>1</sd>  
      <rt>1</rt>  
      <dc/>  
      <ad>0</ad>  
      <ar>0</ar>  
      <bw>1</bw>  
      <bb>0</bb>  
    </item>  
    <item>  
      <ln>Demo 2</ln>  
      <fn></fn>  
      <ct>8027</ct>  
      <sd>2</sd>  
      <rt>1</rt>  
      <dc/>  
      <ad>0</ad>  
      <ar>0</ar>  
      <bw>1</bw>  
      <bb>0</bb>  
    </item>  
    <item>  
      <ln>Demo 3</ln>  
      <fn></fn>  
      <ct>8036</ct>  
      <sd>3</sd>  
      <rt>1</rt>  
      <dc/>  
      <ad>0</ad>  
      <ar>0</ar>  
      <bw>1</bw>  
      <bb>0</bb>  
    </item>  
  </item_list>  
</directory>
```

**Fragment from "<mac>-directory.xml"**

In the above example:

- The text between `<ln>` and `</ln>` (last name) and `<fn>` and `</fn>` (first name) is shown on the phone's LCD display.
- The number between `<ct>` and `</ct>` (contact) is the extension number you want to monitor.
- The number between `<sd>` and `</sd>` (speed dial) determines the line appearance to use for this line indicator. Position 1 is the first free line position available.
- The number between `<rt>` and `</rt>` (ring type) is not relevant since these appearances only represent the status of extensions and do not ring.
- The `<dc/>` (divert contact) on its own means there is no divert contact and must be included even though not relevant for this application.
- The number between `<ad>` and `</ad>` (auto divert) should be **0** although it is not used in the application.
- The number between `<ar>` and `</ar>` (auto reject) should be **0** although it is not used in the application.
- The number between `<bw>` and `</bw>` (buddy watch) must be **1** to indicate that we wish to monitor the presence of this entry.
- The number between `<bb>` and `</bb>` (buddy block) should be **0** although it is not used in the application.

An alternative to creating **<mac>-directory.xml** on the provisioning server is to manually add the entries using the phone's Directory button and the Add Contact soft key. If you add entries this way, follow the descriptions in the above list to create each monitored contact.

4. In the Asterisk dial plan, add a hint line for each extension you want to monitor. In the example below, three hint lines are added after the context `[polycom]`.

```
.  
.
[polycom]
exten => 8003,hint,SIP/8003
exten => 8027,hint,SIP/8027
exten => 8036,hint,SIP/8036
exten => 8003,1,Dial(SIP/8003,25)
exten => 8003,2,VoiceMail,u8003
exten => 8003,102,VoiceMail,b8003
exten => 8027,1,Dial(SIP/8027,25)
exten => 8027,2,VoiceMail,u8027
exten => 8027,102,VoiceMail,b8027
exten => 8036,1,Dial(SIP/8036,25)
exten => 8036,2,VoiceMail,u8036
exten => 8036,102,VoiceMail,b8036
```

***Fragment from "extensions.cfg"***

5. Reboot the phones to ensure they use the configured directory entries and have the presence feature enabled. The main line for each phone should be registered and the next two line keys should show the status of the other lines.

You can check your configuration using the Asterisk commands **sip show subscriptions** and **core show hints**, as shown below.

```
asterisk*CLI> sip show subscriptions
Peer           User           Call ID        Extension      Last state     Type           Mailbox
192.168.3.242  8003           da782481-a626e3 8003@polycom   Idle           xpidf+xml     <none>
192.168.3.242  8003           da782481-a626e3 8003@polycom   Idle           xpidf+xml     <none>
192.168.3.243  8027           eb782481-a643e3 8027@polycom   Idle           xpidf+xml     <none>
192.168.3.243  8027           da782481-a626e3 8027@polycom   Idle           xpidf+xml     <none>
192.168.3.244  8036           da782481-a626e3 8036@polycom   Idle           xpidf+xml     <none>
192.168.3.244  8036           da782481-a626e3 8036@polycom   Idle           xpidf+xml     <none>
```

***Fragment from "sip show sub"***

```
asterisk*CLI> core show hints
-- Registered Asterisk Dial Plan Hints ==
      8003@polycom           : SIP/8003           State:Idle           Watchers  2
      8027@polycom           : SIP/8003           State:Idle           Watchers  2
      8036@polycom           : SIP/8003           State:Idle           Watchers  2
```

***Fragment from "core show hints"***

## Enabling Auto Answer for Intercom

You can configure the SoundPoint IP or SoundStation IP phones with an Asterisk server to operate like an intercom system. For example, you could dial a specific number and the remote party phone would automatically answer after a short ring. Once answered, the call would connect and you could speak freely in both directions.

### To enable auto answer for intercom:

1. Modify the Asterisk dial plan.

The following example shows how a simple dial plan looks before you change it. In the example, extension 8027 will ring for 25 seconds and if the user does not answer within that time, the call will forward to a voice mailbox. The call will also forward to a voice mailbox if the extension is busy.

```

.
.
[polycm]
exten => 8027,1,Dial(SIP/8027,25,tr)
exten => 8027,n,VoiceMail(8027,u)
exten => 8027,102,VoiceMail(8027,b)
.
.

```

**Fragment from "extensions.conf"**

Since it is unlikely you will always want extension 8027 to automatically answer a call, it is a good idea to assign a separate number for the same physical extension.

In the following example, the first digit is changed (from 8027 to 9027). To dial the person on extension 8027, you would dial 8027. To dial them and force them to automatically answer, you would dial 9027.

To modify the dial plan, add the additional extension (if desired) and set the ALERT\_INFO variable, as shown in the example below.

```

.
.
[polycm]
exten => 9027,1,SIPAddHeader(Alert-Info: info=intercom)
exten => 9027,n,Dial(SIP/8027)

exten => 8027,1,Dial(SIP/8027,25,tr)
exten => 8027,n,VoiceMail(8027,u)
exten => 8027,102,VoiceMail(8027,b)
.
.

```

**Fragment from "extensions.conf"**

The value of the ALERT\_INFO variable is not important, but it must match the phone configuration performed in step 2 (below). Ideally, it should be a short descriptive string. (In this example, the value is set to `intercom`.)

In the above example, when the digits 9027 are dialed, the variable is set, and then Asterisk sends the SIP invite (with the set variable) to extension 8027.

2. Configure the phone to interpret the variable sent by Asterisk, as shown in the example below.

Open the phone's **sip.cfg** file with a text editor and locate the line starting with the text `<alertinfo`. Then, add the key `voIpProt.SIP.alertinfo.2.value` with the value you set in the Asterisk dial plan in step 1. As well, add the key `voIpProt.SIP.alertinfo.2.class` with a value of **4**.

```
.
.
<alertInfo ...
voIpProt.SIP.alertInfo.2.value="intercom" voIpProt.SIP.alertInfo.2.class="4"/>
```

**Fragment from "sip.cfg"**

These changes allow the phone to recognize the alert in the invite message sent by Asterisk and to give it an internal ring class of 4. This internal class is used to reference the ring type to use, as shown below.

```
.
.
<VISUAL_ONLY se.rt.2.name="Visual" se.rt.2.type="visual"/>
<AUTO_ANSWER se.rt.3.name="Auto Answer" se.rt.3.type="answer"/>
<RING_ANSWER se.rt.4.name="Ring Answer" se.rt.4.type="ring-answer" ...
... se.rt.timeout="200" ...
... se.rt.ringer="2" ...
... se.rt.4.callWait="6" ...
... se.rt.4.mod="1"/>
```

**Fragment from "sip.cfg"**

With an internal ring class of 4, the line starting with `<RING_ANSWER` is referenced. If the `voIpProt.SIP.alertinfo.1.class` was set to a value of 3, the line starting with `<AUTO_ANSWER` would have been referenced, prompting the phone to answer without any ringing at all.

The default setting for `se.rt.4.timeout` (the amount of time the phone will ring before auto-answering) is 2000 (milliseconds). For the intercom application, the setting is changed for a much shorter ring time of 200 milliseconds.

## Paging a Group of Extensions

You can page a group of extensions so that when you dial a single number, a defined group of phones will automatically answer. Then, for example, the caller can make an announcement to all the answered phones.

**To page a group of extensions:**

1. Configure the phones to auto answer as described in [Enabling Auto Answer for Intercom](#), above. Do the following:
  - Create an alert-info entry for paging.

- Add entries for `voIpProt.SIP.alertinfo.3` to the line starting with `<ALERTINFO.`
- Modify the `<CUSTOM_1` entry in the ringtype, as shown below.

```
.
.
.
.
voIpProt.SIP.alertInfo.3.value="pager" voIpProt.SIP.alertInfo.3.class="8"/>
.
.
.
<CUSTOM_1 se.rt.8.name="Custom 1" se.rt.8.type="ring-answer" se.rt.8.ringer="10"
se.rt.8.callWait="7" se.rt.8.mod="1"/>
```

**Fragment from “sip.cfg”**

Notice the `se.rt.ringer` distinctive ring tone to be played to indicate an imminent page. This will provide a different ring tone from the intercom application described in the previous section, [Enabling Auto Answer for Intercom](#).

2. Modify the **extensions.conf** file, as shown below.

```
exten => *77,1,SIPAddHeader(Alert-Info: Ring Answer)
exten => *77,n,Page(SIP/200&SIP/217&SIP/203&SIP/205)
```

**Fragment from “extensions.conf”**

With the previous configuration complete, dialing the pager extension number will initiate a call to all the extensions defined in the dial plan (extensions 200, 217, 203 and 205 for this example) and the calls will be placed in a listen-only MeetMe room. Extensions that are busy will be unaffected. All extensions will be disconnected when the caller/pager hangs up. (Note: This requires a working MeetMe installation.)

## Using the Call Park Feature

The call park feature may be enabled by setting `feature.11.enabled` to a value of **1**. However, the call park feature implemented on Polycom phones does not work with Asterisk. If you use the new Enhanced Feature Keys (EFK) and Configurable Soft Keys features available in 3.1, you can create a custom Park soft key.

## To use the call park feature:

1. In **sip.cfg**, enable the Enhanced Feature Keys feature by changing the key `feature.18.enabled` to a value of **1**, as shown below.

```
.  
.br/>feature.18.name="enhanced-feature-keys" feature.18.enabled="1"  
.br/>.
```

### Fragment from "sip.cfg"

2. In **sip.cfg**, create a custom soft key by changing the values of `softkey.1`, as shown in the example below. (In the example, extension 700 is used as the parking lot number.)

```
.  
.br/>softkey.1.label="Park"  
softkey.1.action="$FDialpadPound$$FDialpad1$$Cp1$700$Tdtmf$"  
softkey.1.enable="1" softkey.1.precede="1" softkey.1.use.active="1"  
.br/>.
```

### Fragment from "sip.cfg"

3. Configure the Asterisk parking lot by updating **features.conf**, as shown in the example below. (In the example, extension 700 is used as the parking lot number.)

```
parkext => 700 ; What extension to dial to park  
parkpos => 701-720 ; What extensions to park calls on. These needs to be  
 ; numeric, as Asterisk starts from the start position  
 ; and increments with one for the next parked call.  
context => parkedcalls ; Which context parked calls are in  
parkingtime => 45 ; Number of seconds a call can be parked for  
 ; (default is 45 seconds)  
courtesytone = beep ; Sound file to play to the parked caller  
 ; when someone dials a parked call  
 ; or the Touch Monitor is activated/deactivated.  
parkedplay = caller ; Who to play the courtesy tone to when picking up a parked  
call  
 ; one of: parked, caller, both (default is caller)  
adsipark = yes ; if you want ADSI parking announcements  
findslot => next ; Continue to the 'next' free parking space.  
 ; Defaults to 'first' available  
parkedmusicclass=default ; This is the MOH class to use for the parked channel  
 ; as long as the class is not set on the channel directly  
 ; using Set(CHANNEL(musicclass)=whatever) in the dialplan
```

### Fragment from "features.conf"

When the phone is in an active state (that is, the phone has an active call), an extra soft key (the Park soft key) displays. The Park soft key is the leftmost soft key (since `softkey.1.precede="1"`). When you press the Park soft key, a # (Transfer tone) is sent (via DTMF) followed by the digits 700.

The following figure shows an active call. The regular soft keys have shifted to the right to accommodate the new Park soft key. You can access the Transfer and Confer soft keys via the More soft key.



## Using the Transfer Key

The Transfer key on the Polycom phone will initiate an Attended/Blind Transfer using the SIP REFER message. However, some customers have set up their server to use “star codes” to do transfers, and, in some cases, the Transfer key does not work. Using the EFK feature and the ability to remap hard keys to speed dial numbers, the Transfer key functionality can be changed to accommodate this.

### To use the Transfer key:

Note: In the following examples, `blingxfer` and `attxfer` have been set to **#1** and **#2** in `features.conf`, and extensions are 3 digits.

1. In `sip.cfg`, enable the Enhanced Feature Keys feature by changing the key `feature.18.enabled` to a value of **1**, as shown below.

```
.  
.br/>feature.18.name="enhanced-feature-keys" feature.18.enabled="1"  
.br/>.
```

**Fragment from “sip.cfg”**

2. In **sip.cfg**, define an EFK macro at the end of the file, as shown below.

```
.
.
<efk>
  <version efk.version="2"/>
  <efklist
    efk.efklist.1.mname="AstTrans"
    efk.efklist.1.use.active="1"
    efk.efklist.1.status="1"
    efk.efklist.1.label="Transfer"
    efk.efklist.1.action.string="$P1N5$$Tdtmf$" />
  <efkprompt
    efk.efkprompt.1.status="1"
    efk.efkprompt.1.label="Enter Number: "
    efk.efkprompt.1.userfeedback="visible"
    efk.efkprompt.1.type="numeric"
    efk.efkprompt.1.digitmatching="style1" />
  </efk>
</sip>
```

**Fragment from "sip.cfg"**

3. In **<mac>-directory.xml**, create a Speed Dial with a contact number that is the name of the macro (in this case, AsteriskTransfer prefixed with a ! sign), as shown in the example below.

```
.
.
  <item>
    <ct>!AstTrans</ct>
    <sd>4</sd>
  </item>
.
.
```

**Fragment from "<mac>-directory.xml"**

4. In **sip.cfg**, bind the Transfer key to the macro, as shown below.

```
.
.
  <keys
    key.scrolling.timeout="1"
    key.IP_650.37.function.prim="SpeedDial"
    key.IP_650.37.subPoint.prim="4"
  .
  .
```

**Fragment from "sip.cfg"**

In the above example, the value of `x` for `key.x.37.function.prim` was set to `IP_650`. For other possible values of `x`, see the *Administrator's Guide for the Polycom SoundPoint/SoundStation IP Family*, available at <http://www.polycom.com/voicedocumentation>.

The above steps bound the Transfer key (keynumber 37) to the function SpeedDial. The function SpeedDial was given an index number of 4.

This index number corresponds to the speed dial entry that was created with a contact (the `<ct>` entry) of `!AstTrans`. The “!” indicates that this is an EFK macro name. The `efk` entries defined this macro. The macro prompts for a number using the `efkprompt` number 1 (the 1 in the P1N5 string), as shown in the example below. The number can be up to 5 digits (the 5 in the P1N5 string). (This allows “#1XXX or #2XXX, where XXX is a 3-digit extension. For longer extensions, change P1N5 to P1N<length of an extension>.) This string is then sent using RFC2833 DTMF.



Prefixing the extension number with either #1 or #2 will conduct a blind or attended transfer respectively when you press the Enter soft key.

## Configuring Idle Display Bitmaps

You can use the unused screen area on SoundPoint IP phones to display a customizable image (for example, an Asterisk logo).



For detailed information on how to add a background logo and animate it, see the *Administrator's Guide for the Polycom SoundPoint IP/SoundStation IP Family*, available at <http://www.polycom.com/voicedocumentation>.

## Remotely Rebooting the Phone to Force a Configuration Reload

You can remotely reboot a SoundPoint IP or SoundStation IP phone by using the SIP NOTIFY method with an event type of check-sync. When the phone receives the SIP message, it will do one of the following (depending on how the phone is configured):

- If the key `voIpProt.SIP.specialEvent.checkSync.alwaysReboot` (in `sip.cfg`) is set to **0**, then the phone will check the provisioning server and only reboot if any of the files listed in `<mac>.cfg` have changed.
- If the key `voIpProt.SIP.specialEvent.checkSync.alwaysReboot` (in `sip.cfg`) is set to **1**, then the phone will always reboot.

When the phone reboots, it will reload its configuration files from the provisioning server.

### To remotely reboot the phone to force a configuration reload:

1. If you have Asterisk 1.4 or later, you can send a check-sync to the phone from the CLI. In `sip_notify.conf`, add the lines shown below. (If you have Asterisk 1.6, the lines already exist in `sip_notify.conf`.)

```
.  
.br/>[polycom-check-cfg]  
Event=>check-sync  
Content-Length=>0
```

**Fragment from “sip\_notify.conf”**

2. From the CLI command line, add the line shown below (1.6 syntax is used).

```
sip_notify polycom-check-cfg <peer>
```

## Removing Configuration Overrides

Configuration changes made to the SoundPoint IP and SoundStation IP phones via the menu keys or a web browser are stored locally in flash memory and also written back to the boot server (if configured to do so). To delete these settings, you must use the phone’s menu key.

### To remove configuration overrides:

1. Press the **Menu** key on the phone console.
2. Select **Settings > Advanced**.
3. Enter the administrator password (the default is 456), and then press the **Enter** soft key.
4. Select **Admin Settings > Reset to Default > Reset Local Config**.
5. Press the **Yes** soft key. The phone will restart.

If the phone is configured to also write the information back to the boot server, there will be files on the boot server in the form **<mac-address>-phone.cfg**. If these files exist, you must remove them, otherwise the phone will reload the overrides when it reboots. Since you must remove these files, an easier and alternative method to remove a phone's overrides is to delete its **<mac-address>-phone.cfg** file, and then reboot the phone.

## Enabling CDP

All SoundPoint IP and SoundStation IP phones support Cisco Discovery Protocol (CDP). CDP does the following:

- When using PoE, CDP can help the switch determine how much power is required by the phone to operate. For more information, see the *Administrator's Guide for the Polycom SoundPoint/SoundStation IP Family*, available at <http://www.polycom.com/voicedocumentation>.
- CDP can also allocate VLANs. For more information, see [Assigning VLANs](#), below.

CDP settings can only be configured through the phone's menus.

### To enable the CDP setting:

1. Press the **Menu** key on the phone console.
2. Select **Settings > Advanced**.
3. Enter the administrator password (the default is 456), and then press the **Enter** soft key.
4. Select **Admin Settings > Network Configuration**.
5. Scroll to **CDP**, and then press the **Edit** key.
6. Use the arrow keys to choose either Enabled or Disabled, and then press the **Ok** soft key.
7. Press the **Select** soft key.

The phone will now use send and receive CDP packets.

## Assigning VLANs

VLANs provide segregated voice and data paths and allow security and QoS policies to be optimised for each specific application.

The SoundPoint IP and SoundStation IP phones can place their voice traffic into any given VLAN while, at the same time, passing data traffic from any attached PC through the native VLAN of the 802.1q link to the access switch.

By default, the voice VLAN is unspecified. This means that voice and data traffic are untagged and transported in the native VLAN.

You can assign the voice VLAN in one of three ways:

- Manually, using the menus on the phone
- Automatically, using DHCP
- Automatically, using CDP

You cannot specify the VLAN using the configuration files or the phone's web interface.

If the VLAN is specified by more than one of the above methods, then the allocation will take place in this order: allocation via CDP, manual allocation, and then allocation via DHCP.

#### To manually assign a VLAN ID:

(Note: You can access similar configuration screens by pressing the SETUP soft key during the phone boot process.)

1. Press the **Menu** key on the phone console.
2. Select **Settings > Advanced**.
3. Enter the administrator password (the default is 456), and then press the **Enter** soft key.
4. Select **Admin Settings > Network Configuration**.
5. Scroll to **VLAN ID**, and then press the **Edit** soft key.
6. Enter the VLAN ID, press the **Ok** soft key, and then press the **Exit** soft key.
7. Press the **Select** soft key.  
The new voice VLAN is now assigned, but you need to manually reboot the phone so it will operate correctly.
8. Manually reboot the phone.

#### To automatically assign a VLAN ID using DHCP:

1. Press the **Menu** key on the phone console.
2. Select **Settings > Advanced**.
3. Enter the administrator password (the default is 456), and then press the **Enter** soft key.
4. Select **Admin Settings > Network Configuration**.
5. Scroll to **DHCP Client** and make sure it is Enabled. If it is Disabled, press the **Select** soft key, and then use the arrow keys to select Enabled. Then, press the **Select** soft key.
6. Scroll to **VLAN Discovery**, and then press the **Edit** soft key.
7. Use the arrow keys to select either Fixed or Custom (depending on the DHCP option you want to use). Then, press the **Ok** soft key. The VLAN Discovery field can be set to one of three values:
  - Disabled—DHCP discovery of the VLAN is not used.
  - Fixed—DHCP discovery of the VLAN is enabled and DHCP will look for the ID in DHCP options 128, 144, 157, or 191.
  - Custom—DHCP discovery of the VLAN is enabled and DHCP will look for the ID in the DHCP option specified in the **VLAN ID Opt** field.
8. Do one of the following:
  - If you selected the Fixed option, press the **Exit** soft key twice, and then go to step 9.
  - If you selected the Custom option, scroll to **VLAN ID Opt**, press the **Edit** soft key, and then enter the DHCP option you want to use to supply the VLAN ID. Then, press the **Ok** soft key and then press the **Exit** soft key.
9. Press the **Select** soft key.
10. Manually reboot the phone.

#### To automatically assign a VLAN ID using CDP:

- Enable CDP. For more information, see [Enabling CDP](#).

## Additional Resources

### Technical Bulletins

Refer to the technical bulletins available at

[http://www.polycom.com/support/voice/soundpoint\\_ip/VoIP\\_Technical\\_Bulletins\\_pub.html](http://www.polycom.com/support/voice/soundpoint_ip/VoIP_Technical_Bulletins_pub.html).

### Software Release Matrix

A table showing a list of releases, the phones that are supported by each release, the release date, as well as links to the download page for each release and release notes, is available at

[http://downloads.polycom.com/voice/voip/sip\\_sw\\_releases\\_matrix.html](http://downloads.polycom.com/voice/voip/sip_sw_releases_matrix.html)

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