

MAX Communication Server  
Release 7.0 Update 1

Polycom IP Phone  
Configuration Guide



**Intended audience:**  
**AltiGen Authorized Partners**

June 18, 2014

## Contents

1	Introduction .....	3
1.1	About Firmware .....	3
1.2	Supported IP Phones.....	3
1.3	Prerequisites .....	4
2	Configuration Procedures .....	4
2.1	Configure the Extension to Accept a Third-party SIP Phone .....	4
2.2	Set Up FTP Server to Load Configuration Files to Polycom Phone .....	5
2.3	Update Firmware and the Default Configuration Files .....	7
2.4	Prepare the Configuration Files .....	8
	Create the <i>altigen2014.cfg</i> Configuration File.....	9
	Create the Extension-Specific Configuration Files .....	11
	Create the Phone-Specific Configuration File – <i>MAC_Address.cfg</i> .....	12
2.5	Configure the FTP and Password for Each Phone.....	13
2.6	Set the Codec Profile to DTMF Delivery.....	14
2.7	Troubleshooting: Reset Configuration to Factory Default.....	15
2.8	Advanced Phone Features Not Supported in Polycom IP phones .....	16
2.9	Limitations When Using Polycom IP Phones with MaxCS 7.0.....	16
3	Altigen Technical Support .....	17

Altigen Communications, Inc.

679 River Oaks Parkway  
San Jose, CA 95134  
Telephone: 888-Altigen (258-4436)  
Fax: 408-597-9020  
E-mail: [info@altigen.com](mailto:info@altigen.com)  
Web site: [www.altigen.com](http://www.altigen.com)

All product and company names herein may be trademarks of their registered owners.

Copyright © Altigen Communications, Inc. 2014. All rights reserved.

## 1 Introduction

This document provides configuration details for Polycom VVX, SoundPoint, and SoundStation IP phones and lists the limitations of using the Polycom IP phones with AltiGen's MaxCS 7.0 system.

This document applies to MaxCS 7.0 or later.

### 1.1 About Firmware

If you have a Polycom SoundPoint IP phone model 6000, 7000, IP 321, IP 331, IP 335, IP 560 or IP 650, still using SIP Application firmware 3.2.7 or earlier, we recommended that you **do not** update the firmware to 4.x.x version for these phones. This recommendation is based upon the fact that the configuration files between these two firmware versions are incompatible.

Some models, such as the IP 430, cannot be upgraded to firmware 4.x.x or later. Refer to [http://downloads.polycom.com/voice/voip/sip\\_sw\\_releases\\_matrix.html](http://downloads.polycom.com/voice/voip/sip_sw_releases_matrix.html) for more information.

If you do choose to upgrade the firmware on one of these phones to version 4.x.x, you must modify the configuration according to this document. Otherwise, the phone will not work properly.

There is one exception to this guideline: If you have a Polycom SoundPoint IP phone with firmware 3.x.x and are deploying a **new MaxCS installation** (either Premise or Private Cloud), we recommend that you do upgrade to firmware version 4.x.x from version 3.x.x.



Upgrading from firmware version 3.x.x to 4.x.x is irreversible.

For details regarding upgrading the firmware for SoundPoint phones, refer to the following site:

[http://support.polycom.com/PolycomService/support/us/support/eula/ucs/UCagreement\\_combined\\_4\\_0\\_4\\_combined.html](http://support.polycom.com/PolycomService/support/us/support/eula/ucs/UCagreement_combined_4_0_4_combined.html).

If the link is missing or becomes invalid in the future, search the Internet for the phrase, "Polycom Matrix."

We recommend that you use a different FTP server to perform firmware upgrades, if possible. Mixing the configuration files, the firmware files, and the BootROM files in one folder is not recommended.

### 1.2 Supported IP Phones

MaxCS 7.0 supports the following Polycom phone models:

- VVX phone models: 300/310, 400/410, 500 , 600, with firmware version 4.1.6.4835
- SoundPoint models: , IP 321, IP 331, IP 335, IP 560, IP 650, with firmware version 4.0.4.2906 (BootROM 5.0.4.x or BootROM 5.0.5.x)
- SoundStation models: 6000, 7000 with firmware version 4.0.4.x

### 1.3 Prerequisites

Before you proceed, make sure that your environment meets the following minimum requirements:

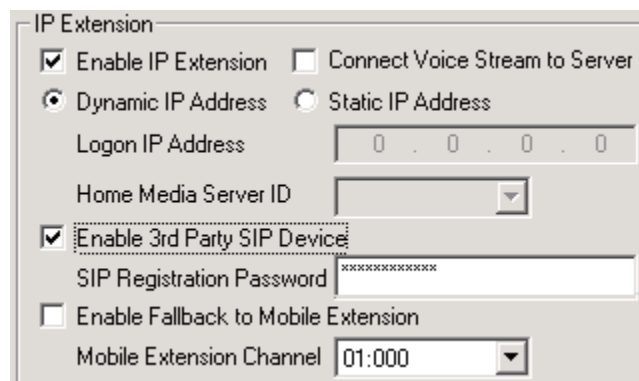
- You must have MAXCS Release 7.0 or later
- You must have an AltiGen third-party SIP Device license for each Polycom IP phone

**Note:** NAT traversal for a Polycom IP phone to a remote site behind NAT is supported.

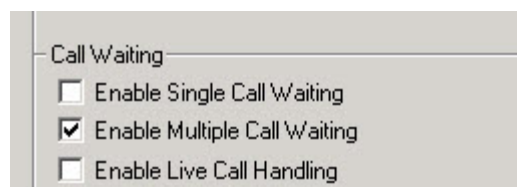
## 2 Configuration Procedures

### 2.1 Configure the Extension to Accept a Third-party SIP Phone

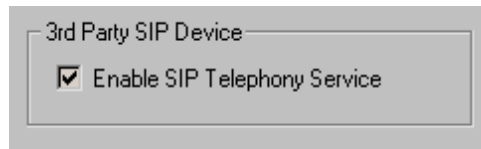
1. Log into MaxAdministrator.
2. Choose **PBX > Extension Configuration > General**.
3. Create an extension and set the following parameters:
  - a) Enable IP extension
  - b) Dynamic IP Address (do *not* configure a static IP address for a third-party SIP device; doing so may cause SIP registration failure)
  - c) Enable 3<sup>rd</sup> Party Sip Device and enter a SIP registration password. (This password can be different from the extension password.)



4. Choose **PBX > Extension Configuration > Answering**.
5. In the **Call Waiting** section, select **Enable Multiple Call Waiting**. (Single Call Waiting does not work with third-party phones.)



6. Choose **PBX > AltiGen IP Phone Configuration**.
7. Select the extension number to which the Polycom IP phone will connect.
8. Check **Enable SIP Telephony Service** in the 3<sup>rd</sup> Party SIP Device panel. This will enable the extension to use SIP Hold, Transfer, and Conference.



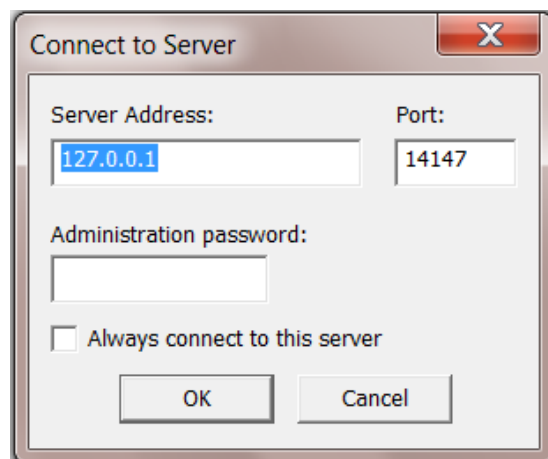
## 2.2 Set Up FTP Server to Load Configuration Files to Polycom Phone

In this guide, the open-source FTP server *FileZilla* is used to demonstrate how to load configuration files to the Polycom IP phone. If you are using a different FTP server, the configuration procedure will be similar.

**Note:** Stop other FTP servers while using FileZilla as a configuration server for Polycom IP phones.

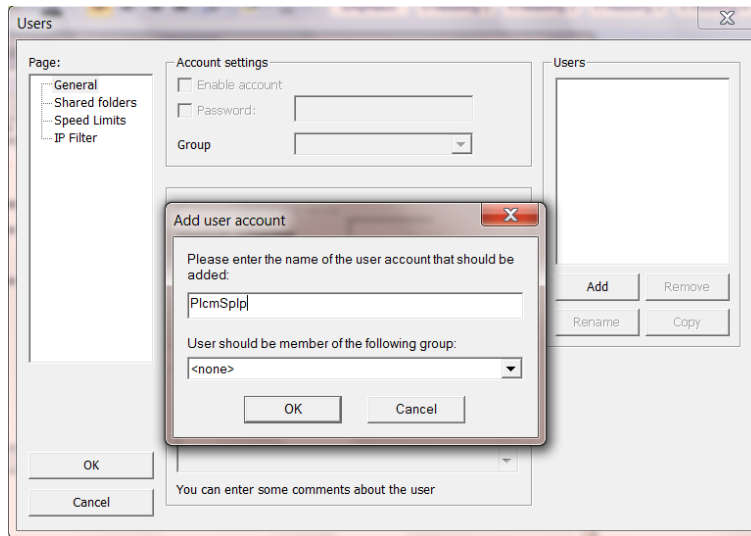
For a Private Cloud or On-Premise setup, consider installing the FTP server on the MaxCS server. Port forwarding for FTP is required if the MaxCS is behind NAT.

1. Download FileZilla FTP Server from <http://filezilla-project.org/download.php?type=server>.
2. Set up the FTP user and home directory.
  - a) Start FileZilla Server from the **Start** menu and connect to the server. (The Administration password is blank by default.)

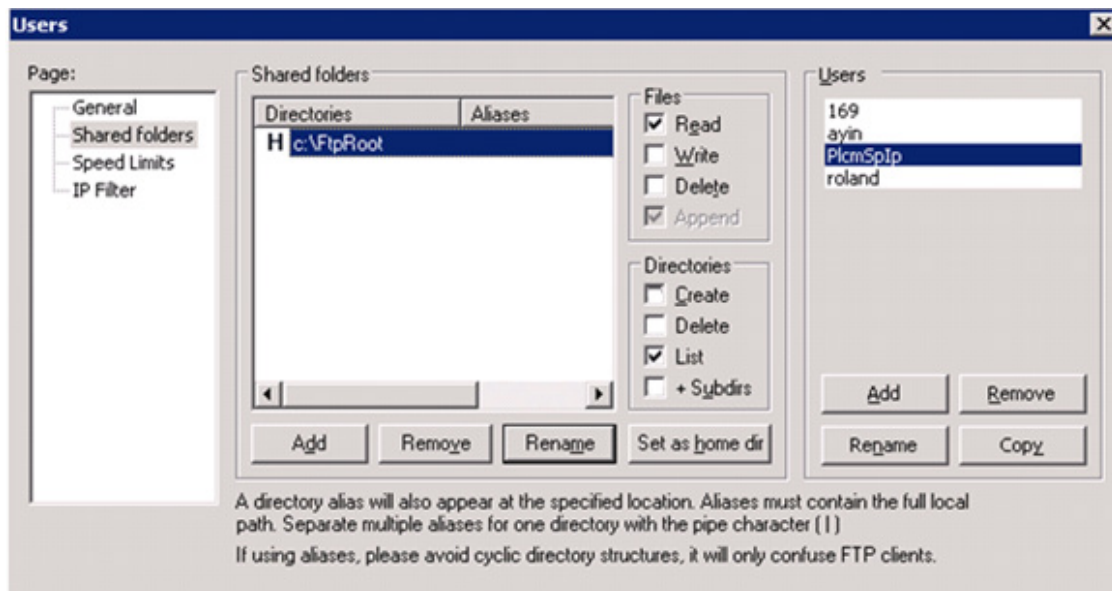


- b) From the **Edit** menu, select **Users**.

- c) In the **General** page, add an FTP user “PlcmSpIp” and set “PlcmSpIp” as the password. These are Polycom phone factory defaults. The password is case-sensitive.



- d) Click the **Shared folders** page and add the directory you want to use as the FTP root, for example “C:\FtpRoot”, then click the **Set as home dir** button to make it the home directory.
- e) Uncheck all of the privileges but “read” and “list” in the *Files and the Directories* sections and click **OK**. If you enable “write” privileges, then the extension’s SIP password will be sent back to the FTP server in plain text. That may cause security concerns.



## 2.3 Update Firmware and the Default Configuration Files

For SoundPoint phones, in order update the firmware from the *Software Update* page, a version 5.x.x.x Updater (BootROM) is required. Please refer to the “Polycom Release Matrix” at this URL:

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

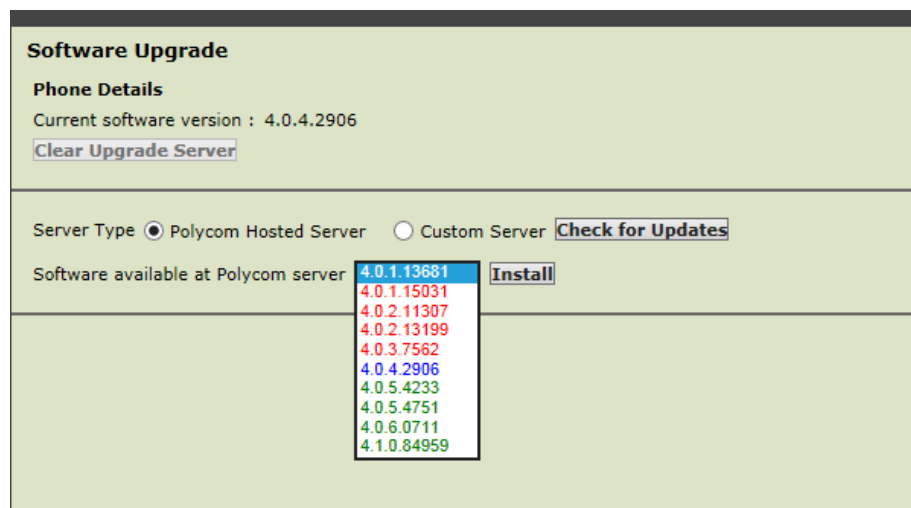
Use one of these sites to upgrade:

[http://support.polycom.com/PolycomService/support/us/support/eula/ucs/UCagreement\\_combined\\_4\\_0\\_4\\_combined.html](http://support.polycom.com/PolycomService/support/us/support/eula/ucs/UCagreement_combined_4_0_4_combined.html)

[http://support.polycom.com/PolycomService/support/us/support/eula/ucs/UCagreement\\_4\\_0\\_5\\_C\\_combined.html](http://support.polycom.com/PolycomService/support/us/support/eula/ucs/UCagreement_4_0_5_C_combined.html)

1. On the VVX or SoundPoint phone’s menu, under **Settings > Status > Network > TCP/IP Parameters**, determine the IP address of the phone. In this example, the address is 10.40.0.71.
2. Open a web browser to visit the IP phone’s configuration page (in our example, this URL is <http://10.40.0.71>. Log into the web interface (as admin) and enter **456** as the password.
3. Select **Utilities > Software upgrade > Server type** and click **Check for update**.
4. Under **Utilities > Software upgrade > Available software versions**.

For a VVX model phone, select **4.1.6.4835**.

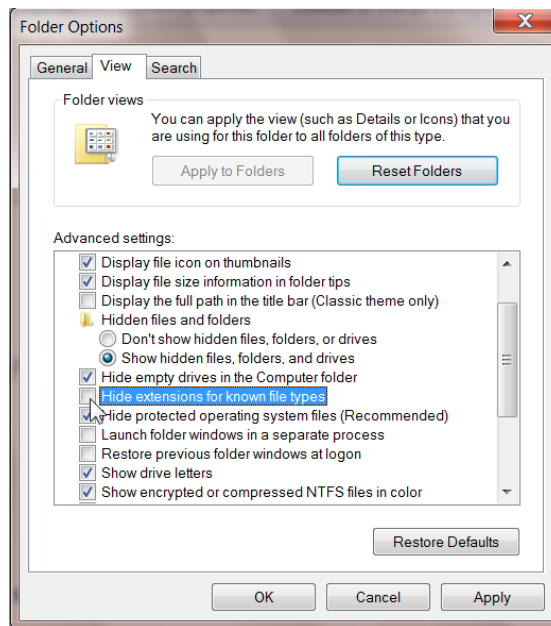


For a SoundPoint phone, select **4.0.4.2906** – DO NOT select version 4.1.0.x.

5. Allow viewing of .cfg files in Windows. Configuration files are in XML format and can be opened in Notepad or any other text editor.

To be able to view and save files with a “.cfg” file extension properly, configure Windows to display all file extensions. (The exact steps to configure this may vary from one version of Windows to another)

In Windows, go to **My Computer > Tools > Folder Options > View Tab**; and uncheck **Hide extensions for known file types**.



## 2.4 Prepare the Configuration Files

This section discusses how to create new configuration files that contain only the settings appropriate for a MaxCS installation.

There are two ways to configure Polycom IP phones:

- By using a configuration file
- By using the browser interface

Because it is more efficient to configure the phones via a configuration file, this guide provides steps for configuring phones via a configuration file.

If this is the new FTP setup, after you set up one new extension, there will be three files in the FTPRoot folder:

- altigen2014.cfg
- 0004f2000607.cfg (mac address)
- 201.cfg (extension)

In other words, only three files are needed for one extension. To add more extensions, you will create corresponding mac address and extension configuration files for each.



If you share one FTP between pre-MaxCS 7.0 Polycom configuration files and MaxCS 7.0 Polycom configuration files, make sure not to overwrite any old files; otherwise, you may break the phones on the older deployment.



Here is the general process; refer to the samples in the next section:

1. Create a SIP configuration file that contains only the parameters specific to MaxCS deployment. For example, `altigen2014.cfg`.
2. Create a local phone file for each phone, containing only the parameters that you want to change. For example, `201.cfg`.
3. Create a master configuration for the phone with a specific Ethernet MAC address. For example, `0004f2000607.cfg`.

### Create the `altigen2014.cfg` Configuration File

The `altigen.cfg` file defines several important parameters for all Polycom IP phone models connecting to the MaxCS system.

To create a new `altigen2014.cfg` file, copy the following configuration lines and save them as “`altigen2014.cfg`.”

```
<?xml version="1.0" standalone="yes"?>
<!-- Altigen specific configuration parameters -->
<sip>
  <voIpProt>
    <SIP>
      <conference voIpProt.SIP.conference.address="conf"/>
      <alertInfo
        voIpProt.SIP.alertInfo.1.value="info=alert-autoanswer"
        voIpProt.SIP.alertInfo.1.class="3"
        voIpProt.SIP.alertInfo.2.value="auto-answer"
        voIpProt.SIP.alertInfo.2.class="3" />

      <altigenport reg.1.server.1.port="10060"
voIpProt.SIP.outboundProxy.port="10060"
        voIpProt.SIP.local.port="10060"/>

    </SIP>
  </voIpProt>

  <TCP_IP>
    <SNTP tcpIpApp.snntp.daylightSavings.enable="1"
tcpIpApp.snntp.gmtOffset="-28800" tcpIpApp.snntp.address="us.pool.ntp.org" />
  </TCP_IP>

  <dialplan dialplan.removeEndOfDial=="0">
    <digitmap dialplan.digitmap="0|##|#[1-3]x|#[5-9]x|#41xx|#4[2-9]| [1-
7]xx|9,11|9,[2-9]xxxxxx|9,1xxxxxxxxxx|xxx.T" />
  </dialplan>
  <call call.lastCallReturnString="#69" />
  <tones>
    <DTMF tone.dtmf.onTime="120" tone.dtmf.offTime="120" />
  </tones>
</sip>
```

```

</tones>

<sound_effects>

    <MESSAGE_WAITING
        se.pat.misc.messageWaiting.inst.1.type="silence"
se.pat.misc.messageWaiting.inst.1.value="0"
        se.pat.misc.messageWaiting.inst.2.type="silence"
se.pat.misc.messageWaiting.inst.2.value="0"
        se.pat.misc.messageWaiting.inst.3.type="silence"
se.pat.misc.messageWaiting.inst.3.value="0"
        se.pat.misc.messageWaiting.inst.4.type="silence"
se.pat.misc.messageWaiting.inst.4.value="0"
    />

</sound_effects>

</sip>

```

- The <SNTP> part defines the time server. The default value is “us.pool.ntp.org”. This can be changed to any SNTP-compliant time server if needed.
- “tcpIpApp.snntp.gmtOffset” is the offset in seconds of the local time zone from GMT, -28800 = -8 \* 3600, so the time zone here is GMT-8 US Pacific time. You can change it to your local time zone.
  - Eastern Time = -18000
  - Central Time = -21600
  - Mountain Time = -25200
  - Pacific Time = -28800
- The “tcpIpApp.snntp.daylightSavings.enable” option enables daylight saving time. It is **enabled** in this configuration. If you don’t use daylight saving time, change the value to “0”.
- **digitmap** dialplan.digitmap="0|##|[1-3]x|[5-9]x|#41xx|#4[2-9]|[[1-7]xx|9,11|9,[2-9]xxxxxx|9,1xxxxxxxxxx|xxx.T "
  - **0** means user can dial 0 to ring operator extension or WG defined in the AltiGen server.
  - **##** means accepting ## to access voice mail.
  - **#[1-3]x** means accepting feature code #1x, #2x, and #3x.
  - **#[5-9]x** means accepting feature code #5x, #6x, #7x, #8x, and #9x.
  - **#41xx** means accepting system call park with two-digit ID.
  - **#4[2-9]** means accepting other feature codes starting with #4

- **[1-7]xx** means your extension length is 3-digit with a prefix starting with 1 to 7. (This means extension 123 is a valid extension number, but 823 or 923 are not valid extension numbers). If your extension digit length is 4, then change **[1-7]xx** to **[1-7]xxx**
  - **9,11** forces a call to be sent immediately if 911 is dialed
  - **9,[2-9]xxxxxx** means dialing trunk access code 9 plus a 7-digit local call. If your local call is 10-digit dialing, you need to change this to **9,[2-9]xxxxxxxxxx**. The comma (,) provides a second dial tone when the caller presses 9.
  - **9,1xxxxxxxxxx** means dialing trunk access code 9 plus an 11-digit long distance number. The comma (,) provides a second dial tone when the caller presses 9.
  - **xxx.T** forces a call to be sent immediately if 3 or more digits are dialed and the “dial” softkey is pressed.
- Note:** If a dialed number is not defined in this dial plan, the user needs to wait for time out or press the softkey **Send** to initiate dialing.
- Note:** If there is a need for extension based digit-map, just copy this section to the extension-specific configuration files. The digit-map setting in extension configuration will override this one.
- **MESSAGE\_WAITING** line will silence the beep tone when there is a voice mail. If you’d like to hear the beep notification of WMI, delete the whole **<sound\_effects>** paragraph.

## Create the Extension-Specific Configuration Files

For each extension that uses a Polycom IP phone, create an extension-specific configuration file that defines the user name, extension number, and password.

The filename format is “xxx.cfg” where the xxx is the extension number, for example 201.cfg.

All ext.cfg files should be saved to the FTP root directory.

To create a new “ext.cfg” file, you can copy the following configuration lines and save this text as a new “ext.cfg” file:

```
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<!-- Extension Registration Configuration File -->
<phone1>
  <reg reg.1.displayName="Ext_Name" reg.1.address="201" reg.1.label="201"
    reg.1.auth.userId="201" reg.1.auth.password="1212"
    reg.1.server.1.address="10.100.2.127" reg.1.server.1.expires="90" />

  <msg msg.bypassInstantMessage="1" msg.mwi.1.callBackMode="contact"
    msg.mwi.1.callBack="##@10.100.2.127"
    msg.mwi.1.subscribe="201@10.100.2.127" />

  <conference voIpProt.SIP.conference.address="conf@10.100.2.127"/>
```

```
</phone1>
```

When creating ext.cfg file for each extension, change the following parameters.

1. Change reg.1.displayName="Ext\_Name" to the extension's user name. reg.1.displayName is the SIP display name used in SIP from header.
2. Change 201 in reg.1.address="201" reg.1.label="201" reg.1.auth.userId="201" to the real extension number. reg.1.lable can be any string that will be displayed on the LCD next to the softkey on a Polycom phone. If the string is too long it will be truncated on the LCD of a Polycom phone.
3. Reg.1.auth.password is a fake password. The real password needs to be set in the Polycom phone's web interface for security. This will be mentioned later in this document.
4. Change the server addresses in reg.1.server.1.address="10.100.2.127" to the real MAXCS server address.
5. Change the message waiting indicator settings:
  - Change the server IP address in msg.mwi.1.callBack="##@10.100.2.127"
  - Change the extension number and server IP address in msg.mwi.1.subscribe=201@10.100.2.127
  - Change the server IP address in volpProt.SIP.conference.address=conf@10.100.2.127.

## Create the Phone-Specific Configuration File – *MAC\_Address.cfg*

Check the phone's Ethernet MAC address. You can find it on the back of the phone.

Create a configuration file with the MAC address as its file name, [MAC\_address].cfg. For example, 0004f214ab37.cfg.

Copy the following configuration lines and save the text to the FTP root directory.

```
<?xml version="1.0" standalone="yes"?>
<!-- Default Master SIP Configuration File-->
<!-- Edit and rename this file to [Ethernet-address].cfg for each phone.-->
<!-- $RCSfile: 000000000000.cfg,v $ $Revision: 1.18 $ -->
<APPLICATION APP_FILE_PATH="sip.ld" CONFIG_FILES="201.cfg, altigen2014.cfg"
  MISC_FILES="" LOG_FILE_DIRECTORY="" OVERRIDES_DIRECTORY="" CON-
  TACTS_DIRECTORY="" LICENSE_DIRECTORY="">

  <APPLICATION_SPIP300 APP_FILE_PATH_SPIP300="sip_212.ld"
    CONFIG_FILES_SPIP300="phone1_212.cfg, sip_212.cfg"/>
  <APPLICATION_SPIP500 APP_FILE_PATH_SPIP500="sip_212.ld"
    CONFIG_FILES_SPIP500="phone1_212.cfg, sip_212.cfg"/>

</APPLICATION>
```

When creating the MAC\_Address.cfg file for each phone, change the following parameter.

- Change “201.cfg” in line 5 to the file name of [extension number].cfg.

## 2.5 Configure the FTP and Password for Each Phone

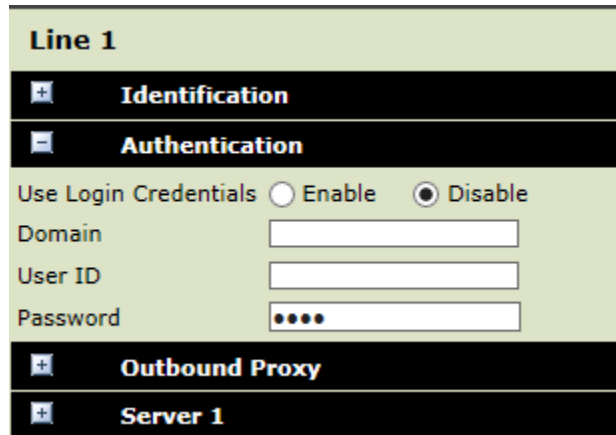


It is important to change *only* the parameters we specify here. All other configurations will be downloaded from the FTP server. If you change any other parameters, you will override the configuration from FTP server. This may prevent the phone from registering properly.

For example, if you change the *User ID* in the web interface, it will override the FTP’s setting. This may prevent the phone from registering properly.

If you accidentally change other parameters and inadvertently compromise a configuration, read [Troubleshooting: Reset configuration to Factory Default](#) on page 15 for instructions on how to reset the configuration to the factory default.

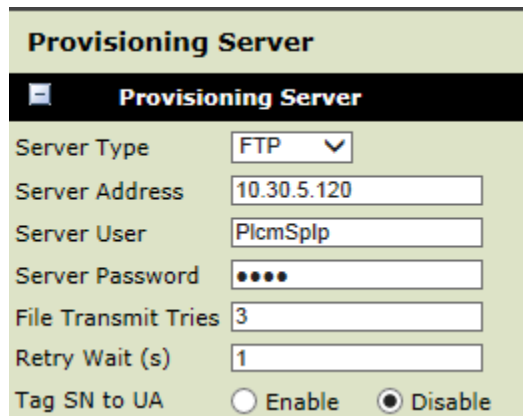
1. Log into the Polycom Web Interface again, as described in [Update Firmware and the Default Configuration Files](#) on page 7.
2. Select **Settings > Lines > Line1**.  
Set the *Password* here to the same password that you set in the *SIP Registration Password* field in MaxAdministrator.



3. Select **Settings > Provisioning Server > Provisioning Server**.

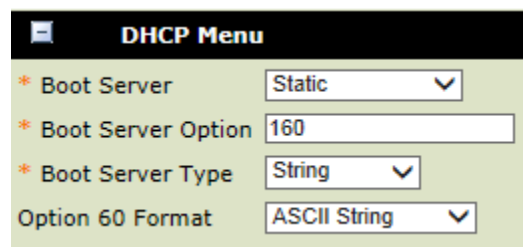
Set the *Server Type* to **FTP**.

Set the *Server Address* to the address of FileZilla FTP server. This server can reside on the same MaxCS system.



4. Select **Settings > Provisioning Server > DHCP Menu**.

Set *Boot Server* to **Static**.



5. Apply these settings. Afterward, the phone should reboot automatically. If it does not, select **Utilities > Reboot Phone**, or unplug the phone and plug it back in again.

Check the FTP server to see if the configuration is downloading to the Polycom phone.

## 2.6 Set the Codec Profile to DTMF Delivery

1. In Enterprise Manager, click the **Codec** button.
2. In the Codec list, choose the **Default** codec profile

3. Set *DTMF Delivery* to **RFC 2833**.
4. Set *SIP Early Media* to **Disable**.

G.711 RTP Packet Length (ms)	20
G.729 RTP Packet Length (ms)	20
DTMF Delivery	RFC 2833
SIP Early Media	Disable
SIP Transport	UDP

5. After you change the default codec setting, if MaxCS has SIP trunk connections to a SIP gateway, then make sure they have a dedicated codec profile. On the IP Codec tab, assign the profile in the IP Device Range section.

Information	IP Networks	IP Dialing Table	IP Codec	Number Plan
Inter Gateway Codec: G711 Only				
IP Device Range				
From	To	Codec		
10.40.0.81	10.40.0.81	polycom		Local
10.40.1.11	10.40.1.11	bandwidth		Local
216.115.20.41	216.115.20.41	vonage		NA
192.168.0.0	192.168.255.255	G711 Only		NA
172.16.0.0	172.31.255.255	G711 Only		NA
10.0.0.0	10.255.255.255	G711 Only		Local

6. Review the IP Dialing Table's codec setting. If necessary, create a dedicated Codec Profile:

Information	IP Networks	IP Dialing Table	IP Codec	Number Plan			
ID	Type	Name	Address	Ext Len	Dial Scheme	Protocol	Codec
00	Local	10.40.1.43	10.40.1.43	none	Overlapping(ATGN)	SIP	Default
01	Local	10.40.1.99	10.40.1.99	none	Enblock	SIP	lync
02	Local	salesman	10.20.0.50	none	Enblock	SIP	G711 ...
03	Local	heavy	10.20.0.60	none	Overlapping(ATGN)	SIP	G711 ...

## 2.7 Troubleshooting: Reset Configuration to Factory Default

If the Polycom IP phone you try to deploy has been configured before and you are not able to reconfigure it to work with the MaxCS system, you need to reset the phone's settings to the factory defaults.

1. To reset the phone, press the **Menu** key on the phone to display the phone menu.
2. Select **Settings > Advanced** and enter the default password **456**.
3. Select **Admin Settings > Reset to Defaults > Reset To Factory> Yes**.

**Note:** If phone's firmware is 4.x.x or above, this can also be done through Polycom Web Interface; select **Utilities > Phone Backup & Restore > Global Settings > Restore phone to factory settings**.



## 2.8 Advanced Phone Features Not Supported in Polycom IP phones

Feature	Description
Display extension DND/FWD status	Polycom IP phones cannot display DND and FWD (Forward) status when a user enables DND or FWD using a feature code or through a client application.
Display Activity (Presence)	Activity status is an AltiGen proprietary feature. Polycom IP phones cannot display this information.
Extension override by another IP phone or IPTalk	While a Polycom IP phone is registered to the MAXCS server, another IP phone or IPTalk <b>cannot</b> register to the same extension number to override the phone.
IP Paging	Polycom IP phones cannot accept IP paging calls.

## 2.9 Limitations When Using Polycom IP Phones with MaxCS 7.0

Feature	Description
Call handling	<p>You are not able to Unhold, Transfer, or Conference another party from MaxCommunicator/MaxAgent when Hold, Transfer, or Conference is initiated from a third-party IP phone. You must complete the operation from that phone.</p> <p>Likewise, you cannot hold a call from a client application and try to Unhold the call from the third-party IP phone.</p>



Feature	Description
Cannot return PSTN call from IP phone call log	Polycom IP phones cannot insert the trunk access code to the call back number.
Cannot enter Account Code (Flash#32)	Polycom IP phones do not support the FLASH key.
System and Station Call Park (Flash#31 and Flash#41)	Polycom IP phones do not support the FLASH key.
Concurrent use of extension number	When you associate an extension number with a third-party phone, you cannot concurrently use the extension number with another third-party IP phone, AltiGen IP phone, mobile extension, or IPTalk without powering down the active third-party phone.

### 3 AltiGen Technical Support

AltiGen provides technical support to Authorized AltiGen Partners and distributors only. End user customers, please contact your Authorized AltiGen Partner for technical support.

**Authorized AltiGen Partners** and distributors may contact AltiGen technical support by the following methods:

- You may request technical support on AltiGen’s Partner web site, at <https://partner.altigen.com>. Open a case on this site. A Technical Support representative will respond within one business day.
- Call 888-ALTIGEN, option 5, or 408-597-9000, option 5, and follow the prompts. Your call will be answered by one of AltiGen’s Technical Support Representatives or routed to the Technical Support Message Center if no one is available to answer your call.

Technical support hours are 5:00 a.m. to 5:00 p.m., PT, Monday through Friday, except holidays.

If all representatives are busy, your call will be returned in the order it was received, within four hours under normal circumstances. Outside AltiGen business hours, only urgent calls will be returned on the same day (within one hour). Non-urgent calls will be returned on the next business day.

Please be ready to supply the following information:

- Partner ID
- AltiGen Certified Engineer ID
- Product serial number
- AltiWare or MAXCS version number
- Number and types of boards in the system
- Server model
- The telephone number where you can be reached