MITEL – SIPCoE

Technical
Configuration Notes

Configure MCD 4.1 for use with the Polycom SoundStation IP7000

SIP CoE 08-5159-00020

MITEL
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NOTICE

Mitel Technical Configuration Notes – Configure MCD 4.1for use with the Polycom SoundStation IP 7000 phone
October 2012, 08-5159-00020_5

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel 3300 ICP to host Polycom SoundStation IP 7000. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

Interop History

Version	Date	Reason
1	August 25. 2008	Initial Interop with Mitel 3300 9.0 and Polycom SoundStation IP 7000 SIP 3.0.2.0580
2	July 21, 2009	Documentation Update
3	August 19, 2009	Refresh Interop with Mitel 10.0.0.10_2 and Polycom SoundStation IP 7000 SIP 3.1.3.0439
4	April 15, 2010	Interop with Mitel 3300 10.1.0.69_1 and Polycom SoundStation IP 7000 SIP 3.2.2.0477
5	October 23, 2012	Added System Based Conferencing procedure

Interop Status

The Interop of Polycom SoundStation IP 7000 has been given a Certification status. This device will be included in the SIP CoE Reference Guide. The status the Polycom SoundStation IP 7000 achieved is:

COMPATIBLE	The most common certification which means the device/service has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between the Polycom SoundStation IP 7000 device and the 3300 ICP.

Manufacturer	Variant	Software Version
Mitel	3300 ICP – Mxe Platform	10.1.0.69_1
Mitel	MBG - Teleworker	5.2.9.0
Mitel	5340, 5212 SIP Phones	R8.0.01.06.01.02
Mitel	5340, 5215 IP Phones	01.06.01.02
Polycom	SoundStation IP 7000	3.2.2.0477

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Line Side Interoperability Test Pans for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call	☑
DTMF Signal	Sending DTMF after call setup (i.e. mailbox password)	☑
Call Hold	Putting a call on hold	☑
Call Transfer	Transferring a call to another destination	☑
Call Forward	Forwarding a call to another destination	⚠
Conference	Conferencing multiple calls together	☑
Redial	Last Number Redial	☑
MWI	Message Waiting Indication	☑
Dynamic Extension	Personal Ring Group configuration	☑
Resiliency	Basic calls through a Secondary SIP proxy	⚠
T.38 Fax	Fax Messages	N/S
Video	Video Capabilities	N/S
Teleworker	Mitel remote connectivity with Teleworker	☑

☑ - No issues found ✗ - Issues found, cannot recommend to use ⚠ - Issues found

Resiliency

The following table lists the scenarios of resilience supported by this device when connected to the MCD 4.1 on the 3300 ICP.

Device	Scenario 1	Scenario 2	Scenario 3	Scenario 4
Polycom SoundStation IP 7000	⚠	⚠	Not Supported	Not Supported

☑ - No issues found ✗ - Issues found, cannot recommend use ⚠ - Issues found

Note: Refer to list of device limitations and known issues later in the document for recommendations.

The various scenarios are described below. The scenario names are a convenience for understanding this section of the configuration guide.

Scenario 1: Resiliency is achieved by utilizing the ability of DNS servers to provide multiple IP addresses against a single FQDN. This is generally achieved by using DNS SRV or A records. This scenario requires nothing from a SIP Endpoint except that it supports standard DNS behaviour
NOTE: Polycom SoundStation IP 7000 supports both methods SRV and A-record.

Scenario 2: The device has inherent knowledge of the primary and secondary 3300 ICPs and will switch between them if a SIP request (**REGISTER**, **INVITE**, or **SUBSCRIBE**) times out. Behaviour will be characterized based on whether the device returns to primary ICP and when this occurs. This scenario has some dependency on user action in order to detect a failure, especially if configured with a long registration expiry time, so the chance of a user experiencing a long delay making a call goes up.

Scenario 3: The behaviour of the device is the same as that of scenario 2, except that the device will “ping" the currently active server with an **OPTIONS** request. If the **OPTIONS** request times out, the device will switch to the alternate server for all future requests. The intent of this scenario is to provide much faster failure detection by the device. This will allow devices to failover to their alternate ICP much more quickly, and much more unnoticeably. (If the device can detect a failure of the primary ICP, and can failover immediately, the chance that the user even notices a lack of service falls dramatically.)

Scenario 4: The device will support a new SIP header designed specifically for resiliency. The *P-Alternate-Server* header must be included in a **200 OK** or **301 Moved Permanently** response. This header will include data that designates the potential servers and which server the UA must use.

Device Limitations

This is a list of problems or not supported features when the Polycom SoundStation IP 7000 device is connected to the Mitel 3300.

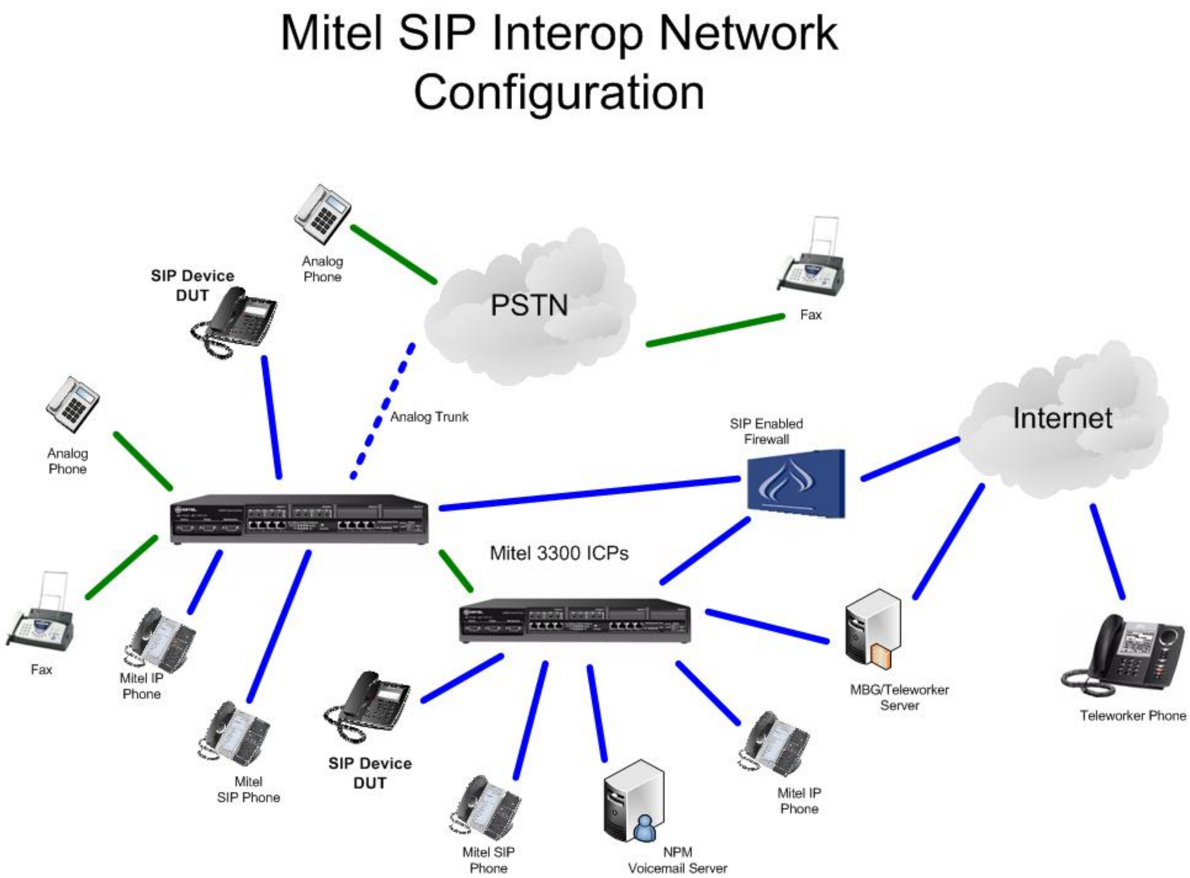
Feature	Problem Description
Call Forward	Even though Call Forward (e.g. on No Answer or Always) is enabled thorough the web interface, it remains inactive. **Recommendation:** Activate transfer manually on the phone when it is ringing or contact Polycom for support of this feature.
Conference	The Polycom 7000 is limited to initiate a 3 party conference and is unable to add a 4th party. **Recommendation:** To add 4 or more parties to a conference, the conferencing capabilities of the MCD will need to be used. Change to System Based incall features in the SIP Device Capabilities form. See Appendix A for instructions
Resiliency	**Scenario 1**: When registered on a secondary 3300, Polycom SoundStation IP 7000 can not receive the calls **Recommendation:** none **Scenario 2**: Polycom SoundStation IP 7000 sticks to the primary PBX and does not try to register on a secondary (when the primary is unavailable). **Recommendation:** The phone reboot is required

Device Recommendations

The Polycom SoundStation IP 7000 is recommended for deployment with Device Based In-Call Features enabled. See Sip Device Capabilities form below for more information. Although if more than 3 party conferences are required, then leave SYstem based in-call features enabled.

Network Topology

This diagram shows how the testing network is configured for reference.



Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how the Polycom SoundStation IP 7000 was configured in our test environment.

We recommend that the Polycom SoundStation IP 7000 is configured in Device Mode. You will configure the Device mode in the SIP Device Capabilities Form as described in this section.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

3300 ICP Configuration Notes

The following steps show how to program a 3300 ICP to connect with the Polycom SoundStation IP 7000.

Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the 3300 Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

Assumptions for the 3300 ICP Programming

- The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the 3300 ICP is equipped with enough SIP Device licenses for the connection of SIP end points. This can be verified within the License and Option Selection form.

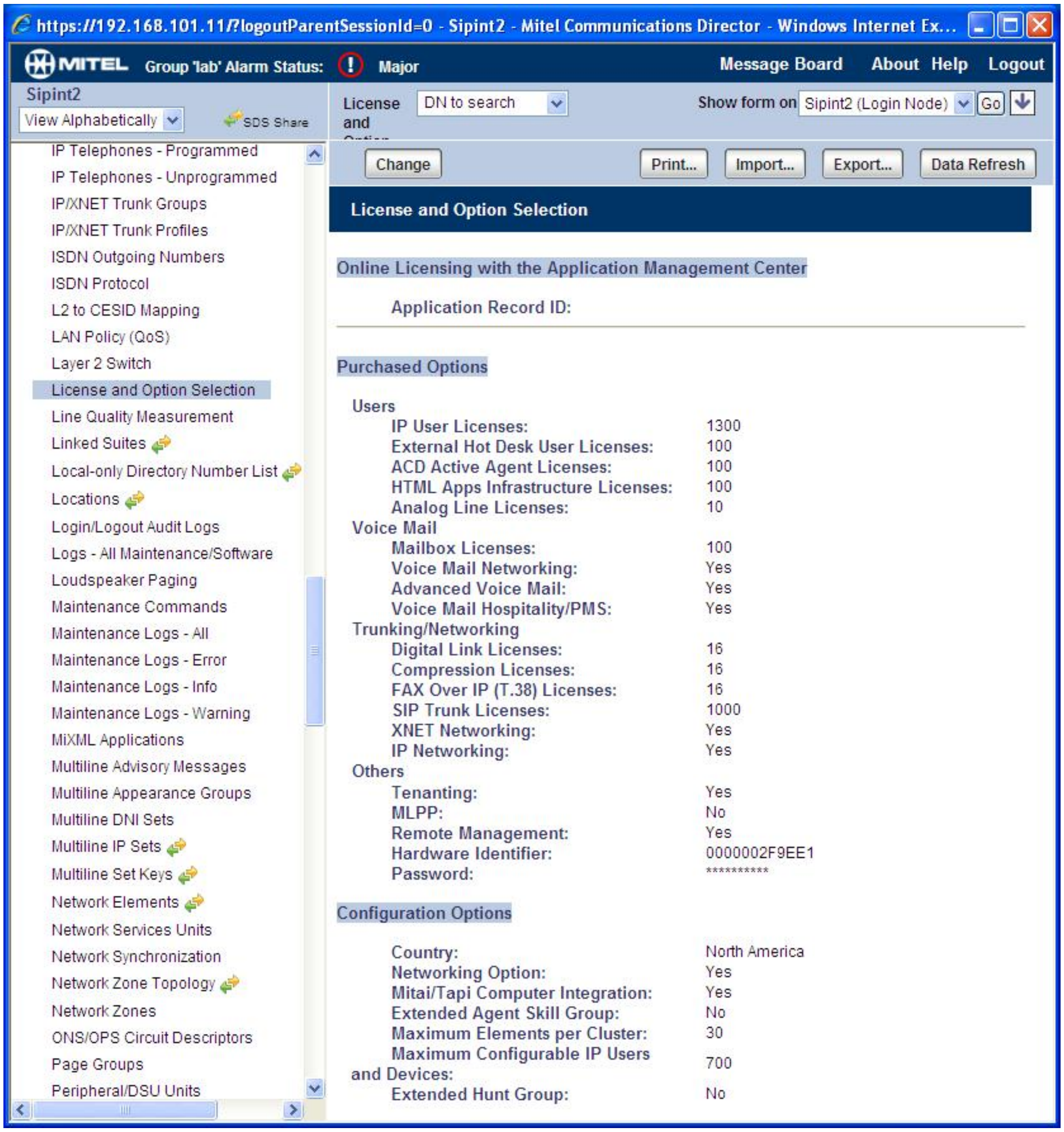


Figure 1 – License and Option Selection form

Multiline IP Set Configuration

On the Mitel 3300 ICP, a SIP device type can be programmed either in the User and Device Configuration form or the Multiline IP Sets form and it should be programmed as a “Generic SIP Phone”. Enterprise Manager can also be used to provision where this application is installed.

The User PIN is the SIP authentication password and the Number is the Directory Number (DN a telephone number). The Number and User PIN must match the information in the Polycom SoundStation IP 7000 configuration file (phone1_<MAC address>.cfg). All other field names should be programmed according to the site requirements or left at default.

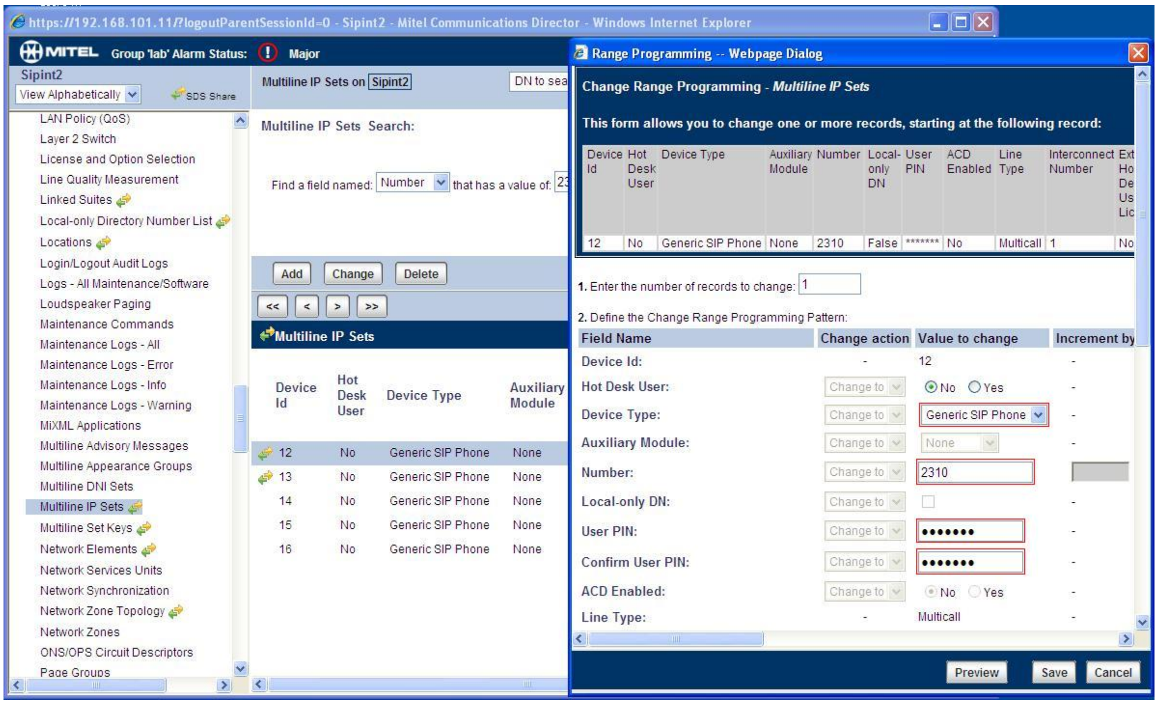


Figure 2 – Multiline IP Sets form

Class of Service Assignment

The Class of Service Options Assignment form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced by the Station Attributes form for the SIP devices.

Many different options may be required for your site deployment, but these are the options that are required to be changed from the default for a Generic SIP Device to work with the 3300 ICP.

- Conference Call set to **Yes**
- HCI/CTI/TAPI Call Control Allowed set to **Yes**
- HCI/CTI/TAPI Monitor Allowed set to **Yes**
- Message Waiting set to **Yes**
- Public Network Access via DPNSS set to **Yes**
- Auto Campon Timer is **blanked (no value)**

Class of Service Options

Class Of Service Number:	4	
Comment:	SIP Sets	
Account Code Verified:	◉ No	○ Yes
ACD Make Busy on Login:	◉ No	○ Yes
ACD Silent Monitor Accept:	◉ No	○ Yes
ACD Silent Monitor Allowed:	◉ No	○ Yes
ACD Silent Monitor Notification:	◉ No	○ Yes
Allow Directed Call Pickup Of Attendant Call:	○ No	◉ Yes
ANI/DNIS/ISDN Number Delivery Trunk:	○ No	◉ Yes
Auto Answer Allowed:	○ No	◉ Yes
Brokers Call:	◉ No	○ Yes
Busy Overide Security:	◉ No	○ Yes
Call Announce Line:	◉ No	○ Yes
Call Forwarding Accept:	○ No	◉ Yes
Call Forwarding (External Destination):	○ No	◉ Yes
Call Forwarding (Internal Destination):	○ No	◉ Yes
Call Forward Override:	◉ No	○ Yes
Call Forwarding Reminder Ring (CFFM and CFIAH only):	◉ No	○ Yes
Call Hold:	○ No	◉ Yes
Call Hold Remote Retrieve:	○ No	◉ Yes
Call Hold - Retrieve with Hold Key:	○ No	◉ Yes
Call Park-Allowed To Park:	○ No	◉ Yes
Call Pickup Dialed Accept:	○ No	◉ Yes
Call Pickup Directed Accept:	○ No	◉ Yes
Call Privacy:	◉ No	○ Yes
Call Reroute after CFFM to Busy Destination:	◉ No	○ Yes
Call Waiting Swap:	◉ No	○ Yes
Called Party Features Override:	◉ No	○ Yes
Calling Name Display - Internal - ONS:	○ No	◉ Yes
Calling Number Display - Internal - ONS:	○ No	◉ Yes
Calling Party Name Substitution:	◉ No	○ Yes
Campon Tone Security / FAX Machine:	◉ No	○ Yes
Check COR after PSTN Dial Tone:	◉ No	○ Yes
Clear All Features Remote:	◉ No	○ Yes
Conference Call:	○ No	◉ Yes
COV/ONS/E&M Voice Mail Port:	◉ No	○ Yes
DASS II OLI/TLI Provided:	◉ No	○ Yes
Dialled Night Service:	○ No	◉ Yes
Direct Voice Call - Accept:	◉ No	○ Yes
Direct Voice Call - Allow:	◉ No	○ Yes
Direct Voice Call - Maximize Volume:	◉ No	○ Yes
Disable Call Reroute Chaining On Diversion:	◉ No	○ Yes
Disable Conference Join Tone:	◉ No	○ Yes
Disable Executive Busy Override Tone:	◉ No	○ Yes
Disable Send Message:	◉ No	○ Yes

Save Cancel

Disable Executive Busy Override Tone:	◉ No	○ Yes
Disable Send Message:	◉ No	○ Yes
Display ANI/ISDN Calling Number Only:	◉ No	○ Yes
Display ANI/DNIS/ISDN Calling/Called Number:	○ No	◉ Yes
Display Caller ID on multicall/keylines:	◉ No	○ Yes
Display DNIS/Called Number Before Digit Modification:	◉ No	○ Yes
Display Dialed Digits during Outgoing Calls:	◉ No	○ Yes
Display Held Call ID on Transfer:	◉ No	○ Yes
Display Transfer Destination on Recall:	◉ No	○ Yes
Do Not Disturb:	○ No	◉ Yes
Do Not Disturb - Access to Remote Phones:	○ No	◉ Yes
Do Not Disturb Permanent:	◉ No	○ Yes
Emergency Call Notification - Audio:	◉ No	○ Yes
Emergency Call Notification - Visual:	◉ No	○ Yes
Enable Call Duration Limit on External Calls:	◉ No	○ Yes
Enable Call Duration Limit on Internal Calls:	◉ No	○ Yes
Executive Busy Override:	◉ No	○ Yes
External Trunk Standard Ringback:	◉ No	○ Yes
Flexible Answer Point:	◉ No	○ Yes
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent:	◉ No	○ Yes
Forced Verified Account Code:	◉ No	○ Yes
Forced Non-Verified Account Code:	◉ No	○ Yes
Group Call Forward Follow Me Accept:	◉ No	○ Yes
Group Call Forward Follow Me Allow:	◉ No	○ Yes
Group Page Accept:	◉ No	○ Yes
Group Page Allow:	◉ No	○ Yes
Group Presence Control:	◉ No	○ Yes
Group Presence Third Party Control:	◉ No	○ Yes
Handset Volume Adjustment Saved:	◉ No	○ Yes
Handsfree AnswerBack Allowed:	◉ No	○ Yes
HCI/CTI/TAPI Call Control Allowed:	○ No	◉ Yes
HCI/CTI/TAPI Monitor Allowed:	○ No	◉ Yes
Head Set Switch Mute:	◉ No	○ Yes
Hot Desk External User - Answer Confirmation:	○ No	◉ Yes
Hot Desk External User - Display Internal Calling ID:	◉ No	○ Yes
Hot Desk External User - Permanent Login:	◉ No	○ Yes
Hot Desk Login Accept:	◉ No	○ Yes
Hot Desk Remote Logout Enabled:	◉ No	○ Yes
Hotel Room Monitor Setup Allowed:	◉ No	○ Yes
Hotel Room Monitoring Allowed:	◉ No	○ Yes
Hotel/Motel Room Personal Wakeup Call Allowed:	◉ No	○ Yes
Hotel/Motel Room Remote Wakeup Call Allowed:	◉ No	○ Yes
Individual Trunk Access:	○ No	◉ Yes
Local Music On Hold source:	◉ No	○ Yes
Loudspeaker Pager Override:	○ No	◉ Yes

Save Cancel

Local Music On Hold source:	◉ No	○ Yes
Loudspeaker Pager Override:	○ No	◉ Yes
Loudspeaker Pager Equivalent Zone Override Security:	◉ No	○ Yes
Maintain Ringing Party During Recall:	◉ No	○ Yes
Message Waiting:	○ No	◉ Yes
Message Waiting Audible Tone Notification:	◉ No	○ Yes
Message Waiting Deactivate On Off-Hook:	○ No	◉ Yes
Message Waiting - Disable Ringing Lamp Notification:	◉ No	○ Yes
Message Waiting Inquire:	○ No	◉ Yes
Multiline Set Loop Test:	◉ No	○ Yes
Multiline Set Message Center Remote Read Allowed:	◉ No	○ Yes
Multiline Set Music:	◉ No	○ Yes
Multiline Set On-hook Dialing:	○ No	◉ Yes
Multiline Set Phonebook Allowed:	○ No	◉ Yes
Multiline Set Voice Mail Callback Message Erasure Allowed:	◉ No	○ Yes
Music on Hold on Transfer:	◉ No	○ Yes
Name Suppression on outgoing Trunk Call:	◉ No	○ Yes
Non DID Extension:	◉ No	○ Yes
Non-Prime Public Network Identity:	◉ No	○ Yes
Non Verified Account Code:	○ No	◉ Yes
Off-Hook Voice Announce Allowed:	◉ No	○ Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate:	◉ No	○ Yes
ONS CLASS/CLIP: Set:	◉ No	○ Yes
ONS CLASS/CLIP: Visual Call Waiting:	○ No	◉ Yes
ONS/OPS Internal Ring Cadence for External Callers:	◉ No	○ Yes
Originator's Display Update In Call Forwarding/Rerouting:	◉ No	○ Yes
Override Interconnect Restriction on Transfer:	◉ No	○ Yes
Pager Access All Zones:	○ No	◉ Yes
Pager Access Individual Zones:	◉ No	○ Yes
PC Port On IP Device - Disable:	◉ No	○ Yes
Phonebook Lookup - Default to User Location:	◉ No	○ Yes
Phonebook Lookup - Display User Location:	◉ No	○ Yes
Phone Lock:	◉ No	○ Yes
Privacy Released:	◉ No	○ Yes
Public Network Access via DPNSS:	○ No	◉ Yes
Public Network Identity Provided:	◉ No	○ Yes
Public Network To Public Network Connection Allowed:	◉ No	○ Yes
Public Trunk:	◉ No	○ Yes
R2 Call Progress Tone:	◉ No	○ Yes
Recall If Transferred to Original Call Destination:	◉ No	○ Yes
Record-A-Call Active:	◉ No	○ Yes
Record-A-Call - Start Automatic Incoming Call Recording:	◉ No	○ Yes

Save Cancel

Record-A-Call - Start Automatic Incoming Call Recording:	◉ No	○ Yes
Record-A-Call - Start Automatic Outgoing External Call Recording:	◉ No	○ Yes
Record-A-Call - Save Recording on Hang-up:	◉ No	○ Yes
Recorded Announcement Device:	◉ No	○ Yes
Recorded Announcement Device - Advanced:	◉ No	○ Yes
Redial Facilities:	○ No	◉ Yes
Return Disconnect Tone When Far End Party Clears:	◉ No	○ Yes
Ringing Line Select:	◉ No	○ Yes
SC1000 Attendant Basic Function Key:	◉ No	○ Yes
SMDR External:	◉ No	○ Yes
SMDR Internal:	◉ No	○ Yes
Speak@Ease Preferred:	◉ No	○ Yes
Suppress Delivery of Caller ID Display between Sets:	◉ No	○ Yes
Suppress Delivery of Caller ID Display between Sets - Override:	◉ No	○ Yes
Suppress Display Of Account Code Numbers:	◉ No	○ Yes
Suppress Redial Display:	◉ No	○ Yes
Suppress Simulated CCM after ISDN Progress:	◉ No	○ Yes
Third Party Call Forward Follow Me Accept:	◉ No	○ Yes
Third Party Call Forward Follow Me Allow:	◉ No	○ Yes
Timed Reminder Allowed:	○ No	◉ Yes
Trunk Calling Party Identification:	○ No	◉ Yes
Trunk Flash Allowed:	◉ No	○ Yes
Two B-Channel Transfer Allowed:	◉ No	○ Yes
Use Held Party Device for Call Re-routing:	○ No	◉ Yes
Use Called Party Call Hold Timer:	◉ No	○ Yes
Voice Mail Softkey:	◉ No	○ Yes

Account Code Length:	12
After Answer Display Time:	
Answer Plus Delay To Message Timer:	20
Answer Plus Expected Off-hook Timer:	30
Answer Plus Message Length Timer:	10
Answer Plus System Reroute Timer:	0
Attendant Busy Out Timer:	10
Auto Campon Timer:	
Busy Tone Timer:	30
Call Duration:	10
Call Duration Forced Cleardown Timer:	0
Call Forward - Delay:	0
Call Forward No Answer Timer:	15
Call Hold Timer:	30
Call Park Timer:	180
Campon Recall Timer:	10

Save Cancel

Figure 3 – Class of Service Options form

SIP Device Capabilities Assignment

This form provides configuration options that can be applied to various types of SIP devices. The association between the SIP device and the form is similar to how the Class of Service options work. The SIP Device Capabilities number provides a SIP profile that can be applied to particular SIP devices to allow for alternate capabilities as recommended through the Mitel interop process.

In the SIP Device Capabilities form, program a SIP Device Capabilities Number for Polycom SoundStation IP 7000 device. Ensure that “Replace System based with Device based In-Call Feature” is set to ‘**Yes**’. Although if more than 3 party conference is required, then leave this option set to ‘**No**’.

NOTE: Ensure that option “**Prevent the Use of IP Address 0.0.0.0 in SDP Messages**” is set to “**Yes**” (see the screenshot below). Otherwise Music-On-Hold is not played on Polycom SoundStation IP 7000.

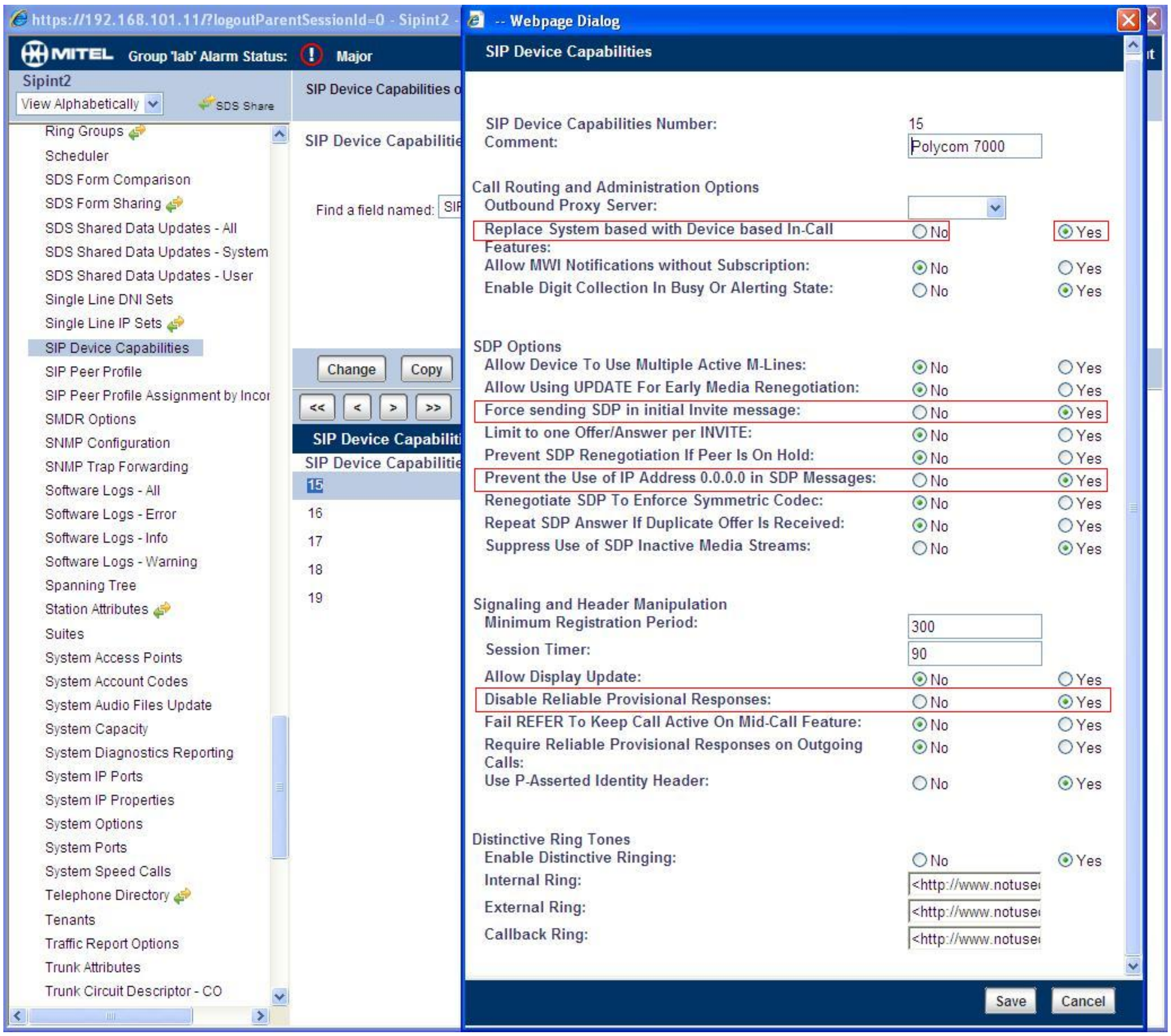


Figure 4 – SIP Device Capabilities form

Station Attributes Assignment

Use the Station Service Assignment form to assign the previously configured Class of Service and SIP Device Capability number to each of the Polycom SoundStation IP 7000 in the 3300. This form utilizes Range Programming.

Select the Polycom SoundStation IP 7000 device number then select Change. Enter the previously configured SIP Device Capability number and Class of Service for Day, Night 1 & Night 2.

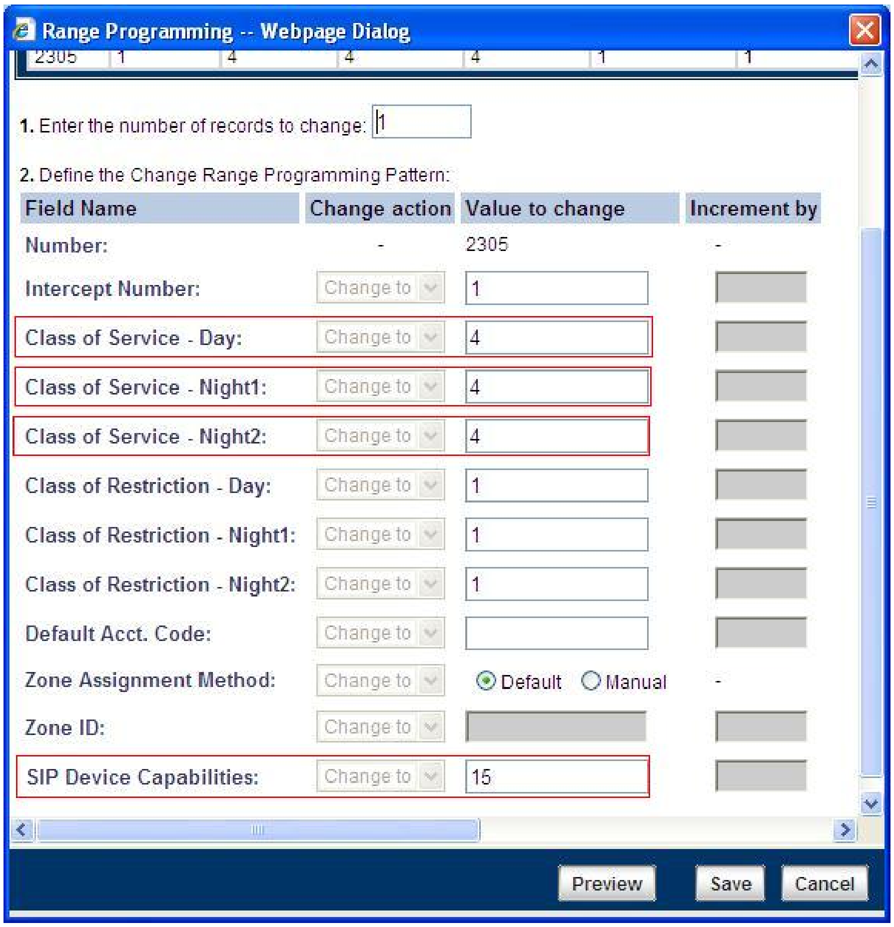


Figure 5 – Station Attributes form

Multiline Set Keys

You use the Multiline Set Keys form to assign the line type, ring type, and directory number to each line selected on the Polycom SoundStation IP 7000 device. For the tests, only 2 calls per line were programmed.

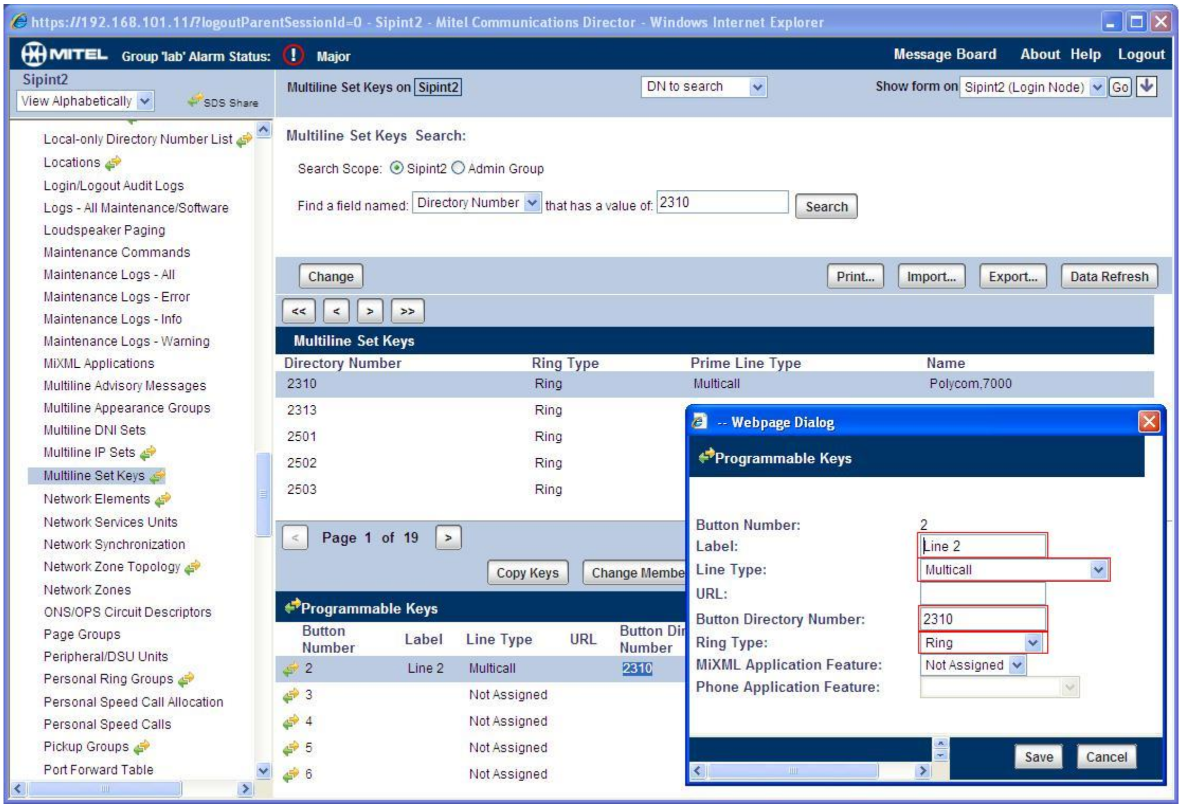


Figure 6 – Multiline Set Key Assignment form

Call Rerouting Assignment

Mitel recommends that call forwarding be programmed using the Call rerouting forms of the 3300. Call forwarding programmed from the Polycom SoundStation IP 7000 has also been tested but we suggest that administrators use Call Rerouting.

Call Rerouting is configured at the system to allow for extensions to forward on different conditions to different extensions, i.e. forward to voicemail when no answer. The following is a description how to configure call rerouting and does not necessarily show how this Polycom SoundStation IP 7000 was programmed.

Program the Call Rerouting First Alternative form with the destination of the call forwarding and the options (Normal, This, Last). Please see the 3300 help files for more info.

There is also a Call Rerouting Second Alternative Assignment form for more complicated forwarding needs.

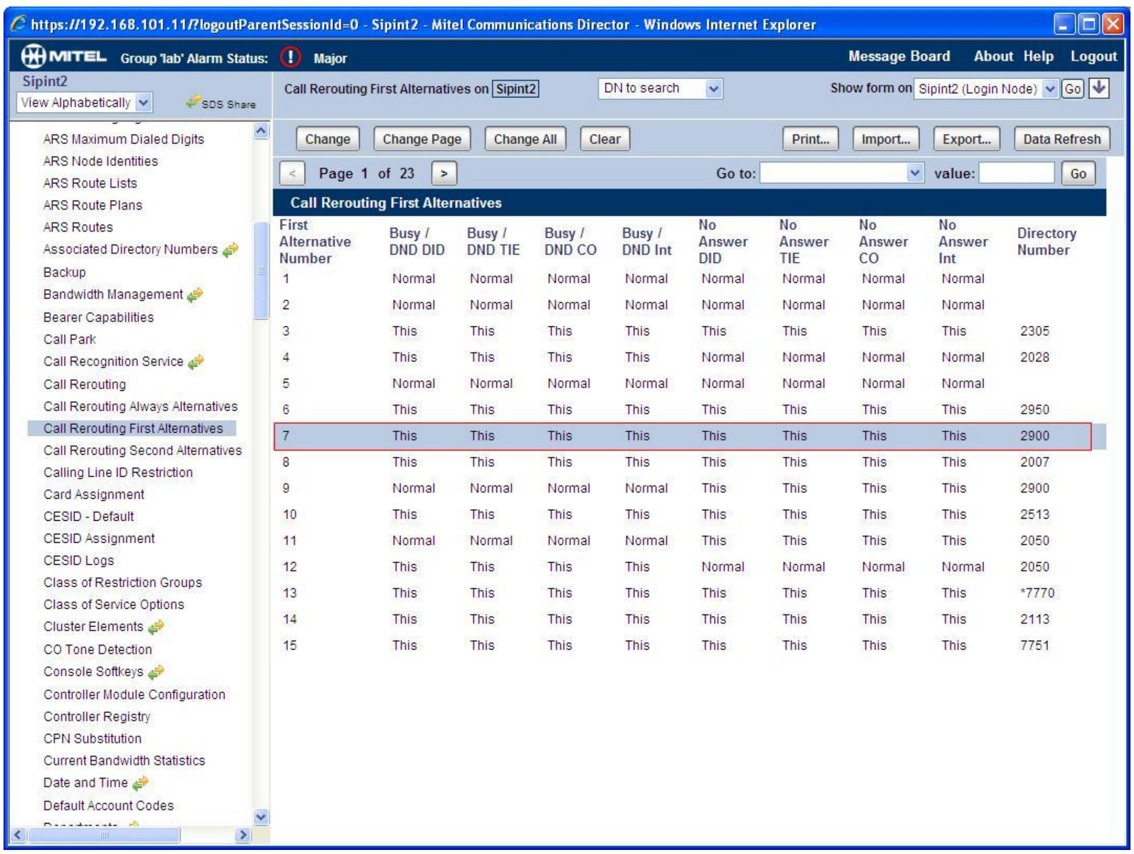


Figure 7 – Call Rerouting First Alternative Assignment

If any Call Forwarding Always were required then the Call Rerouting Always Alternative form would need to be programmed.

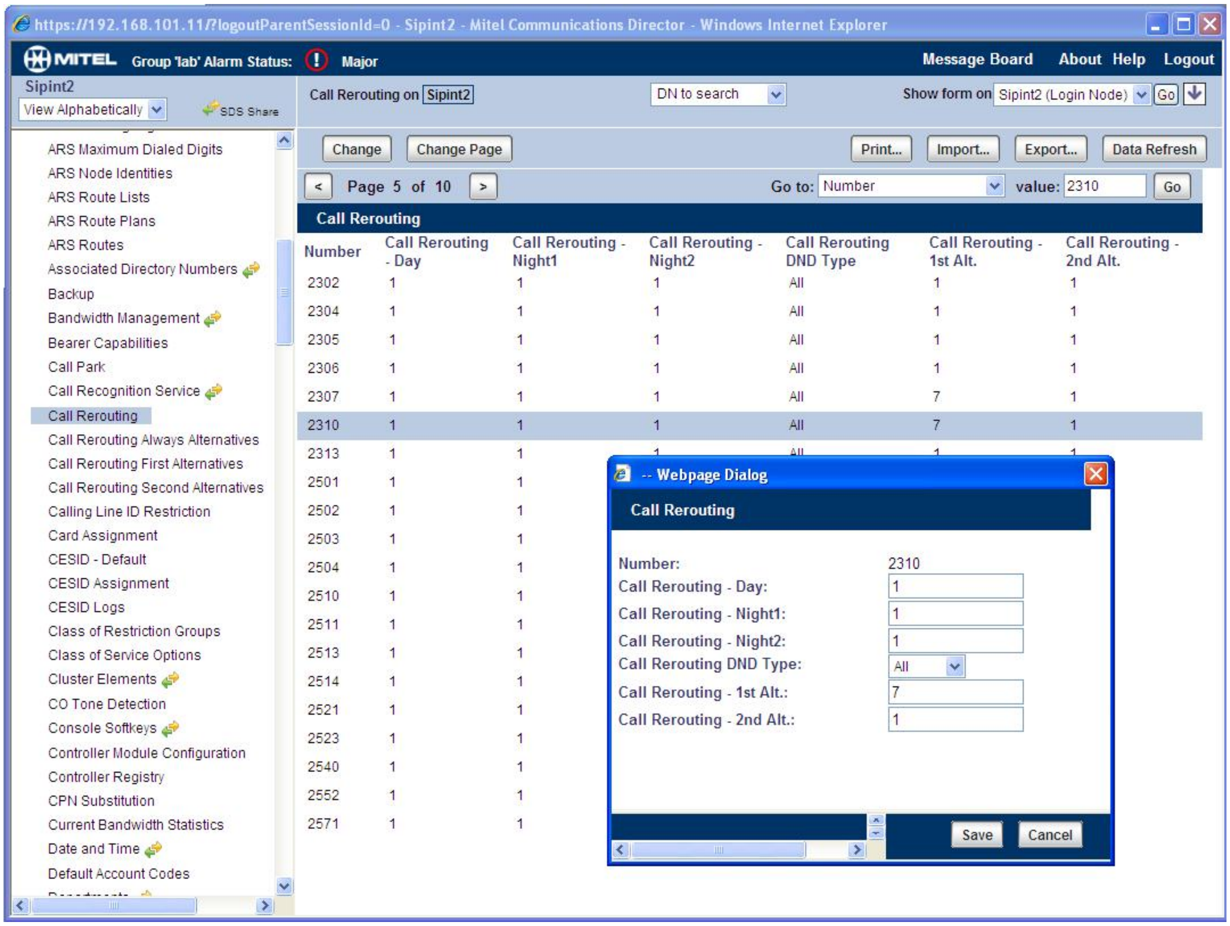


Figure 8 – Call Rerouting Assignment form

Use the Alternative Numbers from the previous forms and fill out the Call Rerouting form for the Polycom SoundStation IP 7000 programmed extension.

Polycom SoundStation IP 7000 Setup Notes

The following steps show how to program the Polycom SoundStation IP 7000 phone to interconnect with the 3300 ICP

The detailed instructions and explanations of the configuration settings for Polycom SoundStation IP 7000 could be found in Administrator's guide at Polycom's web site:

http://www.polycom.com/support/voice/soundstation_ip_series/soundstation_ip7000.html

There are two ways to configure Polycom SoundStation IP 7000: either to use web interface or through the configuration files.

Even though the use of web interface looks simple, for the deployment of dozens or hundreds of SIP telephones this method might be not the best one. For mass deployment, the use of configuration files is much more suitable.

Thus, in this manual we share the instructions on how to configure Polycom SoundStation IP 7000 through the configuration files.

NOTE: The settings submitted through the web interface take precedence over the settings from configuration files. If you want to clear the "web" settings and use configuration files' settings, then you need to reset local configuration on the phone as follows:

- on the phone, press Menu button
- navigate to Settings, choose it and select Advanced
- enter password (default "456") and press "Enter" softkey
- Select Admin Settings
- Navigate down and select Reset to Default…
- Select Reset Local Configuration and confirm your selection by pressing "Yes"

Polycom Phone Configuration Requirements

You can make changes to the configuration files through the web interface to the phone. Using your chosen browser, enter the phone's IP address as the browser address.

By default, Polycom SoundStation IP 7000 requires the use of a File Transfer Protocol (FTP) server. SIP telephones, which are configured to use FTP for provisioning, will look for configuration files on the FTP server specified by option 66 in the DHCP server.

When Polycom SIP phones attempt to retrieve their configuration from the FTP server, they must first log in. So, if the telephones are to be provisioned through an FTP server then it must be configured to allow access for this telephone user account.

The local (or domain) user named "**PlcmSpIp**" with password "**PlcmSpIp**" (capital "i" in the end) should be created on FTP server. In cases when FTP server running on domain controllers or SBS (Small Business Server) the password assignment of "**PlcmSpIp**" could be prohibited since this password does not match the password complexity policy enabled by default. In such situations, we recommend to disable the password complexity policy, create the new user "**PlcmSpIp**" with password "**PlcmSpIp**" and then enable the policy back.

To be provisioned from FTP server, the following files need to be available in the FTP root folder (typically, the FTP root folder location is: **C:\Inetpub\ftproot**):

1. BootROM loader file, e.g. **3111-40000-001.bootrom.ld.**

NOTE: The file name could be different for different Polycom's phone types and in the different firmware releases. For correct file name, check Release Notes for BootROM on Polycom's website.

2. SIP application loader file, **sip.ld** and a specific one e.g. **3111-40000-001.sip.ld**.

NOTE: There are two application files could be downloaded and stored on FTP server, Combined (sip.ld) or Split (e.g. 3111-40000-001.sip.ld). Since sip.ld is significantly bigger in size, it takes more time to load this file from FTP and process it.
From other hand, a specific application file like 3111-40000-001.sip.ld is dedicated only for Polycom SoundStation IP 7000. So, if there are another Polycom phones on site, then administrator must associate every specific SIP application file with the required telephone type. That association needs to be done in the phone's configuration files (e.g. <MAC-address>.cfg) on FTP server.

3. Master configuration file called either **<MAC-address>.cfg** or **000000000000.cfg**.

This file is used by the bootROM and the application for a list of other files that are needed for the operation of the phone.

4. System wide (**sip.cfg)** and per-phone (**phone1.cfg)** configuration files.
You can customize the filenames.

<MAC-address>.cfg

Per-phone master configuration file **<MAC-address>.cfg** indicates which SIP application loader and configuration files should be loaded at the phone's boot up. As in the example below, SIP application for Polycom SoundStation IP 7000 phone and per-phone configuration file phone1_0004f223413d.cfg will be loaded.

```
<APPLICATION APP_FILE_PATH="3111-40000-001.sip.ld" CONFIG_FILES=
"phone1_0004f223413d.cfg, sip.cfg"
```

If per-phone master configuration file **<MAC-address>.cfg** is unavailable in FTP root folder, then the default master configuration file **000000000000.cfg** will be loaded.

sip.cfg

Core configuration file **sip.cfg** contains the settings that are applied to all Polycom phones on the site. Ensure that all common settings are listed in this file.

NOTE: Polycom recommends making a copy of original file sip.cfg and keeping it in a safe place.

For example, it could be a SIP proxy's IP address, the settings for dial plan or timeserver.

We recommend to update the dial plan digitmap with entry "***xxxxxx**" which allows to dial "star" codes after placing the party on-hold.

```
<digitmap dialplan.digitmap="[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|[2-
9]xxxxxxxxx|*xxxxxx|[2-9]xxxT" dialplan.digitmap.timeOut="3|3|3|3|3|3" />
```

Also, you might want to configure the timeserver's IP address to synchronize all Polycom phones in the network (e.g. with public timeserver 128.2.1.21)

```
tcpIpApp.sntp.address="128.2.1.21" tcpIpApp.sntp.address.overrideDHCP="0"
tcpIpApp.sntp.gmtOffset="-18000"
```

where "0" – do not allow DHCP setting to override the setting in this file

"-1800" is GMT offset in seconds for Eastern Standard Time (5x3600=1800).

If you need to change the **audio codec's order**, rank the parameters like in example below:

```
voice.codecPref.IP_7000.Siren22.64kbps="1"
voice.codecPref.IP_7000.G7221C.48kbps="2"
voice.codecPref.IP_7000.G711Mu="3"
voice.codecPref.IP_7000.G729AB="4"
voice.codecPref.IP_7000.G711A="5"
```

In this example, voice codec Siren22.64kbps will be negotiated first, then G7221C.48kbps, etc. to the last one – G.711A.

Some of the sites require the enabling of **SRTP** (Secure Real-Time Transport Protocol) to encrypt the audio streams of SIP phone calls. To enable the support of SRTP, include the following parameters in sip.cfg:

`sec.srtp.enable="1"` -	If set to 1 or Null, the phone accepts SRTP offers. If set to 0, the phone always declines SRTP offers.
`sec.srtp.offer="1"` -	If set to 1 or Null, the phone includes a secure media stream description along with the usual non-secure media description in the SDP of a SIP INVITE. This is for the phone initiating (offering) a phone call.
`sec.srtp.require="1"` -	If set to 1, the phone is only allowed to use secure media streams. Any offered SIP INVITEs must include a secure media description in the SDP or the call will be rejected. For outgoing calls, only a secure media stream description is include in the SDP of the SIP INVITE, meaning that the non-secure media description is not included.

phone1.cfg

The most of the phone's configuration can be done in this file. The default per-phone configuration file (**phone1.cfg**) could be renamed to some specific name to show the connection with the phone, e.g. **phone1_0004f223413d.cfg**. If you do so, then just make sure that you refer to that name in the **<MAC-address>.cfg**.

Find the parameters in **phone1.cfg** and update them accordingly.

Configure the user settings as follows:

`reg.1.displayName="2310"`

`reg.1.address="2310"`

`reg.1.label="John Smith"` – this name appears on the phone's screen

`reg.1.server.1.address="sipint5.mitel.com"` - configure FQDN or IP address of SIP proxy

`reg.1.server.1.port="5060"`

`reg.1.server.1.transport="UDPonly"`

`reg.1.server.1.expires="300"`

`reg.1.callsPerLineKey="2"` It defines the number of calls or conferences which may be active or on-hold per line key associated with this registration. If set to "1" no call waiting allowed. Ensure that this number matches the value set in Multiline Set Keys.

OPTIONAL: Although Polycom SoundStation IP 7000 was not designed as a personal telephone, the **Message Waiting Indication** (MWI) could be still enabled on the phone. You need to enable MWI subscription as follows:

`msg.mwi.1.subscribe="2900"` – Actually this value could any number and it triggers SUBSCRIBE request sent to 3300ICP

`msg.mwi.1.callBackMode="contact"` - This parameter is needed when user presses the key on a phone to retrieve a voicemail message.
If set to "contact" then a call will be placed to the contact specified in the callBack attribute (see next parameter).
If set to "registration" a call will be placed using this registration to the contact registered (the phone will call itself).

`msg.mwi.1.callBack="2900"` - This is the voicemail pilot number on 3300ICP

NOTE: There is no specific key on Polycom SoundStation IP 7000 to place a call to voicemail pilot number.

Users of Polycom SoundStation IP 7000 can activate the **call forwarding** by pressing "Forward" soft button on device when it starts ringing.

Configure these parameters, to enable call forwarding:

`divert.fwd.1.enabled="1"` – It enables the call forwarding. If this parameter set to"0", "Forward" soft button is not displayed on the phones

`divert.busy.1.enabled="1"` – to enable call forwarding on Busy

`divert.noanswer.1.enabled="1"` - to enable call forwarding on No Answer

`divert.noanswer.1.timeout="55"` – it defines timeout before call forwarding on No Answer starts

Resiliency configuration

Polycom has identified two types of redundancy that could be configured on Polycom SoundStation IP 7000:

• **Fail-over**: In this mode, the full phone system functionality is preserved by having a second equivalent capability call server take over from the one that has gone down/off-line. This mode of operation should be done using DNS mechanisms or “IP Address Moving” from the primary to the back-up server. (Scenario 1 in our tests)

• **Fallback**: In this mode, a second less featured call server (router or gateway device) with SIP capability takes over call control to provide basic calling capability, but without some of the richer features offered by the primary call server (for example, shared lines, presence, and Message Waiting Indicator). Polycom phones support configuration of multiple servers per SIP registration for this purpose. (Scenario 2 in our tests)

Polycom’s Recommended Practices for Fallback Deployments

In situations where server redundancy for fall-back purpose is used, the following measures should be taken to optimize the effectiveness of the solution:

1. Deploy an on-site DNS server to avoid long call initiation delays that can result if the DNS server records expire.

2. Do not use OutBoundProxy configurations on the phone if the OutBoundProxy could be unreachable when the fallback occurs. SoundPoint IP phones can only be configured with one OutBoundProxy per registration and all traffic for that registration will be routed through this proxy for all servers attached to that registration. If Server 2 is not accessible through the configured proxy, call signaling with Server 2 will fail.

3. Avoid using too many servers as part of the redundancy configuration as each registration will generate more traffic.

4. Educate users as to the features that will not be available when in “fallback” operating mode.

To **provide the resiliency behavior as in Scenario 1**, configure the following parameter in **phone1.cfg**:

```
reg.1.server.1.address="sipint5sipint4.mitel.com"
```

In this example, sipint5 is the DNS name of primary SIP proxy (3300 ICP) and sipint4 is the secondary SIP proxy (3300 ICP).

NOTE: Before configuring this parameter, make sure that DNS server correctly resolves the names of both SIP proxies to IP addresses! The order, in which the SIP proxies IP addresses are resolved, is also important! To check it, use the command in command shell:

```
nslookup sipint5sipint4.mitel.com
```

If port number is configured, e.g. `reg.1.server.1.port=”5060”`, the only lookup will be an A record. If no port is given, NAPTR and SRV records will be tried, before falling back on A records if NAPTR and SRV records return no results. If no port is given, and none is found through DNS, 5060 will be used.

To **provide the resiliency behaviour as in Scenario 2**, configure the following parameters in **phone1.cfg**:

`reg.1.server.1.address="sipint5.mitel.com"`

`reg.1.server.1.expires="300" –` time in seconds

```
reg.1.server.2.address="192.168.101.20"
reg.1.server.2.expires="500"  -  time in seconds
```

NOTE: Since due to network failure DNS server could be unavailable/unreachable, Polycom recommends using IP address for `reg.1.server.2.address` instead of FQDN.

We recommend keeping the low value for `reg.1.server.1.expires` and ensure that register expiration time for primary and secondary SIP proxies is not the same (like in the example above, there are 300 and 500 seconds).

Multi-Protocol Border Gateway Setup Notes (Optional)

The following steps show how to program the Multi-Protocol Border Gateway server to allow connections between the Polycom SoundStation IP 7000 and the 3300 ICP for teleworking.

Network Requirements

- Please refer to the Multi-Protocol Border Gateway Engineering guidelines for further information.

Assumptions for the Multi-Protocol Border Gateway Configuration

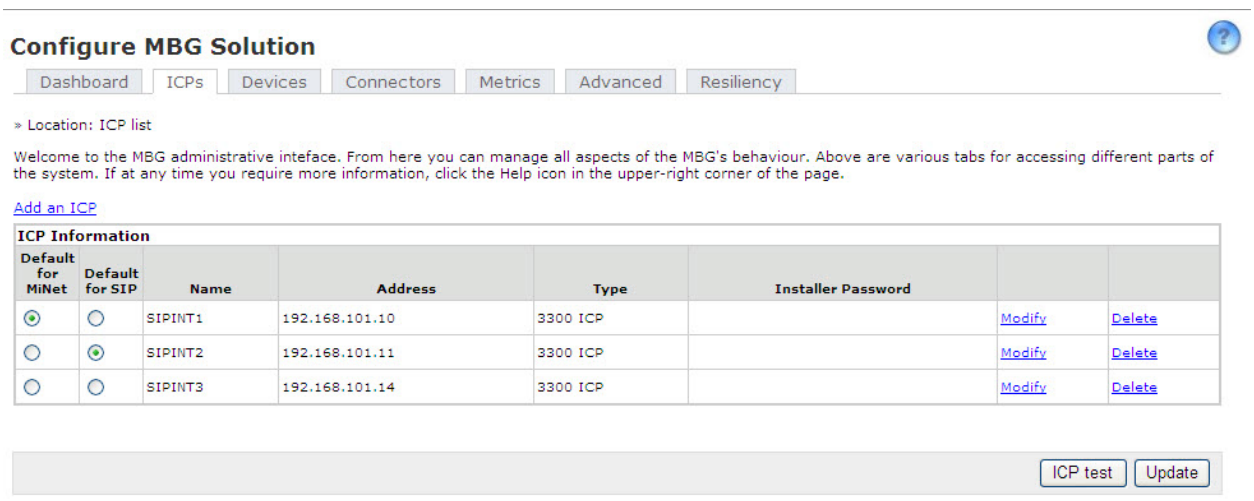
- 3300 ICP configuration completed as per instructions in previous section.
- The SIP signaling connection between the 3300 ICP and the Multi-Protocol Border Gateway server uses UDP on Port 5060.
- Multi-Protocol Border Gateway server installed and configured for SIP client support.

ICPs

On the ICPs tab, click **Add an ICP** and enter ICP information (name, IP address, type).

Select the **Default for SIP** and click **Update**.

In this example, the 3300 ICP with IP address 192.168.10.11 is the default SIP ICP:



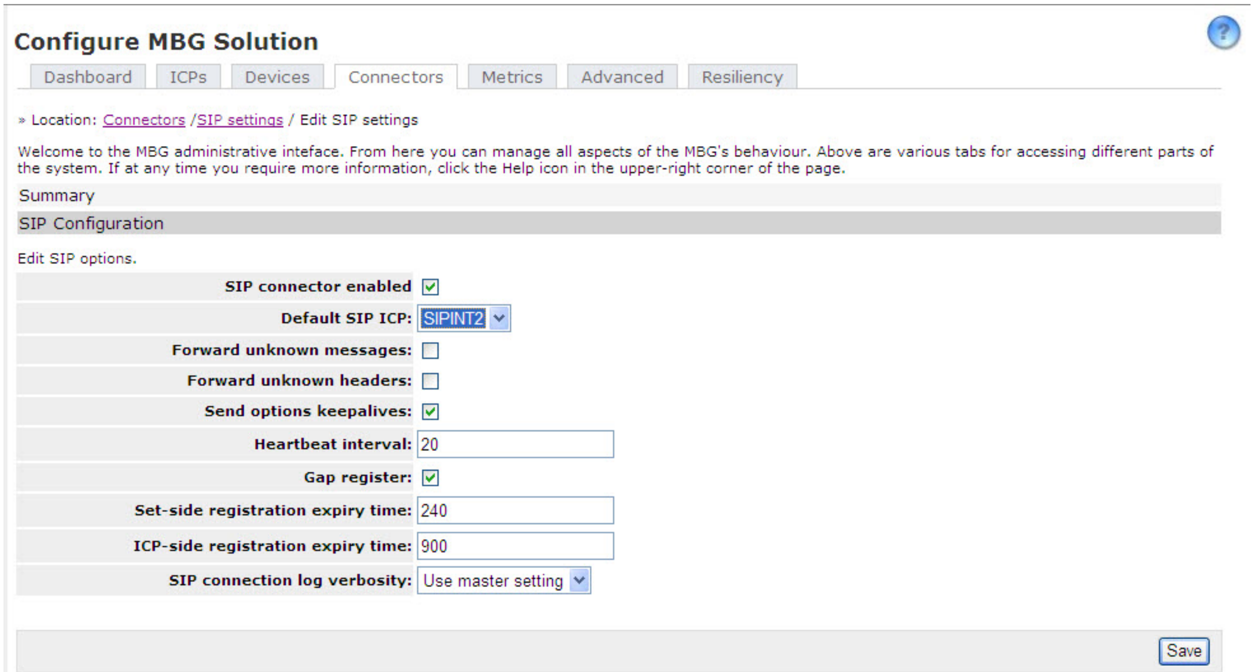
Connectors – SIP Configuration

Enable SIP support:

On the Connectors tab, click **SIP Options** and then click **Edit**.

Click to select the **SIP support enabled** check box.

Click **Save**.

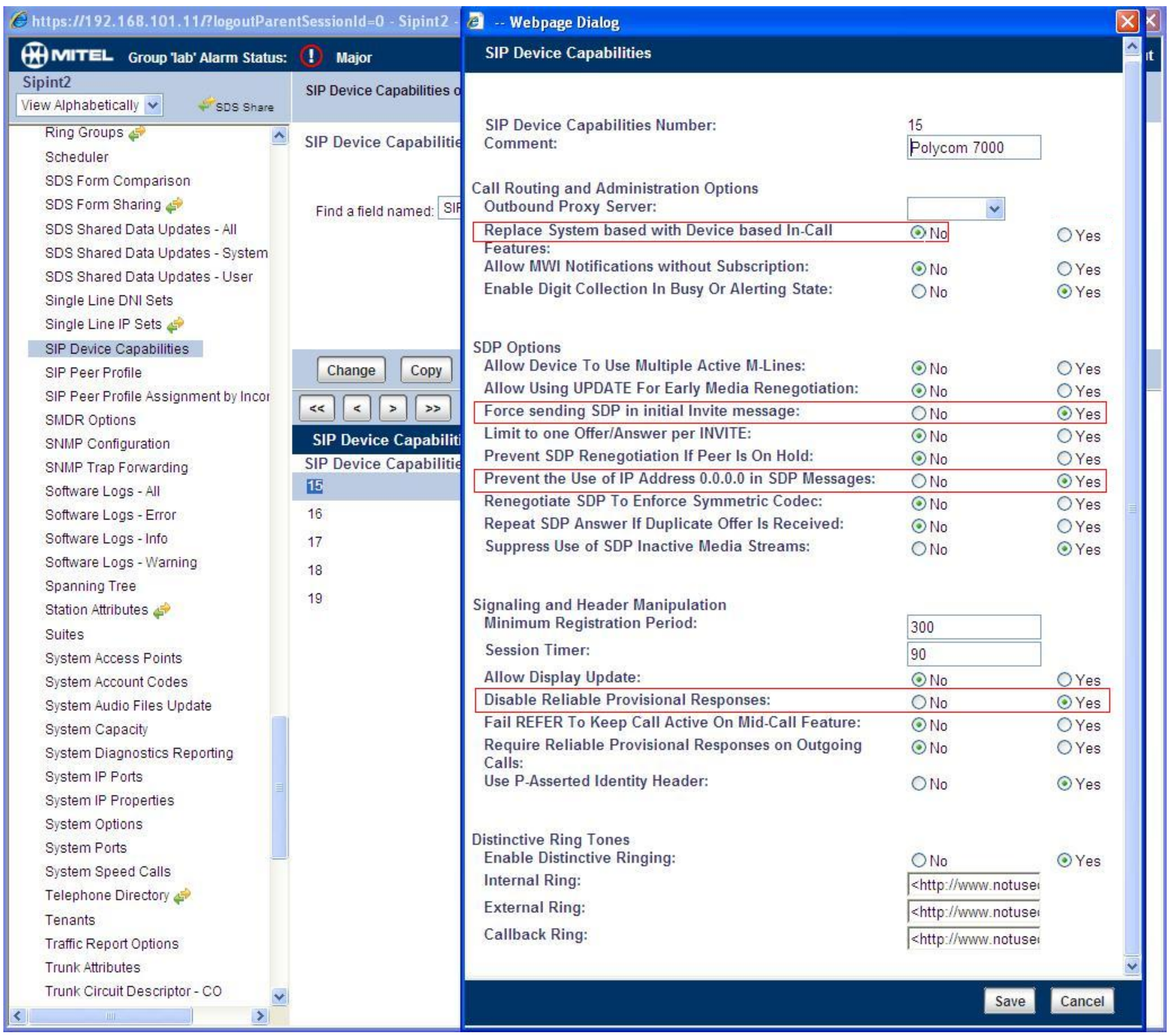


Appendix A

Normally the Polycom 7000 does not allow more than 3 participants in a conference call in device based mode on the MCD. To add 4 or more parties to a conference, the conferencing capabilities of the MCD will need to be used via the MCDs Feature Access code.

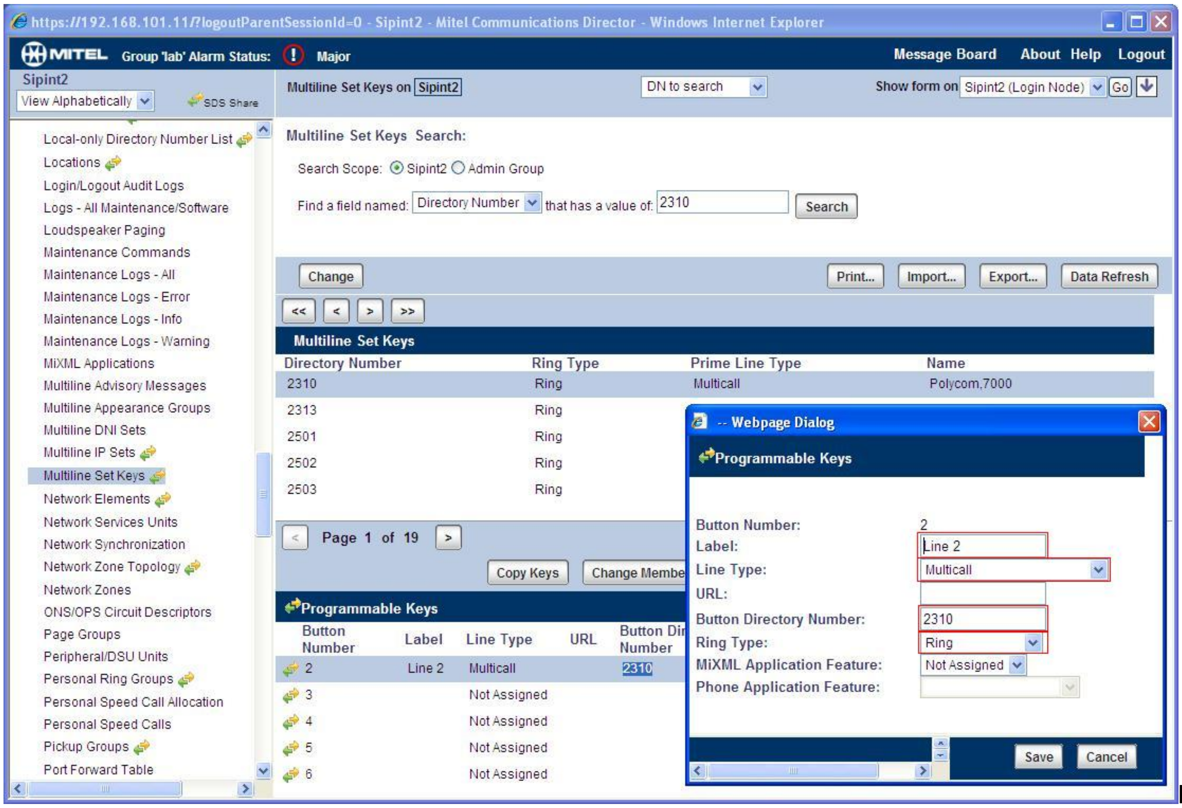
SIP Device Capabilities Assignment

First, change to System Based incall features in the SIP Device Capabilities form. In the SIP Device Capabilities form, program a SIP Device Capabilities Number for Polycom SoundStation IP 7000 device. Ensure that “Replace System based with Device based In-Call Feature” is set to ‘**No**’.



Multiline Set Keys

You use the Multiline Set Keys form to assign the line type, ring type, and directory number to each line selected on the Polycom SoundStation IP 7000 device. Minimum 3 lines are required to perform a conference.



Loc

Locate the feature access code for conference in the Feature Access Code form.

Example of setting up a conference

1. From the Polycom 7000, call 7001
2. put on hold and call 7002
3. put on hold and call *40 (conference fac)
4. put on hold and call 7003
5. put on hold and call *40 (conference fac)
6. put on hold and call 7004
7. put on hold and call *40 (conference fac)

Global Headquarters	U.S.	EMEA	CALA	Asia Pacific
Tel: +1(613) 592-2122	Tel: +1(480) 961-9000	Tel: +44(0)1291-430000	Tel: +1(613) 592-2122	Tel: +852 2508 9780
Fax: +1(613) 592-4784	Fax: +1(480) 961-1370	Fax: +44(0)1291-430400	Fax: +1(613) 592-7825	Fax: +852 2508 9232

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