

MITEL – SIPCoE

Technical Configuration Notes



Configure MCD 4.1 for use with the
Polycom SoundStation IP7000

SIP CoE 08-5159-00020

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

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OVERVIEW	1
Interop History.....	1
Interop Status	1
Software & Hardware Setup.....	2
Tested Features.....	3
Resiliency	4
Device Limitations.....	5
Device Recommendations	5
Network Topology	6
CONFIGURATION NOTES	7
3300 ICP Configuration Notes.....	7
Network Requirements.....	7
Assumptions for the 3300 ICP Programming.....	7
Licensing and Option Selection – SIP Licensing	8
Multiline IP Set Configuration	9
Class of Service Assignment	10
SIP Device Capabilities Assignment.....	12
Station Attributes Assignment	13
Call Rerouting Assignment.....	15
POLYCOM SOUNDSTATION IP 7000 SETUP NOTES.....	17
Polycom Phone Configuration Requirements.....	18
<MAC-address>.cfg	18
sip.cfg	19
phone1.cfg.....	20
Resiliency configuration	21
MULTI-PROTOCOL BORDER GATEWAY SETUP NOTES (OPTIONAL).....	23
Network Requirements.....	23
Assumptions for the Multi-Protocol Border Gateway Configuration.....	23
ICPs.....	23
Connectors – SIP Configuration.....	23
APPENDIX A.....	25
SIP Device Capabilities Assignment	25

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	<p>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.</p>
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

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 Plans 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	<p>Even though Call Forward (e.g. on No Answer or Always) is enabled thorough the web interface, it remains inactive.</p> <p>Recommendation: Activate transfer manually on the phone when it is ringing or contact Polycom for support of this feature.</p>
Conference	<p>The Polycom 7000 is limited to initiate a 3 party conference and is unable to add a 4th party.</p> <p>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</p>
Resiliency	<p>Scenario 1:</p> <p>When registered on a secondary 3300, Polycom SoundStation IP 7000 can not receive the calls</p> <p>Recommendation: none</p> <p>Scenario 2:</p> <p>Polycom SoundStation IP 7000 sticks to the primary PBX and does not try to register on a secondary (when the primary is unavailable).</p> <p>Recommendation: The phone reboot is required</p>

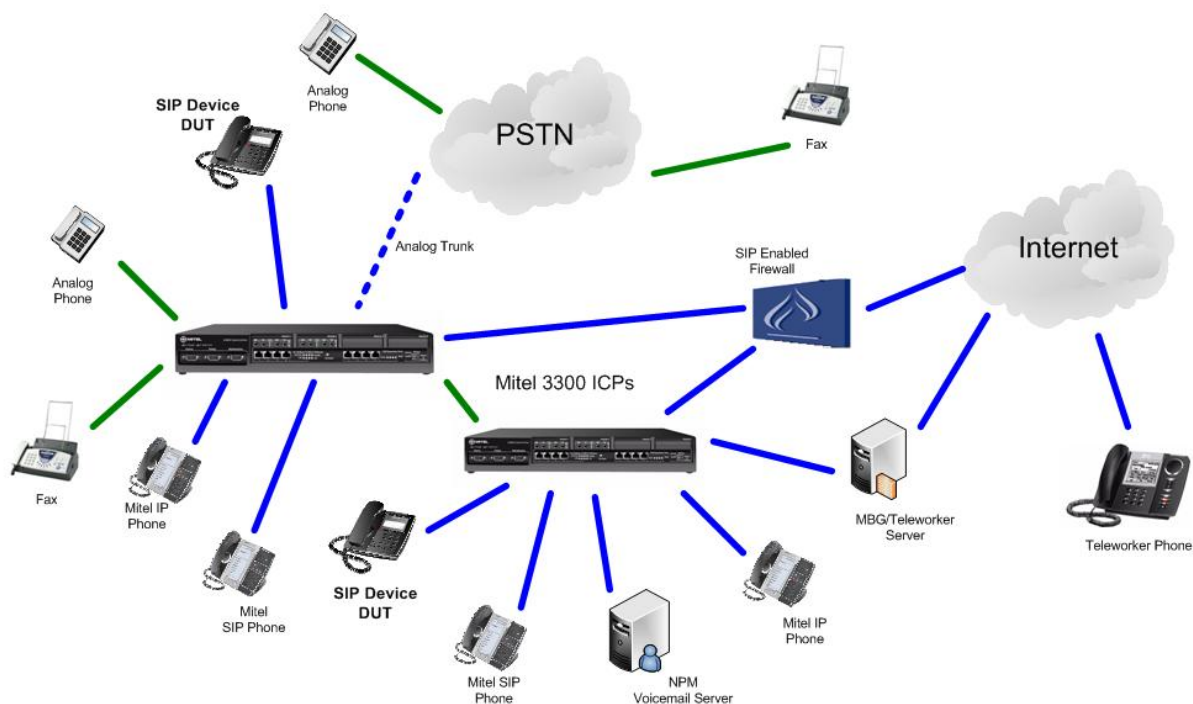
Device Recommendations

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.

Mitel SIP Interop Network Configuration



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.

The screenshot shows the 'License and Option Selection' form in the Sipint2 web interface. The left navigation menu lists various system components, with 'License and Option Selection' currently selected. The main content area displays the following information:

License and Option Selection

Online Licensing with the Application Management Center

Application Record ID:

Purchased Options

Users	
IP User Licenses:	1300
External Hot Desk User Licenses:	100
ACD Active Agent Licenses:	100
HTML Apps Infrastructure Licenses:	100
Analog Line Licenses:	10

Voice Mail

Mailbox Licenses:	100
Voice Mail Networking:	Yes
Advanced Voice Mail:	Yes
Voice Mail Hospitality/PMS:	Yes

Trunking/Networking

Digital Link Licenses:	16
Compression Licenses:	16
FAX Over IP (T.38) Licenses:	16
SIP Trunk Licenses:	1000
XNET Networking:	Yes
IP Networking:	Yes

Others

Tenanting:	Yes
MLPP:	No
Remote Management:	Yes
Hardware Identifier:	0000002F9EE1
Password:	*****

Configuration Options

Country:	North America
Networking Option:	Yes
Mitai/Tapi Computer Integration:	Yes
Extended Agent Skill Group:	No
Maximum Elements per Cluster:	30
Maximum Configurable IP Users and Devices:	700
Extended Hunt Group:	No

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.

The screenshot displays the Mitel Communications Director web interface. The main window shows the 'Multiline IP Sets' configuration page. A sidebar on the left contains various navigation links. The main content area features a table of Multiline IP Sets and a 'Range Programming' dialog box for editing records.

Multiline IP Sets Table:

Device Id	Hot Desk User	Device Type	Auxiliary Module
12	No	Generic SIP Phone	None
13	No	Generic SIP Phone	None
14	No	Generic SIP Phone	None
15	No	Generic SIP Phone	None
16	No	Generic SIP Phone	None

Range Programming - Webpage Dialog:

This form allows you to change one or more records, starting at the following record:

Device Id	Hot Desk User	Device Type	Auxiliary Module	Number	Local-only DN	User PIN	ACD Enabled	Line Type	Interconnect Number	Ext
12	No	Generic SIP Phone	None	2310	False	*****	No	Multicall	1	No

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Device Id:	-	12	-
Hot Desk User:	Change to	<input checked="" type="radio"/> No <input type="radio"/> Yes	-
Device Type:	Change to	Generic SIP Phone	-
Auxiliary Module:	Change to	None	-
Number:	Change to	2310	-
Local-only DN:	Change to	<input type="checkbox"/>	-
User PIN:	Change to	*****	-
Confirm User PIN:	Change to	*****	-
ACD Enabled:	Change to	<input checked="" type="radio"/> No <input type="radio"/> Yes	-
Line Type:	-	Multicall	-

Buttons: Preview, Save, Cancel

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

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

Local Music On Hold source:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Record-A-Call - Start Automatic Incoming Call Recording:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Loudspeaker Pager Override:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Record-A-Call - Start Automatic Outgoing External Call Recording:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Loudspeaker Pager Equivalent Zone Override Security:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Record-A-Call - Save Recording on Hang-up:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Maintain Ringing Party During Recall:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Recorded Announcement Device:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Message Waiting:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Recorded Announcement Device - Advanced:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Message Waiting Audible Tone Notification:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Redial Facilities:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Message Waiting Deactivate On Off-Hook:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Return Disconnect Tone When Far End Party Clears:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Message Waiting - Disable Ringing Lamp Notification:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Ringing Line Select:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Message Waiting Inquire:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	SC1000 Attendant Basic Function Key:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Loop Test:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	SMDR External:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Message Center Remote Read Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	SMDR Internal:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Music:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Speak@Ease Preferred:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set On-hook Dialing:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Suppress Delivery of Caller ID Display between Sets:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Phonebook Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Suppress Delivery of Caller ID Display between Sets - Override:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Multiline Set Voice Mail Callback Message Erasure Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Suppress Display Of Account Code Numbers:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Music on Hold on Transfer:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Suppress Redial Display:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Name Suppression on outgoing Trunk Call:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Suppress Simulated CCM after ISDN Progress:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Non DID Extension:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Third Party Call Forward Follow Me Accept:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Non-Prime Public Network Identity:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Third Party Call Forward Follow Me Allow:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Non Verified Account Code:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Timed Reminder Allowed:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
Off-Hook Voice Announce Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Trunk Calling Party Identification:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Trunk Flash Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ONS CLASS/CLIP: Set:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Two B-Channel Transfer Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
ONS CLASS/CLIP: Visual Call Waiting:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Use Held Party Device for Call Re-routing:	<input type="radio"/> No	<input checked="" type="radio"/> Yes
ONS/OPS Internal Ring Cadence for External Callers:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Use Called Party Call Hold Timer:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Originator's Display Update In Call Forwarding/Rerouting:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Voice Mail Softkey:	<input checked="" type="radio"/> No	<input type="radio"/> Yes
Override Interconnect Restriction on Transfer:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Account Code Length:	12	
Pager Access All Zones:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	After Answer Display Time:		
Pager Access Individual Zones:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Answer Plus Delay To Message Timer:	20	
PC Port On IP Device - Disable:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Answer Plus Expected Off-hook Timer:	30	
Phonebook Lookup - Default to User Location:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Answer Plus Message Length Timer:	10	
Phonebook Lookup - Display User Location:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Answer Plus System Reroute Timer:	0	
Phone Lock:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Attendant Busy Out Timer:	10	
Privacy Released:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Auto Campon Timer:		
Public Network Access via DPNSS:	<input type="radio"/> No	<input checked="" type="radio"/> Yes	Busy Tone Timer:	30	
Public Network Identity Provided:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Call Duration:	10	
Public Network To Public Network Connection Allowed:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Call Duration Forced Cleardown Timer:	0	
Public Trunk:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Call Forward - Delay:	0	
R2 Call Progress Tone:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Call Forward No Answer Timer:	15	
Recall If Transferred to Original Call Destination:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Call Hold Timer:	30	
Record-A-Call Active:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Call Park Timer:	180	
Record-A-Call - Start Automatic Incoming Call Recording:	<input checked="" type="radio"/> No	<input type="radio"/> Yes	Campon Recall Timer:	10	

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.

The screenshot shows the 'SIP Device Capabilities' form in the Mitel SIPint2 web interface. The form is titled 'SIP Device Capabilities' and shows configuration options for a Polycom 7000 device. The 'SIP Device Capabilities Number' is set to 15. The 'Comment' is 'Polycom 7000'. The 'Outbound Proxy Server' is set to a dropdown menu. The 'Replace System based with Device based In-Call' option is set to 'Yes'. The 'Features' section includes 'Allow MWI Notifications without Subscription' (No), 'Enable Digit Collection In Busy Or Alerting State' (No), and 'SDP Options' (No). The 'Signaling and Header Manipulation' section includes 'Minimum Registration Period' (300), 'Session Timer' (90), 'Allow Display Update' (No), 'Disable Reliable Provisional Responses' (Yes), 'Fail REFER To Keep Call Active On Mid-Call Feature' (No), 'Require Reliable Provisional Responses on Outgoing Calls' (No), and 'Use P-Asserted Identity Header' (No). The 'Distinctive Ring Tones' section includes 'Enable Distinctive Ringing' (No), 'Internal Ring' (<http://www.notusei), 'External Ring' (<http://www.notusei), and 'Callback Ring' (<http://www.notusei). The form has 'Save' and 'Cancel' buttons at the bottom right.

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.

Range Programming -- Webpage Dialog

2305 1 4 4 4 1 1

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Number:	-	2305	-
Intercept Number:	Change to	1	
Class of Service - Day:	Change to	4	
Class of Service - Night1:	Change to	4	
Class of Service - Night2:	Change to	4	
Class of Restriction - Day:	Change to	1	
Class of Restriction - Night1:	Change to	1	
Class of Restriction - Night2:	Change to	1	
Default Acct. Code:	Change to		
Zone Assignment Method:	Change to	<input checked="" type="radio"/> Default <input type="radio"/> Manual	-
Zone ID:	Change to		
SIP Device Capabilities:	Change to	15	

Preview Save Cancel

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.

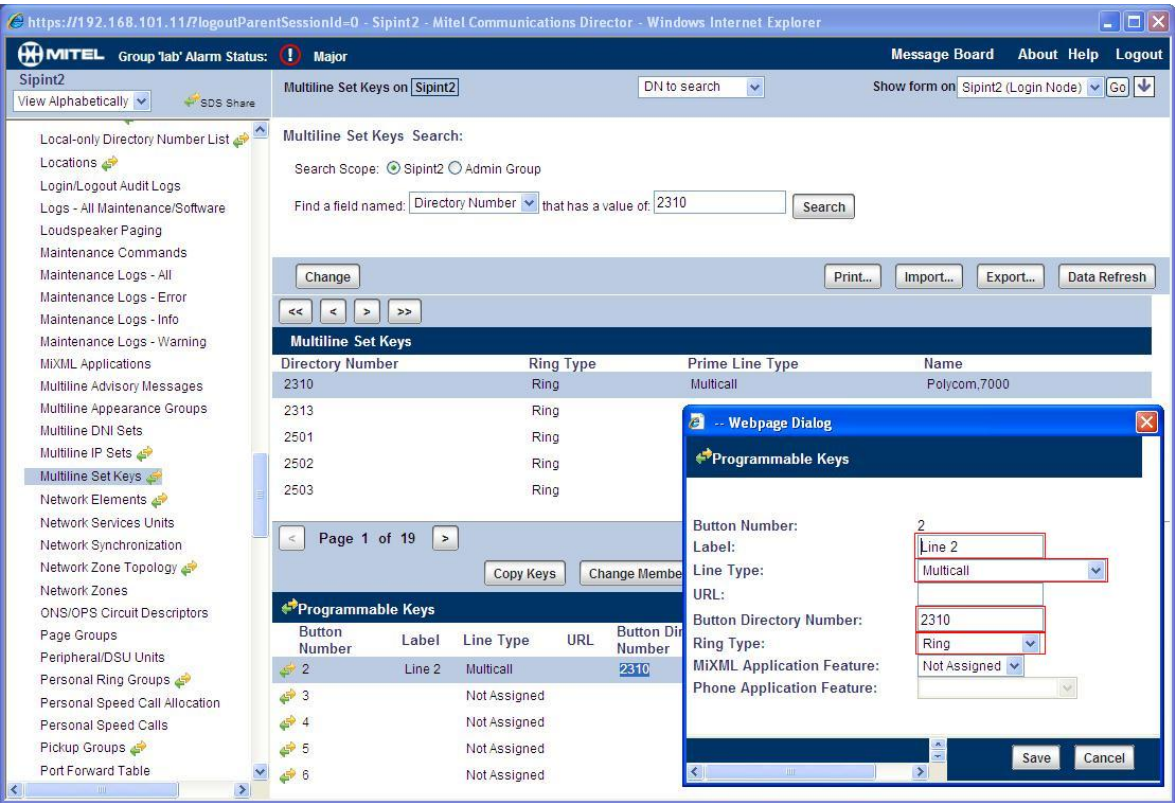


Figure 6 – Multiline Set Key Assignment form

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.

First Alternative Number	Busy / DND DID	Busy / DND TIE	Busy / DND CO	Busy / DND Int	No Answer DID	No Answer TIE	No Answer CO	No Answer Int	Directory Number
1	Normal	Normal	Normal	Normal	Normal	Normal	Normal	Normal	
2	Normal	Normal	Normal	Normal	Normal	Normal	Normal	Normal	
3	This	This	This	This	This	This	This	This	2305
4	This	This	This	This	Normal	Normal	Normal	Normal	2028
5	Normal	Normal	Normal	Normal	Normal	Normal	Normal	Normal	
6	This	This	This	This	This	This	This	This	2950
7	This	This	This	This	Normal	This	This	This	2900
8	This	This	This	This	This	This	This	This	2007
9	Normal	Normal	Normal	Normal	This	This	This	This	2900
10	This	This	This	This	This	This	This	This	2513
11	Normal	Normal	Normal	Normal	This	This	This	This	2050
12	This	This	This	This	Normal	Normal	Normal	Normal	2050
13	This	This	This	This	This	This	This	This	*7770
14	This	This	This	This	This	This	This	This	2113
15	This	This	This	This	This	This	This	This	7751

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.

The screenshot shows the Mitel Communications Director web interface. The main table displays call rerouting data for various numbers. A 'Webpage Dialog' window is open, showing the 'Call Rerouting' form for number 2310.

Number	Call Rerouting - Day	Call Rerouting - Night1	Call Rerouting - Night2	Call Rerouting DND Type	Call Rerouting - 1st Alt.	Call Rerouting - 2nd Alt.
2302	1	1	1	All	1	1
2304	1	1	1	All	1	1
2305	1	1	1	All	1	1
2306	1	1	1	All	1	1
2307	1	1	1	All	7	1
2310	1	1	1	All	7	1
2313	1	1	1	All	1	1
2501	1	1	1	All	1	1
2502	1	1	1	All	1	1
2503	1	1	1	All	1	1
2504	1	1	1	All	1	1
2510	1	1	1	All	1	1
2511	1	1	1	All	1	1
2513	1	1	1	All	1	1
2514	1	1	1	All	1	1
2521	1	1	1	All	1	1
2523	1	1	1	All	1	1
2540	1	1	1	All	1	1
2552	1	1	1	All	1	1
2571	1	1	1	All	1	1

Webpage Dialog: Call Rerouting

Number: 2310

Call Rerouting - Day: 1

Call Rerouting - Night1: 1

Call Rerouting - Night2: 1

Call Rerouting DND Type: All

Call Rerouting - 1st Alt.: 7

Call Rerouting - 2nd Alt.: 1

Buttons: Save, Cancel

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 "**PlcmSplp**" with password "**PlcmSplp**" (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 "**PlcmSplp**" 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 "**PlcmSplp**" with password "**PlcmSplp**" 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]xxxxxxxxxx|[2-9]xxxxxxxxxx|*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:

Configure MBG Solution



[Dashboard](#)
[ICPs](#)
[Devices](#)
[Connectors](#)
[Metrics](#)
[Advanced](#)
[Resiliency](#)

» Location: ICP list

Welcome to the MBG administrative interface. From here you can manage all aspects of the MBG's behaviour. Above are various tabs for accessing different parts of the system. If at any time you require more information, click the Help icon in the upper-right corner of the page.

[Add an ICP](#)

Default for MiNet	Default for SIP	Name	Address	Type	Installer Password		
<input checked="" type="radio"/>	<input type="radio"/>	SIPINT1	192.168.101.10	3300 ICP		Modify	Delete
<input type="radio"/>	<input checked="" type="radio"/>	SIPINT2	192.168.101.11	3300 ICP		Modify	Delete
<input type="radio"/>	<input type="radio"/>	SIPINT3	192.168.101.14	3300 ICP		Modify	Delete

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**.

Configure MBG Solution

[Dashboard](#) [ICPs](#) [Devices](#) [Connectors](#) [Metrics](#) [Advanced](#) [Resiliency](#)

» Location: [Connectors](#) / [SIP settings](#) / Edit SIP settings

Welcome to the MBG administrative interface. From here you can manage all aspects of the MBG's behaviour. Above are various tabs for accessing different parts of the system. If at any time you require more information, click the Help icon in the upper-right corner of the page.

Summary

SIP Configuration

Edit SIP options.

SIP connector enabled:	<input checked="" type="checkbox"/>
Default SIP ICP:	<div>SIPINT2</div>
Forward unknown messages:	<input type="checkbox"/>
Forward unknown headers:	<input type="checkbox"/>
Send options keepalives:	<input checked="" type="checkbox"/>
Heartbeat interval:	<div>20</div>
Gap register:	<input checked="" type="checkbox"/>
Set-side registration expiry time:	<div>240</div>
ICP-side registration expiry time:	<div>900</div>
SIP connection log verbosity:	<div>Use master setting</div>

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’.

The screenshot shows the 'SIP Device Capabilities' configuration form in the Sipint2 web interface. The form is titled 'SIP Device Capabilities' and contains various settings for SIP device capabilities. The 'SIP Device Capabilities Number' is set to 15, and the 'Comment' is 'Polycom 7000'. The 'Replace System based with Device based In-Call' feature is set to 'No'. Other features like 'Allow MWI Notifications without Subscription' and 'Enable Digit Collection In Busy Or Alerting State' are set to 'Yes'. The 'SDP Options' section includes settings for 'Force sending SDP in initial Invite message' (Yes), 'Prevent the Use of IP Address 0.0.0.0 in SDP Messages' (Yes), and 'Disable Reliable Provisional Responses' (Yes). The 'Signaling and Header Manipulation' section includes settings for 'Minimum Registration Period' (300), 'Session Timer' (90), and 'Use P-Asserted Identity Header' (Yes). The 'Distinctive Ring Tones' section includes settings for 'Enable Distinctive Ringing' (Yes), 'Internal Ring' (<http://www.notusei), 'External Ring' (<http://www.notusei), and 'Callback Ring' (<http://www.notusei).

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.

The screenshot shows the 'Multiline Set Keys' form in the Mitel Communications Director web interface. The search scope is set to 'Sipint2' and the search criteria is 'Directory Number' with a value of '2310'. The table below lists the Multiline Set Keys:

Directory Number	Ring Type	Prime Line Type	Name
2310	Ring	Multicall	Polycom,7000
2313	Ring		
2501	Ring		
2502	Ring		
2503	Ring		

The 'Programmable Keys' dialog box is open, showing the configuration for Button 2:

- Button Number: 2
- Label: Line 2
- Line Type: Multicall
- URL:
- Button Directory Number: 2310
- Ring Type: Ring
- MiXML Application Feature: Not Assigned
- Phone Application Feature:

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 Fax: +1(613) 592-4784	Tel: +1(480) 961-9000 Fax: +1(480) 961-1370	Tel: +44(0)1291-430000 Fax: +44(0)1291-430400	Tel: +1(613) 592-2122 Fax: +1(613) 592-7825	Tel: +852 2508 9780 Fax: +852 2508 9232

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