MITEL – SIPCoE

Technical Configuration Notes

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Configure MCD 4.1 for use with the Polycom SoundStation IP7000

SIP CoE 08-5159-00020



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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:



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	v
Call Forward	Forwarding a call to another destination	
Conference	Conferencing multiple calls together	\checkmark
Redial	Last Number Redial	√
MWI	Message Waiting Indication	√
Dynamic Extension	Personal Ring Group configuration	\checkmark
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	1

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
C	issues found	X - Issues found	. cannot recomm	end use	- Issues found

Image: Provide the second s X - Issues found, cannot recommend use

> 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 guickly, 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.

https://192.168.101.11/?logoutPare	ntSessionId=0 - Sipint2 - Mitel Communications	: Director - Windows Internet Ex 🔳 🗖 🔀
Group 'lab' Alarm Status:	(1) Major	Message Board About Help Logout
Sipint2 View Alphabetically	License DN to search 🗸	Show form on Sipint2 (Login Node) 💌 Go ↓
IP Telephones - Programmed IP Telephones - Unprogrammed	Change Print.	Import Export Data Refresh
IP/XNET Trunk Groups IP/XNET Trunk Profiles	License and Option Selection	
ISDN Outgoing Numbers ISDN Protocol	Online Licensing with the Application Mana	gement Center
L2 to CESID Mapping	Application Record ID:	
LAN Policy (QoS) Layer 2 Switch	Purchased Options	
License and Option Selection	Users	
Linked Suites 🥔	IP User Licenses: External Hot Desk User Licenses:	1300 100
Local-only Directory Number List 🥔	ACD Active Agent Licenses: HTML Apps Infrastructure Licenses:	100 100
Login/Logout Audit Logs	Analog Line Licenses: Voice Mail	10
Logs - All Maintenance/Software	Mailbox Licenses: Voice Mail Networking:	100 Yes
Maintenance Commands	Advanced Voice Mail: Voice Mail Hospitality/PMS:	Yes Yes
Maintenance Logs - All Maintenance Logs - Error	Trunking/Networking Digital Link Licenses:	16
Maintenance Logs - Info	Compression Licenses: FAX Over IP (T.38) Licenses:	16 16
Maintenance Logs - Warning MiXML Applications	SIP Trunk Licenses: XNET Networking:	1000 Yes
Multiline Advisory Messages	IP Networking: Others	Yes
Multiline Appearance Groups Multiline DNI Sets	Tenanting: MLPP:	Yes No
Multiline IP Sets 🧬	Remote Management: Hardware Identifier:	Yes 0000002F9EE1
Multiline Set Keys 🥔 Network Elements 🧬	Password:	******
Network Services Units	Configuration Options	
Network Synchronization	Country: Networking Option:	North America Yes
Network Zones	Mitai/Tapi Computer Integration: Extended Agent Skill Group:	Yes No
ONS/OPS Circuit Descriptors	Maximum Elements per Cluster: Maximum Configurable IP Users	30
Peripheral/DSU Units	and Devices: 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.

Image: Comp Stat/ Atams Status: Major Range Programming - Webpage Dialog. Sipin2 Multiline IP Sets on Signit2 DN to set Urw Alphabetically webpage Dialog. Multiline IP Sets on Signit2 DN to set Law Policy (CQS) Lay 2 Switch Multiline IP Sets Search: Device Hot Device Type Addline DN Loca-Influence Multiline IP Sets Search: Device Hot Device Type Addline DN Page Range Programming - Multiline IP Sets Loca-Influence Multiline IP Sets Search: Device Hot Device Type Addline DN Page Range Programming - Multiline IP Sets Loca-Influence Multiline IP Sets Multiline IP Sets Multiline IP Sets Device Type Addline DN Loca-Influence Addl Change Delete I. Enter the number of records to change I Discolution MulticalI 1 No Loca-Influence Logs - MI Multiline IP Sets Device Type Auxiliary Multiline IP Sets Increment by Maintenance Logs - MI Multiline IP Sets Device Type Auxiliary Multiline IP Sets Increment by Maintenance Logs - MI Multiline IP Sets Device Type Auxiliary Multiline IP Sets Increment by	6 https://192.168.101.11/?logoutPare	ntSessionId=0 - Sipint2 - Mitel Communications Direc	tor - Windows Internet Explorer
Spiniz Multiline IP Sets on Spiniz DH to set LAN Policy (QoS) Multiline IP Sets on Spiniz DH to set Lange 2 Switch Lices and Option Selection Multiline IP Sets Search: Lices and Option Selection Find a field named: Number I to a field named: This form allows you to change one or more records, starting at the following record: Locations Location Selection Find a field named: Number I to a field named: This form allows you to change one or more records, starting at the following record: Location Selection Location Selection Find a field named: Number I to a field named: This form allows you to change one or more records, starting at the following record: Location Selection Location Selection Find a field named: Number I to a field named: The field Name Location Selection Add Change Delete 1. Enter the number of records to change:	Group 'lab' Alarm Status:	Major	🖉 Range Programming Webpage Dialog
LAN Policy (005) Autilian IP Sets Layer 2 Switch Intercent of Option Selection Line Quality Measurement Intercent of the Intercent of	Sipint2 View Alphabetically	Multiline IP Sets on Sipint2 DN to se	Change Range Programming - Multiline IP Sets
Lices and Option Selection Line Quality Measurement Linked Suites and Local-only Directory Number List and Number Account List and Nu	LAN Policy (QoS)	Multiline IP Sets Search:	This form allows you to change one or more records, starting at the following record:
Locations LoginLogout Audit Logs Logs - All Maintenance/Software Loudspeaker Paging Maintenance Commands Maintenance Logs - All Maintenance Logs - All Maintenance Logs - Riror Maintenance Logs - Kiror Maintenance Logs - Warning MiXILL Applications Muttline Advisory Messages Muttline Advisory Messages Muttline Advisory Messages Muttline IP Sets Muttline Set Keys Network Services Units Network Zenes ONSIOPS Circuit Descriptors ONSIOPS Circuit Descriptors Page Groups	License and Option Selection Line Quality Measurement Linked Suites 🛷	Find a field named: Number 💌 that has a value of. 2	Device Hot Device Type Auxiliary Number Local-User ACD Line Interconnect Ext Id Desk Module only PIN Enabled Type Number Ho User DN Use Lice
LoginLogout Audit Logs Logs - All Maintenance/Software Loudspeaker Paging Maintenance Commands Maintenance Logs - All Maintenance Logs - All Maintenance Logs - All Maintenance Logs - Kino Multiline PS ets 12 No Generic SIP Phone Nuttiline PS ets 13 No Generic SIP Phone 14 No Generic SIP Phone 15 No Generic SIP Phone 16 No Generic SIP Phone 18 Network Zene ONSIOPS Circul Descriptors Page Groups Vetwork Zones ONSIOPS Circul Descriptors Page Groups	Locations 🧬		12 No Generic SIP Phone None 2310 False ******* No Multicali 1 No
Maintenance Logs - All Maintenance Logs - All Maintenance Logs - All Maintenance Logs - All Maintenance Logs - Info Maintenance Logs - Info Maintenance Logs - Info Maintenance Logs - Info Maintenance Logs - Info Maintenance Logs - Info Maintenance Logs - Warning Multiline Advisory Messages Multiline Advisory Messages Partice 12 Multiline Advisory Messages Partice 13P Phone Multiline DNI Sets 14 No Generic SIP Phone None Multiline Set Keys # 15 No Generic SIP Phone None Network Zene Sould bescriptors 16 No Generic SIP Phone None Network Zones ONSIOPS Circuit Descriptors Page Groups Multicall - Network Zones Verse - Multicall - <tr< td=""><td>Login/Logout Audit Logs Logs - All Maintenance/Software Loudspeaker Paging Haintenana Commende</td><td>Add Change Delete</td><td>I. Enter the number of records to change: 1 2. Define the Change Range Programming Pattern:</td></tr<>	Login/Logout Audit Logs Logs - All Maintenance/Software Loudspeaker Paging Haintenana Commende	Add Change Delete	I. Enter the number of records to change: 1 2. Define the Change Range Programming Pattern:
Maintenance Logs - Kiror Maintenance Logs - Info Maintenance Logs - Viaming Midintenance Logs - Viaming </td <td>Maintenance Commands</td> <td>Hultiline IP Sets</td> <td>Field Name Change action Value to change Increment by</td>	Maintenance Commands	Hultiline IP Sets	Field Name Change action Value to change Increment by
Maintenance Logs - Info Device Hot Desk User Povice Type Auxiliary Maintenance Logs - Warning Midlike Device Type Auxiliary Module Change to @ No @ Yes - Midlike Advisory Messages Midlike Auxiliary Module Change to @ No @ Yes - Multike Appearance Groups Multike Appearance Groups No Generic SIP Phone None Auxiliary Multike	Maintenance Logs - Error		Device Id: - 12 -
Maintenance Logs - Warning Id Device Type Module MuXIML Applications Image: Device Type Module Device Type: Change to vice Type Auxillary Module: Mutiline Advisory Messages Image: Device Type None Auxillary Module: Device Type: Auxillary Module: Image: Device Type Auxillary Module: Image: Device Type Auxillary Module: Image: Device Type <	Maintenance Logs - Info	Device Hot Auxiliary	Hot Desk User: Change to 😪 💿 No 🔘 Yes -
Multiline Advisory Messages ¹² ¹⁰ ¹² ¹⁰ ¹² ¹¹	Maintenance Logs - Warning MiXML Applications	ld User Module	Device Type: Change to V Generic SIP Phone V -
Multiline Apparance Groups 13 No Generic SIP Phone None Multiline DNI Sets 14 No Generic SIP Phone None Multiline Set Keys 15 No Generic SIP Phone None Multiline Set Keys 15 No Generic SIP Phone None Network Services Units 16 No Generic SIP Phone None Network Services Units 15 No Generic SIP Phone None Network Services Units 16 No Generic SIP Phone None Network Services Units - - Confirm User PIN: Change to wintown of West - Network Zones - - - ACD Enabled: Change to wintown of West - Network Zones - - - Multicall - - NoNiOPS Circus - - - - Multicall - Page Groups - - - - - - ONSIOPS Circus - - - - - Page Groups -	Multiline Advisory Messages	🧀 12 No Generic SIP Phone None	Auxiliary Module: Change to 🔽 None 🔍 -
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Multiline Set Keys # 15 No Generic SIP Phone None Network Elements # 16 No Generic SIP Phone None Network Services Units 16 No Generic SIP Phone None Network Services Units 16 No Generic SIP Phone None Network Services Units 16 No Generic SIP Phone None Network Zone Topology # A ACD Enabled: Change to # Image to # Network Zone Topology # Image to # Image to # Image to # Image to # Network Zone Topology # Image to # Image to # Image to # Image to # Network Zone Society Tops Image to # Image to # Image to # Image to # Page Groups Image to # Page Groups Image to # Page Groups Image to # Page Groups Image to # Image to # Image to # Image to # <td>Multiline DNI Sets</td> <td>14 No Generic SIP Phone None</td> <td>Local-only DN: Change to V</td>	Multiline DNI Sets	14 No Generic SIP Phone None	Local-only DN: Change to V
Network Elements Interview 15 No Generic SIP Phone None Network Services Units Confirm User PIN: Change to Interview - Network Zone Topology Interview ACD Enabled: Change to Interview - Network Zone Topology Interview Interview - Multicall Interview Network Zone Source Topology Interview Interview - Multicall Interview NoNIOPS Circuit Descriptors Page Groups Interview Save Cancel	Multiline Set Keys 🥔	15 No Generic SIP Phone None	User PIN: Change to 🗸
Network Synchronization ACD Enabled: Change to v o Yes Network Zones Line Type: Multicall ONS/OPS Circuit Descriptors Ves Ves	Network Elements 🧀 Network Services Units	16 No Generic SIP Phone None	Confirm User PIN:
Network Zone Topology & Line Type: - Mutticall - Network Zones ONSIOPS Circuit Descriptors Page Groups Preview Save Cancel	Network Synchronization		ACD Enabled: Change to 🗸 💿 No 🔾 Yes -
Network Zones ONSIOPS Circuit Descriptors Page Groups Preview Save Cancel	Network Zone Topology 🧬		Line Type: - Multicall -
Page Groups V Save Cancel	Network Zones		
	Page Groups	<	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			^	Disable Executive Busy Override Tone:	No	◯ Yes	~
				Disable Send Message:	⊙ No	OYes	
				Display ANI/ISDN Calling Number Only:	No	OYes	
Class Of Service Number:	4	-		Display ANI/DNIS/ISDN Calling/Called Number:	ONO	Yes	
Comment:	SIP Sets			Display Caller ID on multicall/keylines:	 No 	OYes	
Account Code Verified:	No	O Yes		Display DNIS/Called Number Before Digit	⊙ No	OYes	
ACD Make Busy on Login:	No	○ Yes		Modification:	1000		
ACD Silent Monitor Accept:	No	○ Yes		Display Dialed Digits during Outgoing Calls:	 No 	OYes	
ACD Silent Monitor Allowed:	No	○ Yes		Display Held Call ID on Transfer:	 No 	OYes	
ACD Silent Monitor Notification:	No	○ Yes		Display Transfer Destination on Recall:	 No 	OYes	
Allow Directed Call Pickup Of Attendant Call:	O No	Yes		Do Not Disturb:	O No	Yes	
ANI/DNIS/ISDN Number Delivery Trunk:	O No	Yes	-	Do Not Disturb - Access to Remote Phones:	O No	Yes	1
Auto Answer Allowed:	O No	Yes		Do Not Disturb Permanent:	No	O Yes	
Brokers Call:	No	○ Yes		Emergency Call Notification - Audio:	 No 	O Yes	
Busy Overide Security:	 No 	○ Yes		Emergency Call Notification - Visual:	 No 	O Yes	
Call Announce Line:	 No 	○ Yes		Enable Call Duration Limit on External Calls:	 No 	O Yes	
Call Forwarding Accept:	O No	Yes		Enable Call Duration Limit on Internal Calls:	No	O Yes	-
Call Forwarding (External Destination):	O No	Yes		Executive Busy Override:	No	O Yes	
Call Forwarding (Internal Destination):	O No	 Yes 		External Trunk Standard Ringback:	 No 	OYes	
Call Forward Override:	 No 	O Yes		Flexible Answer Point:	 No 	OYes	
Call Forwarding Reminder Ring (CFFM and	No	O Yes		Follow 2nd Alternate Reroute for Recall to Busy ACD Agent:	 No 	OYes	
Call Hold:	ONe	(a) Vac		Forced Verified Account Code:	(No	Yes	
Call Hold Remote Retrieve:	ONo	() Yes		Forced Non-Verified Account Code:	No	O Yes	
Call Hold . Retrieve with Hold Key:	O No	@ Ves		Group Call Forward Follow Me Accept:	No	O Yes	
Call Park Allowed To Park:	ONe	Ves		Group Call Forward Follow Me Allow:	() No	O Yes	
Call Pickup Dialed Accent:	ONo	() Yes		Group Page Accept:	(No	O Yes	
Call Pickup Directed Accept	O No	© res		Group Page Allow:	() No	O Yes	
Call Privacy:	() No	⊙ Yes		Group Presence Control:	() No	O Yes	
Call Reroute after CEEM to Busy Destination:	() No	OVes		Group Presence Third Party Control:	() No	O Yes	
Call Waiting Swap:	() No	OVer		Handset Volume Adjustment Saved:	() No	O Yes	
Called Party Features Override	() No	O Yes		Handsfree AnswerBack Allowed:	No	O Yes	
Calling Name Display - Internal - ONS:	O No	O Yes		HCI/CTI/TAPI Call Control Allowed:	O No	() Yes	
Calling Number Display - Internal - ONS:	ON	() Yes		HCI/CTI/TAPI Monitor Allowed:	O No	() Yes	
Calling Party Name Substitution:	() No	O Ves		Head Set Switch Mute:	No	O Yes	
Campon Tone Security / FAX Machine:	() No	O Yes		Hot Desk External User - Answer Confirmation:	O No	() Yes	
Check COR after PSTN Dial Tone:	() No	O Vas		Hot Desk External User - Display Internal	No	OYes	
Clear All Features Remote:	@ No	OVes		Calling ID:			
Conference Call:	O No	() Yes		Hot Desk External User - Permanent Login:	 No 	OYes	
COV/ONS/E&M Voice Mail Port:	@ No	O Yes		Hot Desk Login Accept:	() No	OYes	
DASS II OLI/TLI Provided:	() No	O Yes		Hot Desk Remote Logout Enabled:	() No	OYes	
Dialled Night Service:	O No	() Yes		Hotel Room Monitor Setup Allowed:	 No 	OYes	
Direct Voice Call - Accept:	() No	O Vas		Hotel Room Monitoring Allowed:	 No 	OYes	
Direct Voice Call - Allow:	() No	O Ves		Allowed:	No	OYes	
Direct Voice Call - Maximize Volume:	() No	O Vac		Hotel/Motel Room Remote Wakeup Call	() No	OVes	
Disable Call Reroute Chaining On Diversion:	() No	OVes		Allowed:	0110	0103	
Disable Conference Join Tone:	() No	O Vos		Individual Trunk Access:	O No	Yes	
Disable Executive Busy Override Tope	() No	O Yes		Local Music On Hold source:	 No 	OYes	
Disable Send Message:	No	O Yes	~	Loudspeaker Pager Override:	O No	Yes	~
		Count				-	
	Save	Cancel				Save Cancel	

A DESCRIPTION OF A DESCRIPTION OF A DESCRIPTION OF A DESCRIPTION OF A	1.000				
Local Music On Hold source:		O Yes 📤	Record-A-Call - Start Automatic Incoming Call	 No 	O Yes
Loudspeaker Pager Override:	O No	Yes	Recording:		
Loudspeaker Pager Equivalent Zone Override Security:	⊙ No	O Yes	Record-A-Call - Start Automatic Outgoing External Call Recording:	⊙ No	O Yes
Maintain Ringing Party During Recall:	No	O Yes	Record-A-Call - Save Recording on Hang-up:	No	O Yes
Message Waiting:	O No	Yes	Recorded Announcement Device:	No	○Yes
Message Waiting Audible Tone Notification:	No	O Yes	Recorded Announcement Device - Advanced:	 No 	○ Yes
Message Waiting Deactivate On Off-Hook:	O No	Yes	Redial Facilities:	O No	Yes
Message Waiting - Disable Ringing Lamp Notification:	⊙ No	O Yes	Return Disconnect Tone When Far End Party Clears:	⊙ No	O Yes
Message Waiting Inquire:	O No	Yes	Ringing Line Select:	No	○Yes
Multiline Set Loop Test:	No	○ Yes	SC1000 Attendant Basic Function Key:	No	○Yes
Multiline Set Message Center Remote Read	 No 	○ Yes	SMDR External:	No	O Yes
Allowed:		-	SMDR Internal:	No	O Yes
Multiline Set Music:	 No 	O Yes	Speak@Ease Preferred:	No	OYes
Multiline Set On-hook Dialing:	O No	 Yes 	Suppress Delivery of Caller ID Display	No	O Yes
Multiline Set Phonebook Allowed:	ONo	Yes	between Sets:	2000 C	and the second sec
Multiline Set Voice Mail Callback Message Erasure Allowed:	⊙ No	O Yes	Suppress Delivery of Caller ID Display between Sets - Override:	⊙ No	O Yes
Music on Hold on Transfer:	No	○ Yes	Suppress Display Of Account Code Numbers:	No	○Yes
Name Suppression on outgoing Trunk Call:	No	O Yes	Suppress Redial Display:	No	○ Yes
Non DID Extension:	No	O Yes	Suppress Simulated CCM after ISDN Progress:	No	○Yes
Non-Prime Public Network Identity:	 No 	O Yes	Third Party Call Forward Follow Me Accept:	No	O Yes
Non Verified Account Code:	O No	Yes	Third Party Call Forward Follow Me Allow:	No	○Yes
Off-Hook Voice Announce Allowed:	⊙ No	O Yes	Timed Reminder Allowed:	O No	
ONS CLASS/CLIP: Message Waiting	No	O Yes	Trunk Calling Party Identification:	O No	Yes
Activate/Deactivate:	12		Trunk Flash Allowed:	 No 	OYes
ONS CLASS/CLIP: Set:	No	OYes	Two B-Channel Transfer Allowed:	No	O Yes
ONS CLASS/CLIP: Visual Call Waiting:	ONo	Yes	Use Held Party Device for Call Re-routing:	O No	⊙ Yes
ONS/OPS Internal Ring Cadence for External	No	OYes	Use Called Party Call Hold Timer:	⊙ No	OYes
Originator's Display Update In Call	() No	O Vas	Voice Mail Softkey:	 No 	OYes
Forwarding/Rerouting:	ONO	0 165	Account Code Length:	12	
Override Interconnect Restriction on Transfer:	 No 	O Yes	After Answer Display Time:		
Pager Access All Zones:	O No	Yes	Alter Allswer Display Time.		
Pager Access Individual Zones:	 No 	O Yes	Answer Plus Delay To Message Timer:	20	1
PC Port On IP Device - Disable:	 No 	O Yes	Answer Plus Expected Off-hook Timer:	30	
Phonebook Lookup - Default to User Location:	 No 	◯ Yes	Answer Plus Message Length Timer:	10	
Phonebook Lookup - Display User Location:	 No 	◯ Yes	Answer Plus System Reroute Timer:	0	
Phone Lock:	No	O Yes	Attendant Busy Out Timer:	10	i =
Privacy Released:	No	O Yes	Auto Common Times		1
Public Network Access via DPNSS:	O No	 Yes 	Auto Campon Timer:		L L
Public Network Identity Provided:	No	O Yes	Busy Tone Timer:	30	
Public Network To Public Network Connection Allowed:	 No 	O Yes	Call Duration:	10	ļ
Public Trunk:	No	O Yes	Call Duration Forced Cleardown Timer:	U	1 -
R2 Call Progress Tone:	No	OYes	Call Forward - Delay:	0	
Recall If Transferred to Original Call	⊙ No	O Yes	Call Forward No Answer Timer:	15	
Record-A-Call Active:	() No	O Yes	Call Hold Timer:	30	
Record-A-Call - Start Automatic Incoming Call	⊙ No	O Yes	Call Park Timer:	180	
Recording:	No.	~	Campon Recall Timer:	10	*
	Save	Cancel		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.

6 https://192.168.101.11/?logoutParer	ntSessionId=0 · Sipint2 ·	🥭 Webpage Dialog		
Group 'lab' Alarm Status:	Major	SIP Device Capabilities		
Sipint2 View Alphabetically 💌 💞 SDS Share	SIP Device Capabilities o			
Ring Groups 🧬 🗾 🔥	SIP Device Capabilitie	SIP Device Capabilities Number: Comment:	15 Polycom 7000	
SDS Form Comparison SDS Form Sharing 🥔 SDS Shared Data Updates - All SDS Shared Data Updates - System SDS Shared Data Updates - User Single Line DNI Sets	Find a field named: SIF	Call Routing and Administration Options Outbound Proxy Server: Replace System based with Device based In-Call Features: Allow MWI Notifications without Subscription: Enable Digit Collection In Busy Or Alerting State:	 ○ No ○ No ○ No 	 Yes Yes Yes
Single Line IP Sets 💞 SIP Device Capabilities SIP Peer Profile SIP Peer Profile Assignment by Incor SMDR Options SNIMP Configuration SNIMP Trap Forwarding Software Logs - All Software Logs - Error Software Logs - Info Software Logs - Warning	Change Copy << < > >> SIP Device Capabilitients SIP Device Capabilitients 16 17 18	SDP Options Allow Device To Use Multiple Active M-Lines: Allow Using UPDATE For Early Media Renegotiation: Force sending SDP in initial Invite message: Limit to one Offer/Answer per INVITE: Prevent SDP Renegotiation If Peer Is On Hold: Prevent the Use of IP Address 0.0.0.0 in SDP Messages: Renegotiate SDP To Enforce Symmetric Codec: Repeat SDP Answer If Duplicate Offer Is Received: Suppress Use of SDP Inactive Media Streams:	 No 	 ○ Yes
Spanning Tree Station Attributes 🥔 Suites System Access Points	19	Signaling and Header Manipulation Minimum Registration Period: Session Timer:	300 90	
System Account Codes		Allow Display Update:	No No	OYes
System Audio Files Update		Eail PEEED To Keep Call Active On Mid Call Feature:	ONo	• Yes
System Capacity System Diagnostics Reporting System IP Ports		Require Reliable Provisional Responses on Outgoing Calls: Use P.Asserted Identity Header:	⊙ No	O Yes
System IP Properties System Options System Ports System Speed Calls Telephone Directory		Distinctive Ring Tones Enable Distinctive Ringing: Internal Ring:	No http://www.notused	⊙ Yes
Tenants Traffic Report Options Trunk Attributes		External Ring: Callback Ring:	<http: td="" www.notused<=""><td>*</td></http:>	*
Trunk Circuit Descriptor - CO			Save	Cancel

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.

1. Enter the number of records to	change: 1		
2. Define the Change Range Prog	ramming Pattern:		
Field Name	Change action	Value to change	Increment by
Number:	-	2305	5-1
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 🗸	💿 Default 🛛 Manual	1.2
Zone ID:	Change to 🐱		
SIP Device Capabilities:	Change to 💉	15	

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.

🖨 https://192.168.101.11/?logoutParentSessionId=O - Sipint2 - Mitel Communications Director - Windows Internet Explorer 📃 🗖 🔀							
Group 'lab' Alarm Status:	🕕 Major	Message Board About Help Logout					
Sipint2 View Alphabetically 🗸 🥠 SDS Share	Multiline Set Keys on Sipint2 DN to search 💌	Show form on Sipint2 (Login Node) 🗸 Go ↓					
Local-only Directory Number List and Locations and Login/Logout Audit Logs Logs - All Maintenance/Software Loudspeaker Paging Maintenance Commands	Multiline Set Keys Search: Search Scope: Sipint2 Admin Group Find a field named: Directory Number that has a value of 2310 Search						
Maintenance Logs - All Maintenance Logs - Error Maintenance Logs - Info Maintenance Logs - Warning	Change Print	Import Export Data Refresh					
MiXML Applications	Directory Number Ring Type Prime Line Type	Name					
Multiline Advisory Messages	2310 Ring Multicall	Polycom,7000					
Multiline Appearance Groups Multiline DNI Sets	2313 Ring 2501 Ring						
Multiline IP Sets 🥔 Multiline Set Keys 🚅 Network Elements 🎺	2502 Ring 2503 Ring						
Network Services Units Network Synchronization Network Zone Topology 🧬	< Page 1 of 19 > Button Number: Label: Coov Keys Chance Membe Line Type:	2 Line 2 Mutticall					
Network Zones ONS/OPS Circuit Descriptors	Programmable Keys Button Butt	2310					
Peripheral/DSU Units Personal Ring Groups 🥔	Number Label Line Type UKL Number Ring Type: # 2 Line 2 Multicall 2330 MiXML Application Feature	Ring V re: Not Assigned V					
Personal Speed Call Allocation	A Signed Prione Application Feature A for a signed	e:V					
Personal Speed Calls	4 Not Assigned						
Port Forward Table	er b NotAssigned e 6 NotAssigned	Save Cancel					

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.

🖉 https://192.168.101.11/?logoutParentSessionId=0 - Sipint2 - Mitel Communications Director - Windows Internet Explorer										
Group 'lab' Alarm Status:	Major							Message B	pard Abou	it Help Logout
Sipint2 View Alphabetically View Alphabetically	Call Rerouting F	First Alternativ	es on Sipint	2 [)N to search	~	Sh	ow form on	Sipint2 (Login N	Vode) 🗸 Go 🗸
ARS Maximum Dialed Digits ARS Node Identities	Change Page 1	Change Pag of 23	e Chang	je All Cle	ar	Go to:	Print	Import	Export value:	Data Refresh Go
ARS Route Plans	Call Reroutin	ng First Alter	natives							
ARS Routes Associated Directory Numbers 🍻 Backup	First Alternative Number 1	Busy / DND DID Normal	Busy / DND TIE Normal	Busy / DND CO Normal	Busy / DND Int Normal	No Answer DID Normal	No Answer TIE Normal	No Answer CO Normal	No Answer Int Normal	Directory Number
Bearer Canabilities	2	Normal	Normal	Normal	Normal	Normal	Normal	Normal	Normal	
Call Park	3	This	This	This	This	This	This	This	This	2305
Call Recognition Service 🦨	4	This	This	This	This	Normal	Normal	Normal	Normal	2028
Call Rerouting	5	Normal	Normal	Normal	Normal	Normal	Normal	Normal	Normal	
Call Rerouting Always Alternatives	6	This	This	This	This	This	This	This	This	2950
Call Rerouting First Alternatives	7	This	This	This	This	This	This	This	This	2900
Calling Line ID Restriction	8	This	This	This	This	This	This	This	This	2007
Card Assignment	9	Normal	Normal	Normal	Normal	This	This	This	This	2900
CESID - Default	10	This	This	This	This	This	This	This	This	2513
CESID Assignment	11	Normal	Normal	Normal	Normal	This	This	This	This	2050
CESID Logs	12	This	This	This	This	Normal	Normal	Normal	Normal	2050
Class of Restriction Groups	13	This	This	This	This	This	This	This	This	*7770
Class of Service Options	14	This	This	This	This	This	This	This	This	2113
Cluster Elements 🧬 CO Tone Detection	15	This	This	This	This	This	This	This	This	7751
Console someys ar Controller Module Configuration Controller Registry CPN Substitution Current Bandwidth Statistics Date and Time ar Default Account Codes										

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.

08-5159-00020_5

🍘 https://192.168.101.11/?logoutParentSessionId=0 - Sipint2 - Mitel Communications Director - Windows Internet Explorer							
Group 'lab' Alarm Status:		r».				Message Board	About Help Logout
Sipint2 View Alphabetically SDS Share	Call Rerou	ting on Sipint2		DN to search	•	Show form on Sipint2 (Login Node) 💌 Go 💵
ARS Maximum Dialed Digits ARS Node Identities ARS Route Lists ARS Route Plans ARS Routes Associated Directory Numbers Backup Bandwidth Management Bearer Capabilities	Chang Pag Call Rer Number 2302 2304 2305	e Change Page e 5 of 10 > call Rerouting . Day 1 1	Call Rerouting - Night1 1 1	Call Rerouting - Night2 1 1	Print Go to: Number Call Rerouting DND Type All All	Import Exp value Call Rerouting - 1st Alt. 1 1 1	ort Data Refresh e: 2310 Go Call Rerouting - 2nd Alt. 1 1
Call Park Call Recognition Service Call Rerouting Call Rerouting Always Alternatives Call Rerouting First Alternatives Call Recourting Second Alternatives	2306 2307 2310 2313 2501	1 1 1 1	1 1 1 1 1	1 1 1 Webpage Dialog	All All All All	1 7 7	1 1 1
Call Refound Second Americatives Calling Line ID Restriction Card Assignment CESID - Default CESID Logs Class of Restriction Groups Class of Service Options Cluster Elements 🎺 CO Tone Detection Console Softkeys 🗳 Controller Module Configuration Controller Registry CPN Substitution	2502 2503 2504 2510 2511 2513 2514 2521 2523 2540 2552	1 1 1 1 1 1 1 1 1 1	1 Ni 1 Ni 1 Ca 1 Ca 1 Ca 1 Ca 1 Ca 1 Ca 1 Ca 1 1	Call Rerouting umber: all Rerouting - Day: all Rerouting - Nigh all Rerouting - Nigh all Rerouting DND T all Rerouting - 1st A all Rerouting - 2nd <i>i</i>	t1: [t2: [ype:] Jt.: [Alt.:]	1310 1 1 AII V 7 1	
Current Bandwidth Statistics Date and Time 🌧 Default Account Codes	2571	1	1	i iii		Save Car	ncel

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 "**PicmSplp**" with password "**PicmSplp**" (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 "**PicmSplp**" 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 "**PicmSplp**" with password "**PicmSplp**" 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 0000000000.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=
"phonel_0004f223413d.cfg, sip.cfg"</pre>

If per-phone master configuration file **<MAC-address>.cfg** is unavailable in FTP root folder, then the default master configuration file **0000000000.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 "***xxxxx**" 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]xxxxxxxx|[2-
9]xxxxxxxxx|*xxxxxxx|[2-9]xxxT" dialplan.digitmap.timeOut="3|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:

<pre>sec.srtp.enable="1" -</pre>	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 <u>Multiline Set Keys</u>.
```

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" - A	Actually this value could any number and it triggers
	SUBSCRIBE request sent to 3300ICP
msg.mwi.1.callBackMode="contac	tr " - This parameter is needed when user presses the key
	If 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 e	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:

Dasi	nboard	ICPs D	evices Connec	tors Metrics	Advanced	Resiliency		
Locatio	n: ICP li	st						
alcom		IRC administrat	ive inteface. From h		all accepts of the	MRC's hohouigurs. About pro a	uprious tabs for personin	a different nou
e syste	em. If at	any time you re	quire more informat	ion, click the Help ico	n in the upper-rig	ght corner of the page.	various tabs for accessing	g unterent par
dd an 1	CP							
CP Inf	ormatio	n						
Default for	Default	Nama			Tune	Tastallas Dassus		
Minet		SIDINT1	192.158.101.10	33		Installer Passwol	Modify	Delete
0	0	CLOINTO	102 102 101 11				Mandia.	Delete
	•	SIPINIZ	192.168.101.11	33	00100		Modiry	Delete
0		OT DIALTO	102 160 101 14	22	OO TOD		Modifie	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 Connec	tors Metrics	Advanced	Resiliency	
Location: <u>Connectors</u> / <u>SIP settings</u> / Edit SIP setting	s			
Velcome to the MBG administrative inteface. From he he system. If at any time you require more informat	re you can manage a on, click the Help ico	all aspects of the n in the upper-ri	MBG's behaviour ght corner of the	 Above are various tabs for accessing different parts page.
Summary				
SIP Configuration				
dit SIP options.				
SIP connector enabled	I 🔽			
Default SIP ICP	SIPINT2 V			
Forward unknown messages				
Forward unknown headers				
Send options keepalives				
Heartbeat interval	20			
Gap register	. 🗹			
Set-side registration expiry time	240			
ICP-side registration expiry time	900			
SIP connection log verbosity	Use master setting	y ~		

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**'.

6 https://192.168.101.11/?logoutParer	ntSessionId=0 · Sipint2 ·	🥭 Webpage Dialog		
Group 'lab' Alarm Status:	Major	SIP Device Capabilities		🚔 ıt
Sipint2 View Alphabetically 💌 💞 SDS Share	SIP Device Capabilities o		8 .	
Ring Groups 🖨 💦	SIP Device Capabilitie	SIP Device Capabilities Number: Comment:	15 Polycom 7000	
SDS Form Comparison SDS Form Sharing 🧬 SDS Shared Data Updates - All SDS Shared Data Updates - System	Find a field named: SIF	Call Routing and Administration Options Outbound Proxy Server: Replace System based with Device based In-Call Features: Allow MVI Notifications without Subscription:		O Yes
SDS Shared Data Updates - User Single Line DNI Sets		Enable Digit Collection In Busy Or Alerting State:	O No	⊙ Yes ⊙ Yes
Single Life in Sets of SIP Peer Profile SIP Peer Profile Assignment by Incor SMDR Options SNMP Configuration SNMP Trap Forwarding Software Logs - All Software Logs - Error Software Logs - Info Software Logs - Warning Software Logs - Warning	Change Copy << < > >> SIP Device Capabilitient SIP Device Capabilitient 16 17 18	SDP Options Allow Device To Use Multiple Active M-Lines: Allow Using UPDATE For Early Media Renegotiation: Force sending SDP in initial Invite message: Limit to one Offer/Answer per INVITE: Prevent SDP Renegotiation If Peer Is On Hold: Prevent SDP To Enforce Symmetric Codec: Renegotiate SDP To Enforce Symmetric Codec: Ruppress Use of SDP Inactive Media Streams:	 No 	O Yes O Yes O Yes O Yes O Yes O Yes O Yes O Yes O Yes O Yes
Station Attributes 🥔 Suites	19	Signaling and Header Manipulation Minimum Registration Period: Session Timer:	300	
System Access Fornis		Allow Display Update:	@ No	OVer
System Audio Eilos Lindato		Disable Reliable Provisional Responses:	O No	() Yes
System Audio Files Optiale		Fail REFER To Keep Call Active On Mid-Call Feature:	No	OYes
System Capacity System Diagnostics Reporting		Require Reliable Provisional Responses on Outgoing Calls:	⊙ No	OYes
System IP Properties		Use P-Asserted Identity Header:	○ No	
system Options System Ports System Speed Calls Telephone Directory 🖨 Tenants		Distinctive Ring Tones Enable Distinctive Ringing: Internal Ring: External Ring: Callback Ring:	No <http: www.notuser<br=""><http: www.notuser<br=""><http: td="" www.notuser<=""><td>⊙ Yes</td></http:></http:></http:>	⊙ Yes
Trunk Attributes Trunk Circuit Descriptor - CO			Save	Cancel

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.

https://192.168.101.11/?logoutParen	tSessionId=0 - Sipint2 - Mite	l Communications Directo	r - Windows Internet Explorer	
Group 'lab' Alarm Status:	() Major			Message Board About Help Logo
Sipint2 Yiew Alphabetically 👻 🏾 🎺 SDS Share	Multiline Set Keys on Sipint2		DN to search	Show form on Sipint2 (Login Node) 💌 Go 🗸
Local-only Directory Number List e a Locations a Login/Logout Audit Logs Logs - All Maintenance/Software Loudspeaker Paging Maintenance Commands	Multiline Set Keys Search Search Scope: Sipint2 Find a field named: Director 	:) Admin Group y Number 💙 that has a value	e of 2310 Search]
Maintenance Logs - All Maintenance Logs - Error Maintenance Logs - Info Maintenance Logs - Warning	Change		Pri	nt Import Export Data Refresh
MiXML Applications	Directory Number	Ring Type	Prime Line Type	Name
Multiline Advisory Messages	2310	Ring	Multicall	Polycom,7000
Multiline Appearance Groups	2313	Ring	Webpage Dialog	
Multiline DNI Sets	2501	Ring	Webpage Dialog	
Multiline IP Sets 🧬	2502	Ring	Programmable Keys	
Multiline Set Keys 🦨 Network Elements 🏟	2503	Ring		
Network Services Units Network Synchronization Network Zone Topology 🖨 Network Zones	< Page 1 of 19 >	Copy Keys Chang	e Membe URL:	2 Line 2 Mutticall
ONS/OPS Circuit Descriptors	Programmable Keys		Button Directory Numbe	er: 2310
Page Groups	Number Label	Line Type URL N	utton Dir Ring Type:	Ring 🗸
Peripheral/DSU Units	🧀 2 Line 2	Multicall	310 MiXML Application Feat	ture: Not Assigned 🗸
Personal Ring Groups 🧬		Not Assigned	Phone Application Feat	ure:
Personal Speed Call Allocation	e . 3	Norhasiyileu		
Personal Speed Calls	4 7 4	Not Assigned		
Pickup Groups 🧬	🧬 5	Not Assigned		Save Cancel
Port Forward Table			1	Sare cancer

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)

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