MITEL – MSA 3PPV

Technical
Configuration Notes

Configure the 5000 CP for use with the Polycom Spectralink 8400 Series SIP Wireless device

SIP CoE 11-4940-00177

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NOTICE

TRADEMARKS

Mitel Technical Configuration Notes – Configure the Mitel 5000 CP for use with the Polycom Spectralink 8400 Series SIP Sets
April 2012, 11-4940-00177_2

Overview

This document provides a reference to Mitel Authorized Solutions Providers for configuring the Mitel 5000 CP to host the Polycom Spectralink 8400 Series. 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	September 2011	Initial Interop with Mitel 5000 CP 5.0 SP1 and the Polycom Spectralink 8400 Series
2	April 2012	Documentation Update

Interop Status

The Interop of the Polycom Spectralink 8400 Series has been given a Certification status. This device will be included in the SIP CoE Reference Guide. The status the Polycom Spectralink 8400 Series achieved is:

APPROVED	Reserved for MSA Gold Preferred members only, this rare classification is reserved for key strategic components of our portfolio for which Mitel assumes the full responsibility for support, acting as the interface between the customer and the 3rd party as necessary.

NOTE: Polycom asserts that the 8400 series firmware is compatible with several endpoints. Please refer to Appendix A

Software & Hardware Setup

This was the test setup to generate a basic SIP call between the Polycom Spectralink 8400 Series SIP device and the 5000 CP.

Manufacturer	Variant	Software Version
Mitel	5000 CP	5.0 SP1
Polycom	Spectralink 8400 Series	4.0.0.15769
Mitel	5330 SIP Sets	4.1.0.2
Mitel	5330 IP Sets	4.1.0.2
Polycom	8400 Series VoWLAN handsets	UCS 4.0.0 15769

Tested Features

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

Feature	Feature Description	Issues
Basic Call	Making and receiving a call	☑
DTMF Signal	Sending DTMF after call setup (i.e. mailbox password)	☑
Call Hold	Putting a call on hold	☑
Call Transfer	Transferring a call to another destination	☑
Call Forward	Forwarding a call to another destination	☑
Conference	Conferencing multiple calls together	☑
Redial	Last Number Redial	☑
MWI	Message Waiting Indication	☑
T.38 Fax	Fax Messages	Not Applicable
Video	Video Capabilities	Not Applicable

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

Device Limitations

This is a list of problems or not supported features when the Polycom Spectralink 8400 Series SIP device is connected to the Mitel 5000.

Feature	Problem Description
Messages Menu	MWI is supported but the Mitel 5000 does not support sending the message count. Therefore if you check the messages menu it will indicated 0 new messages. **Recommendation:** Contact Mitel support for updates on support.
Do Not Disturb Camp-on	Does not campon because it is not using the 5000 feature codes. But DND does work **Recommendation:** Please contact Mitel technical support and reference defect number MN00349199 for updates.
Registration using Hostname	The 5000 CP currently does not support registration of SIP endpoints by using the hostname of the 5000. **Recommendation:** Use the IP address of the 5000 for registration.

The following is a List of System features and Phone features that are available on the 5000 CP and whether or not they are supported with the Polycom Spectralink 8400 Series:

System Feature Compatibility Matrix

System Feature	Description	Supported	Comments
Attendant	A SIP phone can be configured as a Phone Attendant, Local Primary Attendant, or Primary Attendant. However, the SIP phone display may not indicate an incoming recall (Intercom/Outside Hold, Transfer, or Conference call recall).	✓	
Audio for Calls	A SIP phone can be programmed to have different audio settings for calls that are ringing, on hold, or camp-on to it.	✓	
Caller ID	The 5000 CP uses the Calling Party Name, Calling Party Number, and Emergency Calling Party Number programmed for a SIP phone in a similar fashion as that of any other phone.	✓	
Dynamic Extension Express (DEE)	You can configure a SIP phone as one of the following User Associated Destination types (Softphone, Home IP, or Desk phone). However, you cannot configure a SIP phone as a main extension of the User.	✓	
Extension ID	Extension IDs can be created for SIP phones so that Voice Processor applications can use these Extension IDs for transferring calls to SIP phones in the local node or networked node.	✓	
Extension Lists	SIP phones can be included in IP/Digital Telephone lists. SIP phones are included in system automatic extension lists - All Phones (PP051) and All IP/Digital Telephone (PP052).	✓	
File-Based Music-On-Hold	File Based MOH can be set for calls ringing, camped-on, or on hold at the SIP phone.	✓	
House Phone	Not supported.		
Mailbox	A SIP phone can be programmed to have mailbox(es) and exhibits the same behavior as that of any other phone.	✓	
Off-Node Device	SIP phones can be programmed as an off-node Keyset similar to that of any other phone.	✓	

Peer-to-Peer Audio	Not Supported.		
Phone Feature Codes	Not Supported.		
Ring-In Destination	SIP phones can be programmed as the Ring-In destination for Trunk Groups and Call Routing Tables similar to that of any other IP phone.	✓	
SIP Peer Features	Supported.	✓	
Trunk Access Codes	The only feature codes that are currently supported by SIP phones are: Emergency Call, Outgoing Call, Automatic Route Selection.	✓	
Voice Mail	A SIP phone can be configured to have access to a Voice Mail system. It can use Mitel NuPoint UM, BVM or EM as the Voice Mail system. However, the 5000 CP does not provide display menus or softkeys for SIP phones. 5000 CP supports accepting calls from SIP phones towards Voice Mail applications in similar way as that of any other phones in the system. For BVM and EM Voice Mail systems, the 5000 CP uses the programmed language at the SIP phone to determine the language heard by the SIP phone user.	✓	

Phone Feature Compatibility Matrix

Phone Feature	**Description**		
		Supported	**Comments**
Account Codes	Not Supported.		
Automatic Call Access	Not Supported.		
Background Music	Not Supported.		
Call Logging	**Missed Calls**: Supported	✓	
	Received Calls: Supported.	✓	
	Dialed Calls: Not supported.		
Call Screening	A SIP phone can be programmed to have Call Screening the same way as that of any other phone.	✓	

Call Waiting (Camp-On)	A call from a SIP phone towards any other device in the system may camp-on if the device status is Busy or Offline and the corresponding DID/E&M Receive Busy Instead of Camp-on flag is disabled. The Camp-On flag at the SIP Phone Group determines whether the calls from SIP phones should camp-on in the following scenarios: • If the concurrent call count for a SIP phone reaches the maximum concurrent call count limit, subsequent calls from and towards that SIP phone would either camp-on or get rejected based on the camp-on flag settings. If the concurrent call count limit at the SIP Phone Group-level reaches, any subsequent call from or towards any SIP phone in that group would either camp-on or get rejected based on the Camp-Off flag settings • If a SIP phone could not acquire a Category F Phones license from the system, any calls from or towards that SIP phone would either camp-on or get rejected based on the Camp-On flag settings.	✓	
	Camp-on indications are not supported. Call waiting indicators are supported		
Conference Calls	**Local Feature**: A SIP phone can initiate a local conference call inviting all the held parties on the SIP phone. To the 5000 CP, it would appear as if there are multiple separate calls initiated by the SIP phone, instead of a conference call. The number of parties that can be involved in such a local conference call depends on the capabilities of SIP phone. Also, the maximum number of concurrent call limit of the SIP phone would limit the number of parties that can be part of such a local conference.	✓	
	System Feature: Not supported.		
Configuration Assistant	Not supported. Even though a SIP phone can call Configuration Assistant, it cannot use Configuration Assistant features to change DND, Forwarding and DEE settings. Also, a SIP phone cannot be configured as an administrator; it cannot use Configuration Assistant to change system settings.		
Directory	**Intercom Directory**: Not supported.		
	Speed Dial Directory: Supported for the local feature only.	✓	
	Feature Code Directory: Not supported.		

Do-Not-Disturb (DND)	**Local Feature**: A SIP phone may have local DND settings and might reject the calls when in DND mode. If a user changes local DND settings on the SIP phone, the 5000 CP has no way to know about this change. For example, if a user enables DND on a SIP phone by using the local feature on the phone, calls towards the SIP phone might camp-on because the 5000 CP determines the SIP phone to be busy or not available. While there are calls camped-on or queued for a SIP phone, and a user disables DND by using the local feature on the phone, the 5000 CP would not be notified about this change and the calls towards the SIP phone would continue to be in camp-on or queued state.	✓	Do Not Disturb is supported but campon to the when in DND is not supported.
	System Feature: Not supported.		
DSS/BLF Key	A DSS/BLF key can be programmed for Mitel phones that points to a SIP phone. The assigned key reflects the status of corresponding SIP phone. However, a DSS/BLF key cannot be programmed on a SIP phone.	✓	
Dynamic Extension Express (DEE)	You can configure a SIP phone as one of the following user destination types (Softphone, Home IP, or Desk phone). However, you cannot configure a SIP phone as a main extension of the user.	✓	
Emergency Calls	The 5000 CP allows a SIP phone to make emergency calls. The 5000 CP does not allow normal calls from a SIP phone to go through in the following situations; however, it would allow emergency calls to go through: • The maximum concurrent call limit is reached and a SIP phone is set to be in Busy state. • A SIP phone does not have dynamic binding (active SIP registration) with the system and is set to be in Busy or Offline state. • A SIP phone does not have an active license. • A SIP Phone Group is in Out-of-Service (OOS) operating state. Additionally, a SIP phone can be configured to use Emergency VoIP resources for emergency calls. When a SIP phone places an emergency call, system generates a corresponding Emergency Alarm.	✓	
Group Listen	Not supported.		
Hold Calls	Supported	✓	
Hold Recalls	Not supported.		
Hookflash	Supported for the local feature only.	✓	

Hunt Groups	**UCD Hunt Group Number**: Even though a SIP phone can be a member of UCD Hunt Group, it cannot be an agent or supervisor of a Hunt Group. It can also not use Remove/Replace or DND feature code to stop receiving Hunt Group calls. Also, there may not be display indications on SIP Phone for Hunt Group calls or recalls.	✓	
	ACD Hunt Group Member: Not supported.		
	Recall Destination: Supported.	✓	
	Hunt Group Remove/Replace: Not supported.		
	Announcement Station: Supported.	✓	
	Overflow Station: Supported.	✓	
	Agent Help: Not supported.		
	Station Monitor: Not supported.		
	Group Call Pick Up: Not supported.		
Intercom Calls	**Calls**: SIP phones can make intercom calls, however, the display on SIP phones may not indicate that it is an intercom call (for example, "IC TO …"). Calls towards SIP phones would always ring regardless of the Ring Intercom flags, unless the SIP phone has local features to enable auto answer.	✓	
	Ring Intercom Always: Not supported.		
	Handsfree Feature Code: Not supported.		
	Non-Handsfree Dialing (#): Not supported.		
Intelligent Directory Search	Not supported.		
Manual Forwarding	**Local Feature**: SIP phones may have a local forwarding functionality where it allows user to forward calls to any specified destination. The 5000 CP may not view such calls as forwarded calls.	✓	
	System Feature: Not supported.		
Messages	**Station Messages**: Not supported.		
	Alternate Message Source: Not supported.		
	Silent Messages: Not supported.		

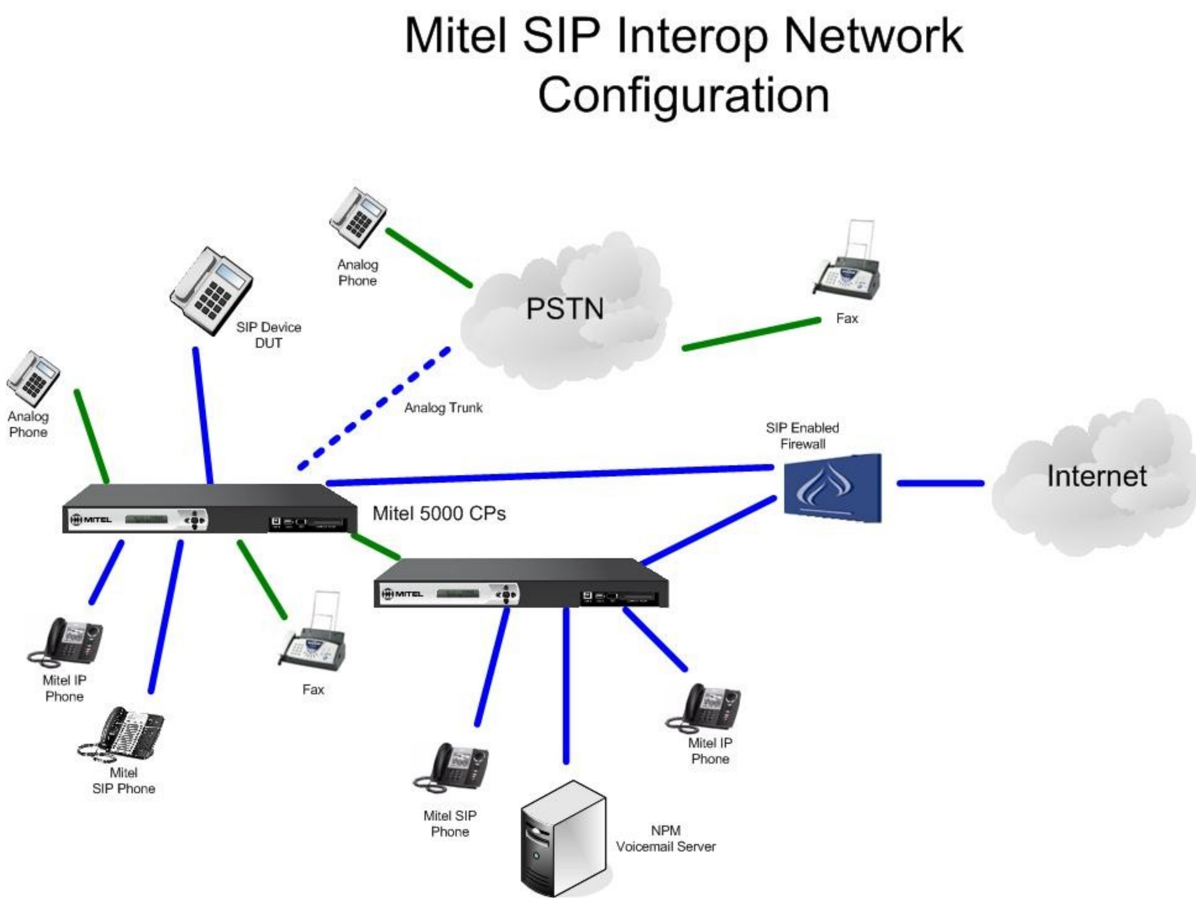
	Message Waiting Indication (MWI): The 5000 CP sends unsolicited MWI notifications towards SIP phones to indicate the current MWI status for that phone. At this point, the 5000 CP does not support accepting MWI subscriptions from SIP phones. If a SIP phone tries to subscribe for “message-summary” event for MWI updates, the 5000 CP rejects the SIP SUBSCRIBE request with the “405 - Method Not Supported” response. If the 5000 CP is using Mitel NuPoint Unified Messaging (UM) as the Voice Mail system, the unsolicited MWI notifications towards SIP phones would only indicate the presence or absence of Voice Mail message(s). Whereas, if BVM or EM is the Voice Mail system, the unsolicited MWI notifications towards SIP phones would indicate the presence or absence of Voice Mail message(s) as well as the message count.	✓	
Microphone Mute	**Local Feature**: A SIP phone may have a local feature to mute/unmute the call. For example, the UCX and IP DECT 5610 has local features to mute/unmute the call.	✓	
	System Feature: Not supported.		
Multilingual Capability	Not supported. However, an administrator can program the language for SIP phone that the callers would hear for Voice Mail and Configuration Assistant.		
Music-on-Hold (MoH)	MOH can be set to Silence, Tick Tone, Ringback, 5000 CP or File-Based MOH for SIP phones.	✓	
Off-Hook Voice Announce	Not supported.		
On-Hook Monitoring	**Local Feature**: SIP phones may allow on-hook monitoring as a local feature.	✓	
	System Feature: Not supported.		
Outside Calls	SIP phones can place and receive outside calls. The displays on SIP phones may not show Mitel 5000 CP formatted outside number as part of caller ID.	✓	
Outgoing Access	A SIP phone can be programmed to have outgoing access the same way as that of any other phone.	✓	
Outgoing Extension	An Outgoing Extension can be programmed for a SIP phone to have outgoing access the same way as that of any other phone.	✓	
Paging	Not supported.		

Phone Feature Codes	Not supported.		
Queue	Not supported.		
Record-A-Call	Not supported.		
Redial Calls	**Local Feature**: SIP phones may provide a redial functionality as a local feature.	✓	
	System Feature: Not supported.		
Redirect Calls	**Local Feature**: SIP phones may provide a redirect functionality as a local feature.	✓	
	System Feature: Not supported.		
Reminder Message	No supported.		
Speed Dials	**System Speed Dial**: Not supported.		
	Station Speed Dial: Supported for the local feature only.	✓	
System Forwarding	Not supported. Even though System Forwarding can be programmed by an administrator, it may not be very useful if a SIP phone user cannot enable or disable it.		
Transfer Calls	A SIP phone can either do a Blind Transfer and/or an Attended Transfer, depending on its capabilities. It can also receive calls transferred by other phones. In other words, in a Transfer Call context, a SIP phone can act as Transferrer, Transferee, or Transfer Target.		
	Transferring Conference Calls: Not Supported		
	Transfer To System Forward	✓	
	Transfer Timers: Supported.	✓	
	Transfer-to-Connect: Not supported.		
	Transfer-to Ring: Supported for the local feature only. The calls transferred to a SIP phone would always be Transfer-to-Ring and cannot be Transfer-to-Hold, even though the transferring phone tried the Transfer-to-Hold option.	✓	
	Transfer-to-Hold: Not supported. Unless the SIP phone has the capability to initiate a Transfer-to-Hold request, whenever it initiates a Transfer Call (Blind or Attendant), it would always be treated as a Transfer-to-Ring request. It uses the local		

	transfer feature to initiate the transfer.		
	Transfer Recalls: A SIP phone can handle the transfer recall. However, the SIP phone display may not indicate that it is receiving a transfer recall. It would appear as if there is a new incoming call and after the user answers that call, it would reconnect the call that is being transferred.	✓	
	Reverse Transfer: Not supported.		

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 Spectralink 8400 Series was configured in our test environment.

For more detailed information on the programming of the Mitel 5000 CP please refer to the Mitel 5000 CP Features and Programming Guide.

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.

5000 CP Configuration Notes

The following steps show how to program a 5000 CP to connect with the Polycom Spectralink 8400 Series Phone.

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 5000 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 5000 CP Programming

- The SIP signaling connection uses UDP on Port 5060.

Software License – SIP Licensing

Ensure that the 5000 CP is equipped with enough Category ‘F’ Phones licenses for the connection of SIP end points. This can be verified within the Software License Feature section form.

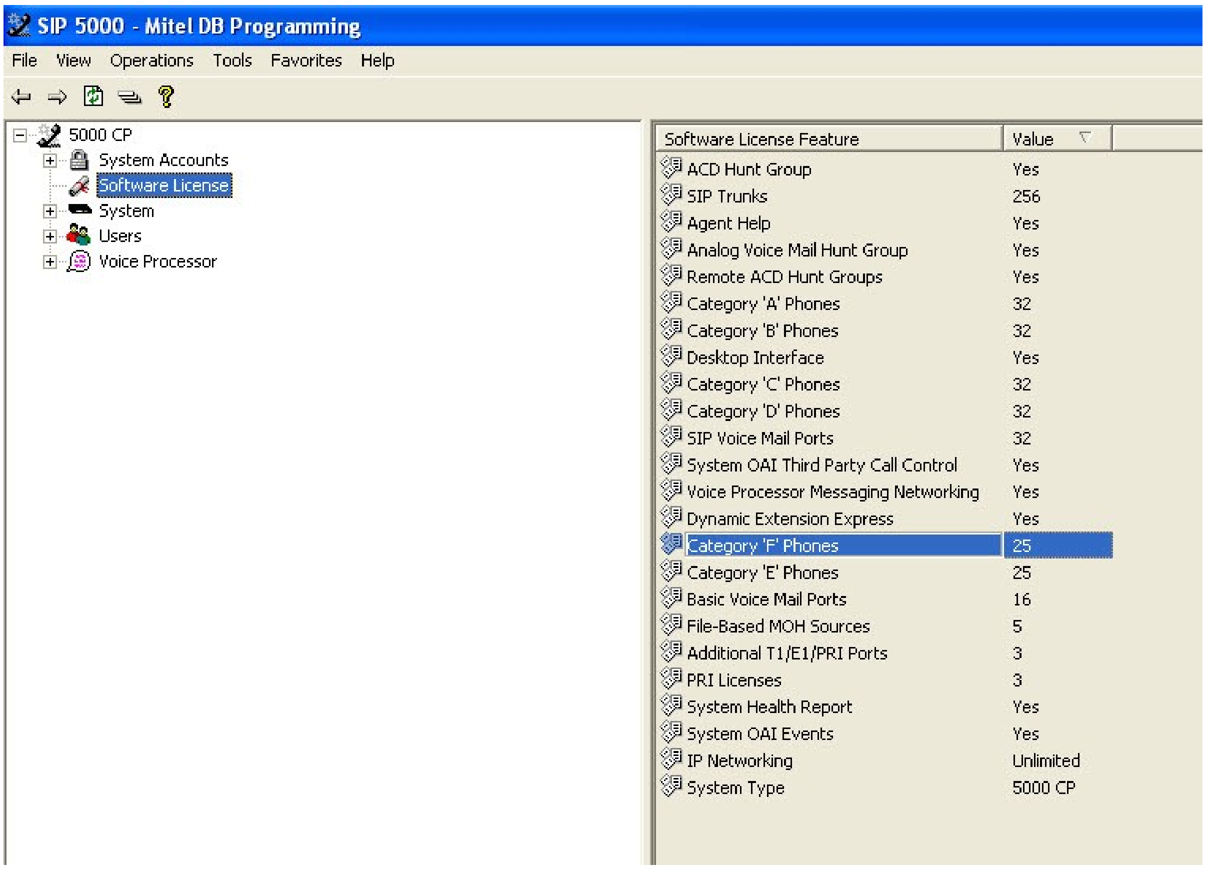


Figure 1 – Software License

Polycom Spectralink 8400 Series SIP Device Configuration

To create an extension for the Polycom Spectralink 8400 Series:

1. Select System – Devices and Feature Codes – **Phones**.

2. Right-click anywhere in the right pane, and then select **Create SIP Phone**. The Create

SIP Phone Extension dialog box appears.

3. Select a starting extension for the phones and the number of extensions.

4. Click **OK**. The system creates a new SIP Phone Group for each of the **Polycom Spectralink 8400 Seriess**. TheSIP Phone groups are created in a stand-alone configuration by default. The associated SIP

Phone Group is displayed in System\Devices and Feature Codes\SIP Peers\SIP Phone Groups

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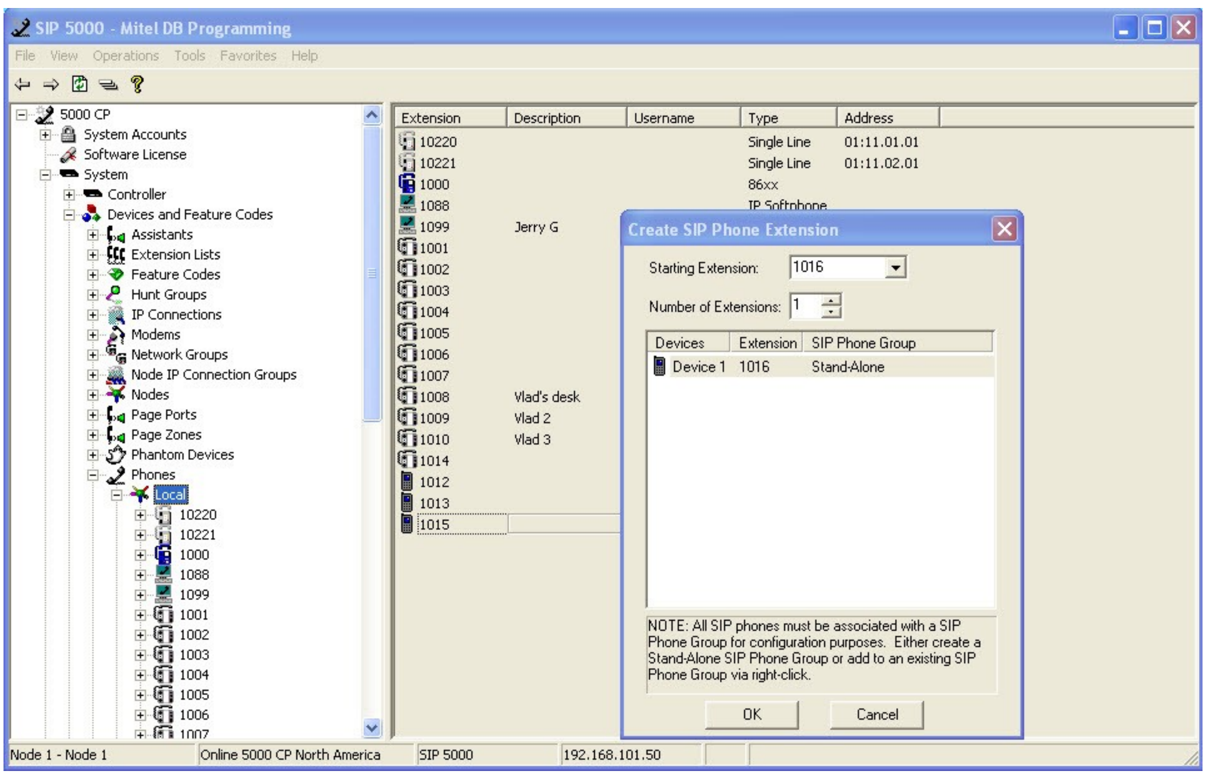

Figure 2 – Create SIP Extension

The Polycom Spectralink 8400 Series was configured as displayed below.

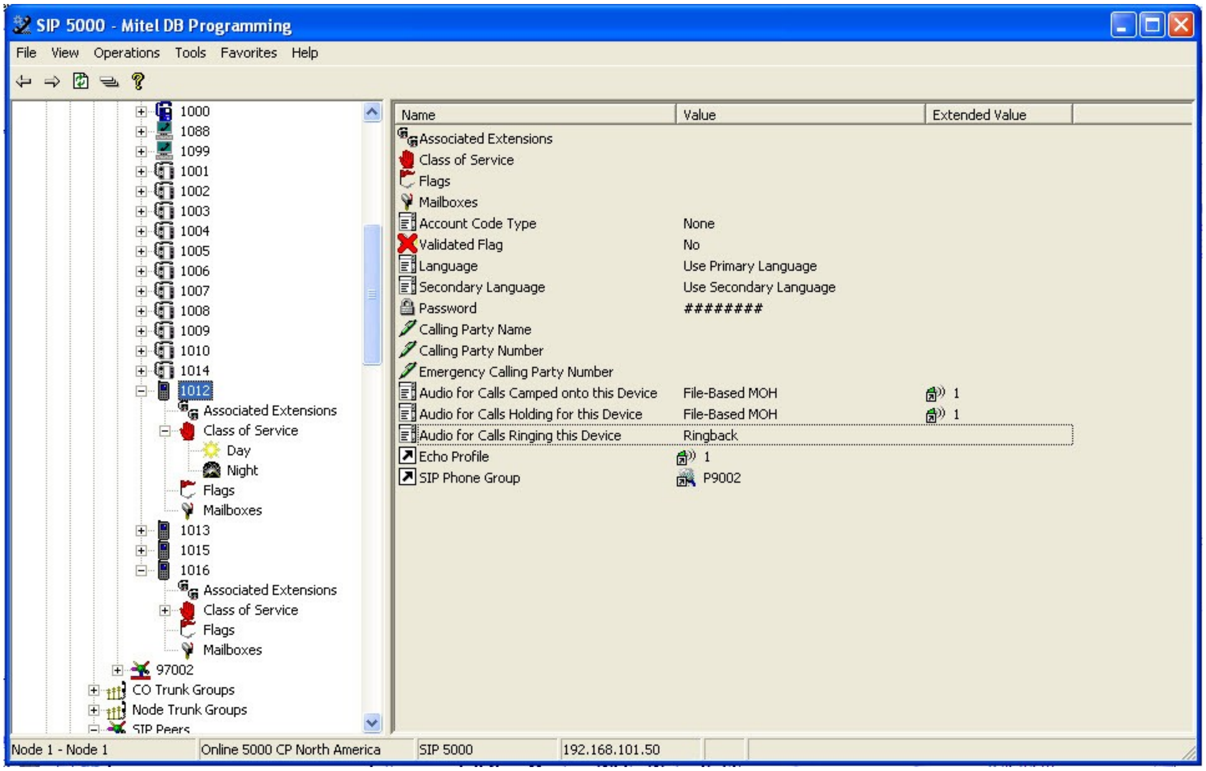

Figure 3 – Polycom Spectralink 8400 Series Configuration

The Password field is for the SIP authentication password and the username is the DN. All other field names should be programmed according to the site requirements or left at default

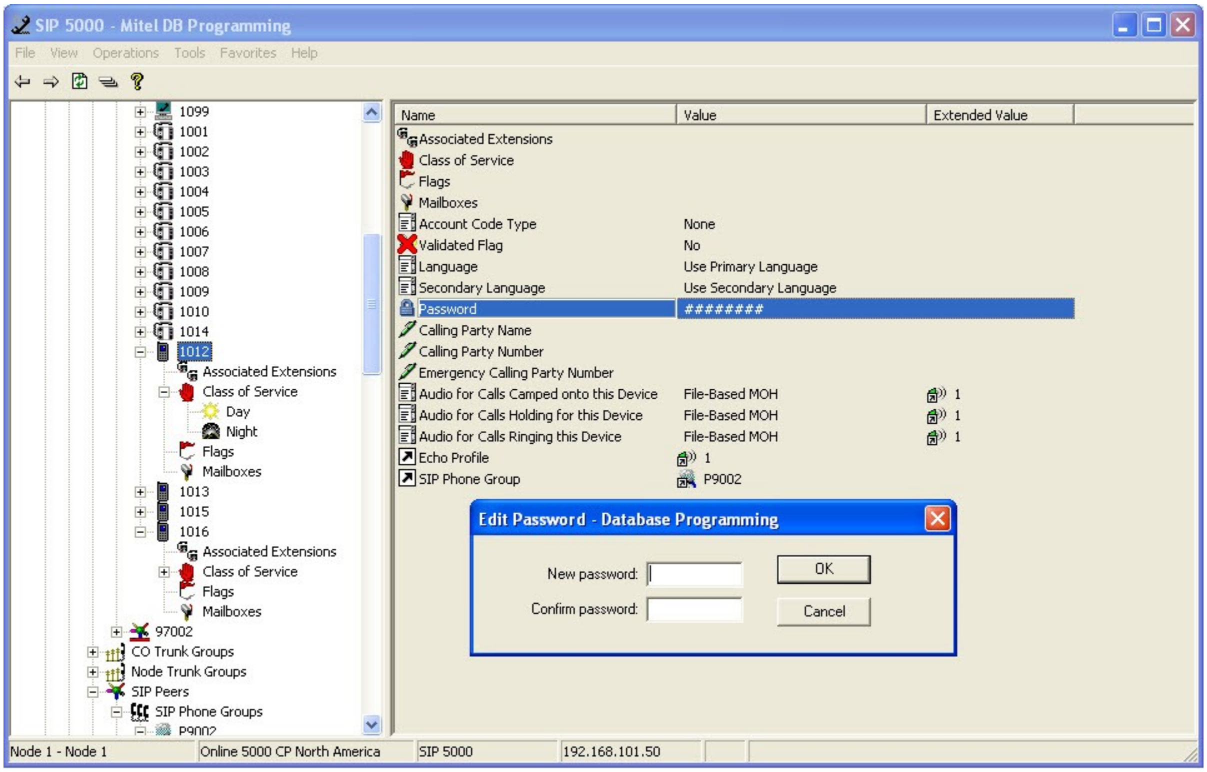

Figure 4 – Password Configuration

SIP Phone Groups

The Polycom 8400 Series can register with the 5000 CP and act as local extensions in the system. To support this feature, DB Programming uses “SIP Phone Groups.” A SIP Phone Group contains a common set of properties for registration that can be shared with either a “stand-alone” SIP Phone Group or multiple SIP Phone Group.

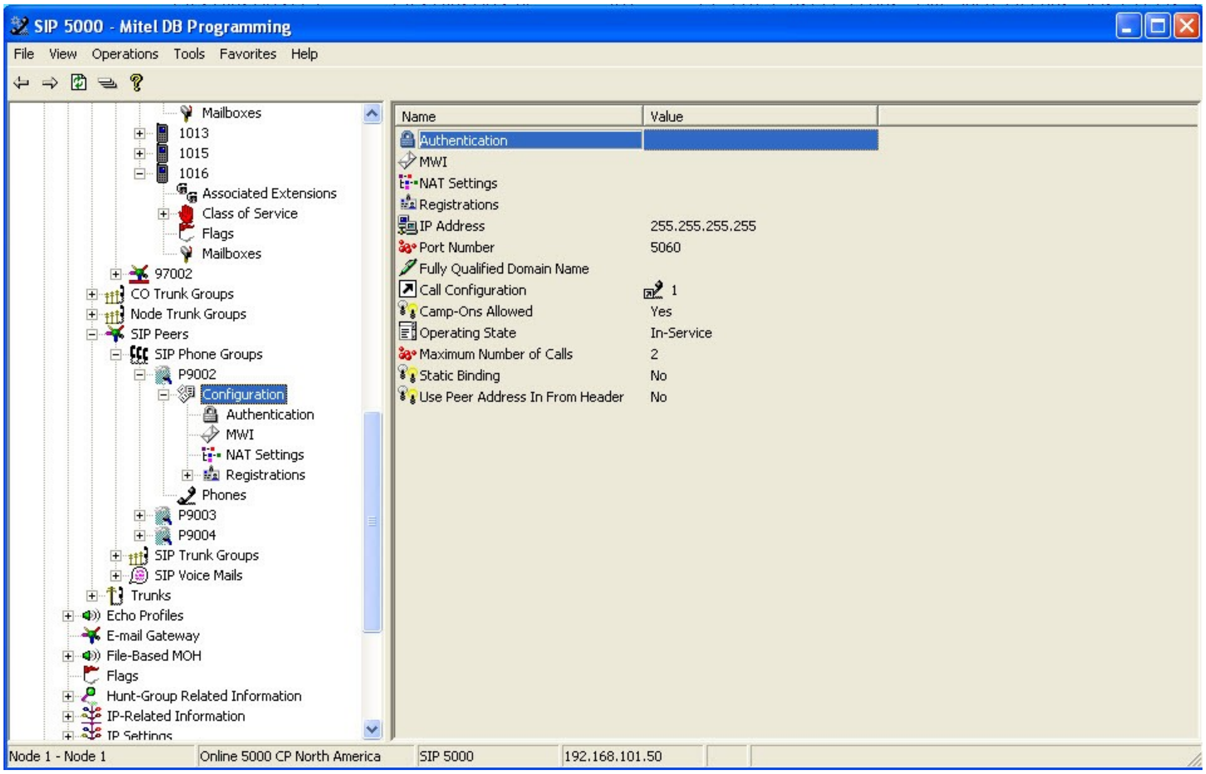

Figure 5 – SIP Phone Groups

Authentication

The Ploycom 8400 Series SIP phone group was configured to use In Bound Authentication:

• **Enable In-Bound Authentication**: If the Enable In-Bound Authentication flag is enabled

For a SIP peer, incoming calls and SIP requests from the SIP peer are authenticated by the

5000 CP.

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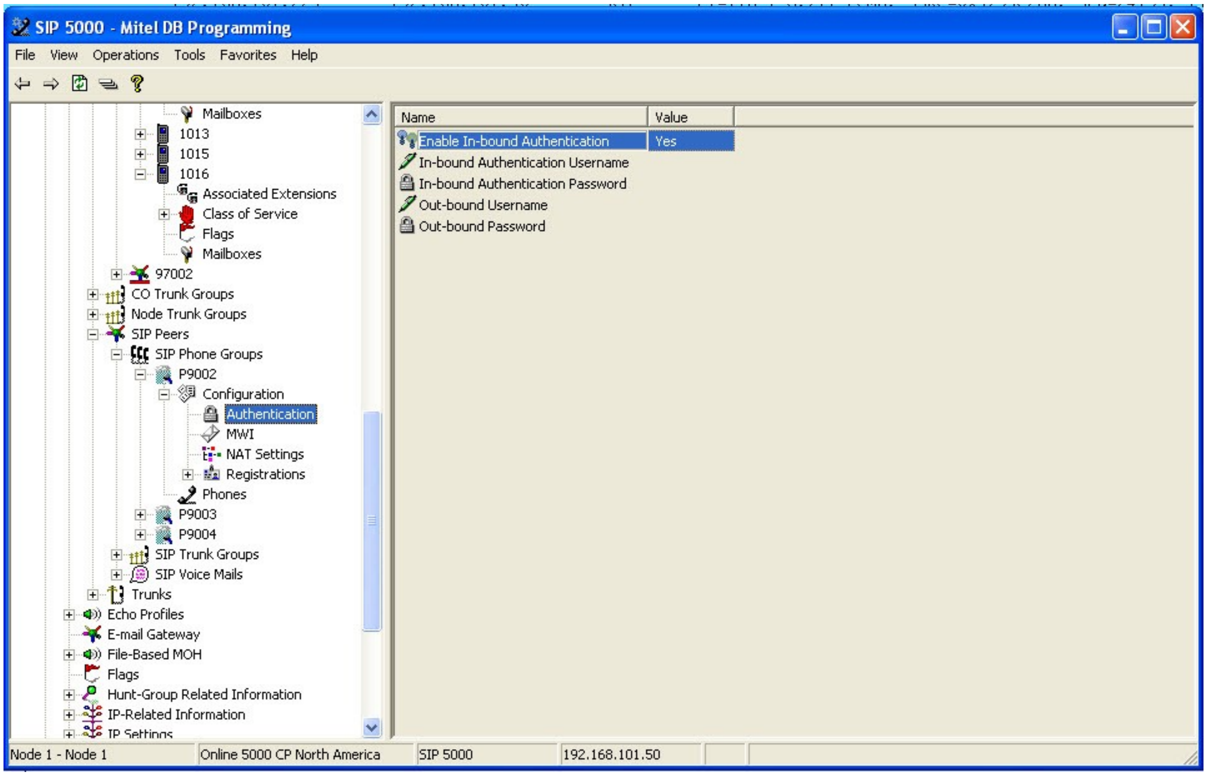

Figure 6 – SIP Phone Groups Authentication Parameters

MWI

The Message Waiting Indication (MWI) field determines whether the system accepts the MWI

From the SIP peer. Verify that the **Accept MWI** option is set to **Yes**. To have the system ignoreMWI from the SIP peer, change the setting to **No**. It is set to Yes by default.

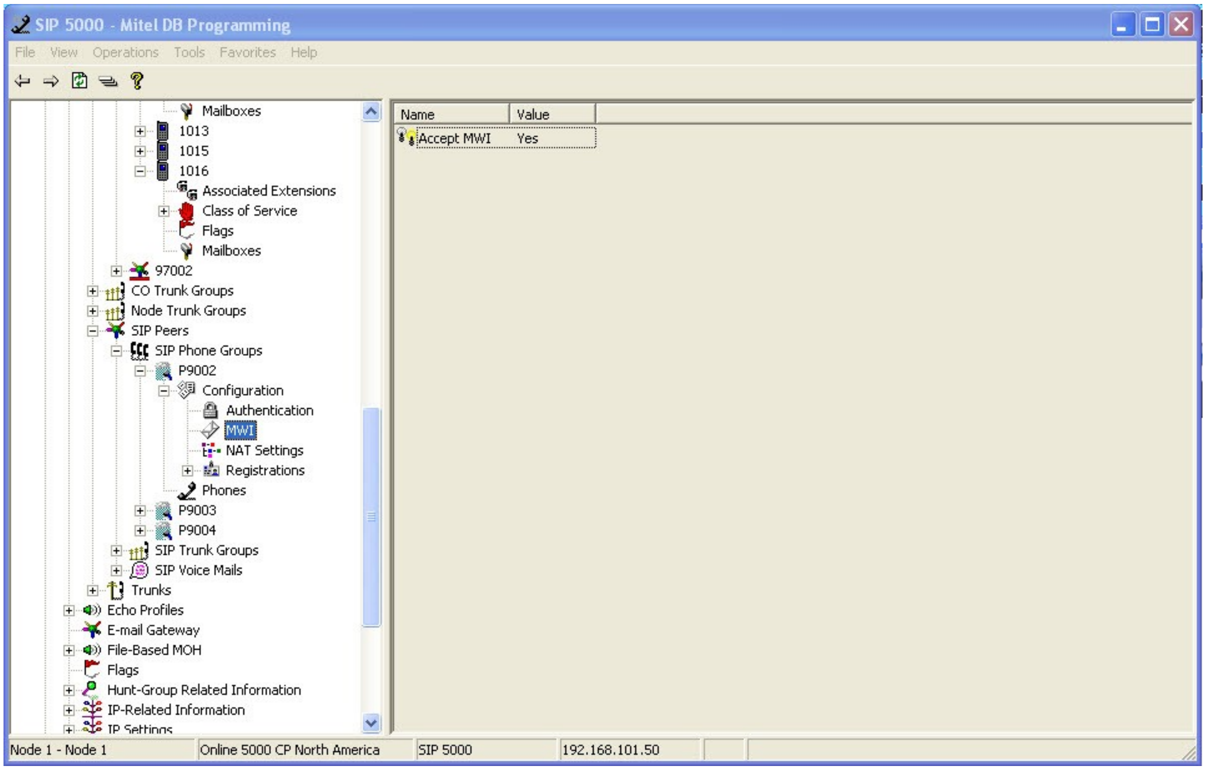

Figure 7 – MWI Configuration

For SIP Phone Groups

There are two choices:

• **Native**: Used for internal Polycom 8400 Series (those that do not pass through near-end NAT).

• **NAT**: Used for external Polycom 8400 Series (those that do pass through near-end NAT)

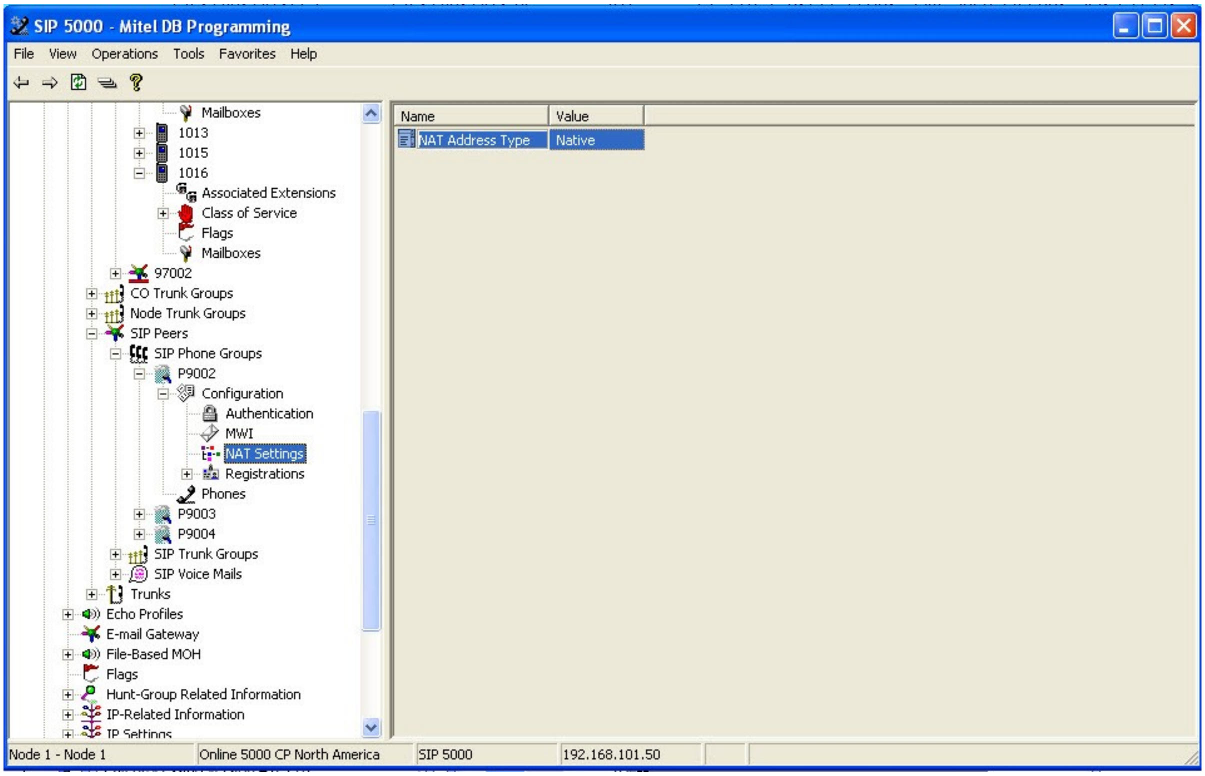

Figure 8 – NAT Settings

Registrations

You can register the Polycom 8400 Series with the 5000 CP dynamically

• For dynamic registration, the status of a Polycom 8400 Series is determined by the existence of an active registration in the system for that Polycom 8400 Series. When a Polycom 8400 Series registers with the system, its status becomes “Idle” (online) as long as there is a valid Polycom Spectralink 8400 Series (Category F Phones) license available and becomes “Offline” when the registration expires or SIP phone un-registers.

This folder allows you to configure the following settings that are required for registration per-

Polycom 8400 Series Group basis:

• **Address of Record**: Indicates the Address of Record that the SIP peer uses to register

with the 5000 CP. This field is for read-only.

• **Registration URI**: Indicates the SIP URI representing the Contact address in the SIP

REGISTER request from the SIP peer that created this dynamic binding. This field is for

read-only.

• **Registration Call ID**: Indicates the SIP Call ID of the SIP REGISTER request received

from the SIP peer that created this dynamic binding. This field is for read-only.

• **Registration Cseq Number**: Indicates the SIP Cseq number of the SIP REGISTER

request received from the SIP peer that created this dynamic binding. This field is for readonly.

• **Registration Update Time**: Indicates the timestamp when the SIP REGISTER request

was received from the SIP peer and updated the dynamic binding. This field is for readonly.

• **Registration Expire Time**: Indicates the time in seconds to expire this registration since it

was last updated. This field is for read-only.

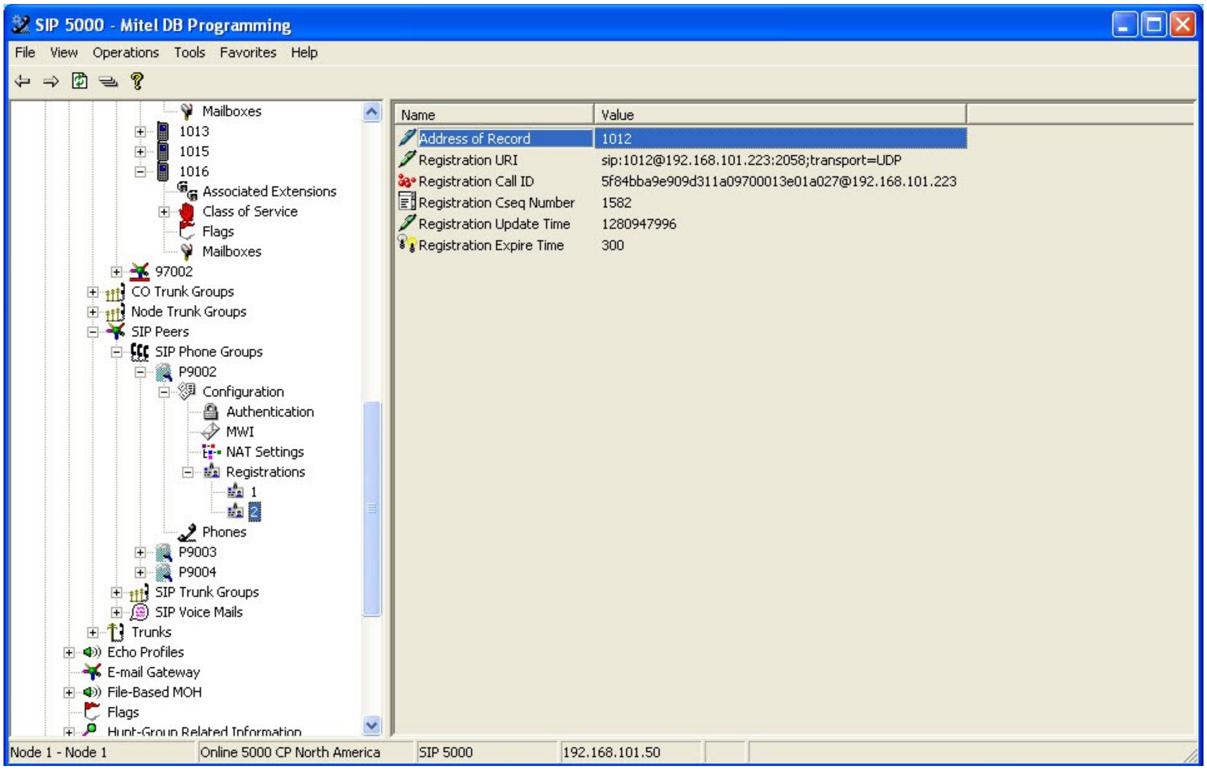

Figure 9 – Registrations

Call Configuration

Clicking **Call Configuration** takes you to the Call Configuration folder (System\IP-Related Information\Call Configurations\<*call configuration number*>). When you create a SIP peer without using a template, by default the new SIP peer is added to Call Configuration 1 <*Local*>.

The following diagram shows call configuration was used with the Polycom 8400 Series.

Please note:

- **Ensure that the DTMF Encoding is set to RFC 2833**

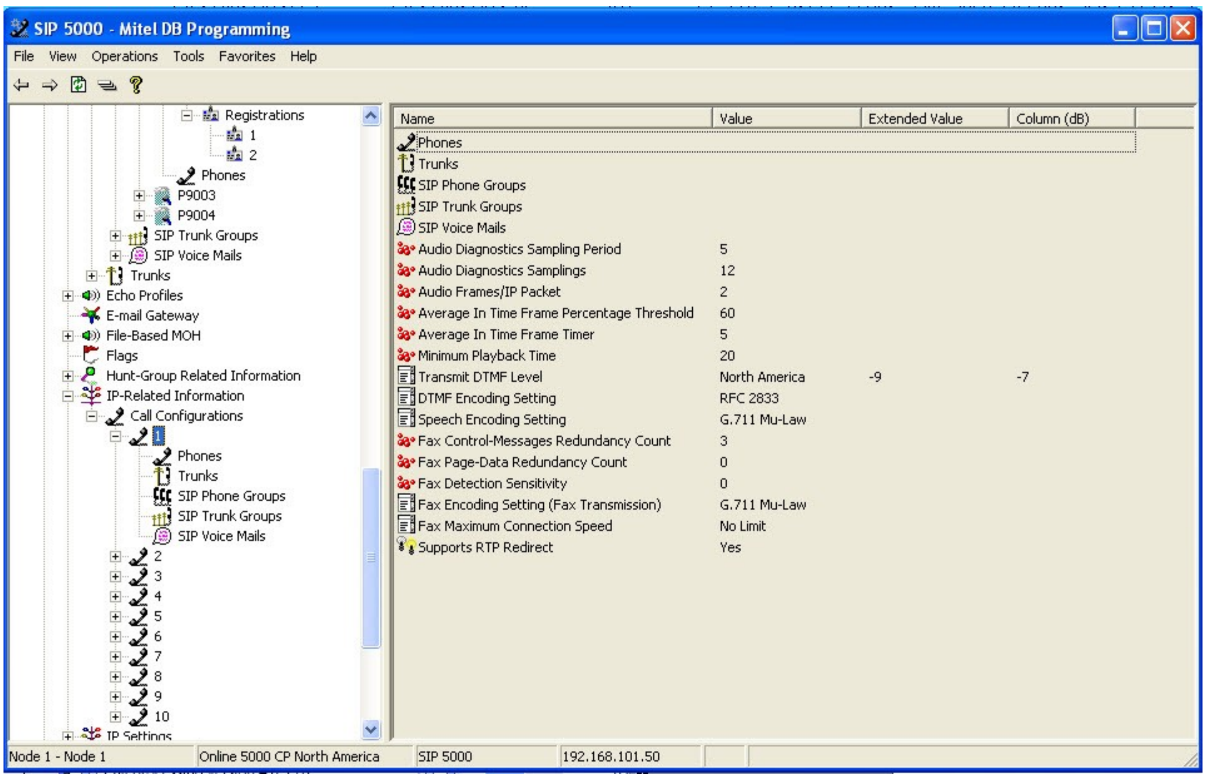

Figure 10 – Call Configuration

Polycom Spectralink 8400 Series Configuration Notes

The following are screen captures of the Spectralink 8400 Series as it was configured with the Mitel 5000
For more detailed configuration please see the Polycom documentation below:

Administration Guide
http://support.polycom.com/global/documents/support/setup_maintenance/products/voice/UC_Software_4_0_0_Administrators_Guide_eng.pdf

Deployment Guide
http://support.polycom.com/global/documents/support/setup_maintenance/products/voice/SpectraLink_8400_Deployment_Guide.pdf

You can web access the Spectralink by its IP address. The default username is Polycom and default password is 456.

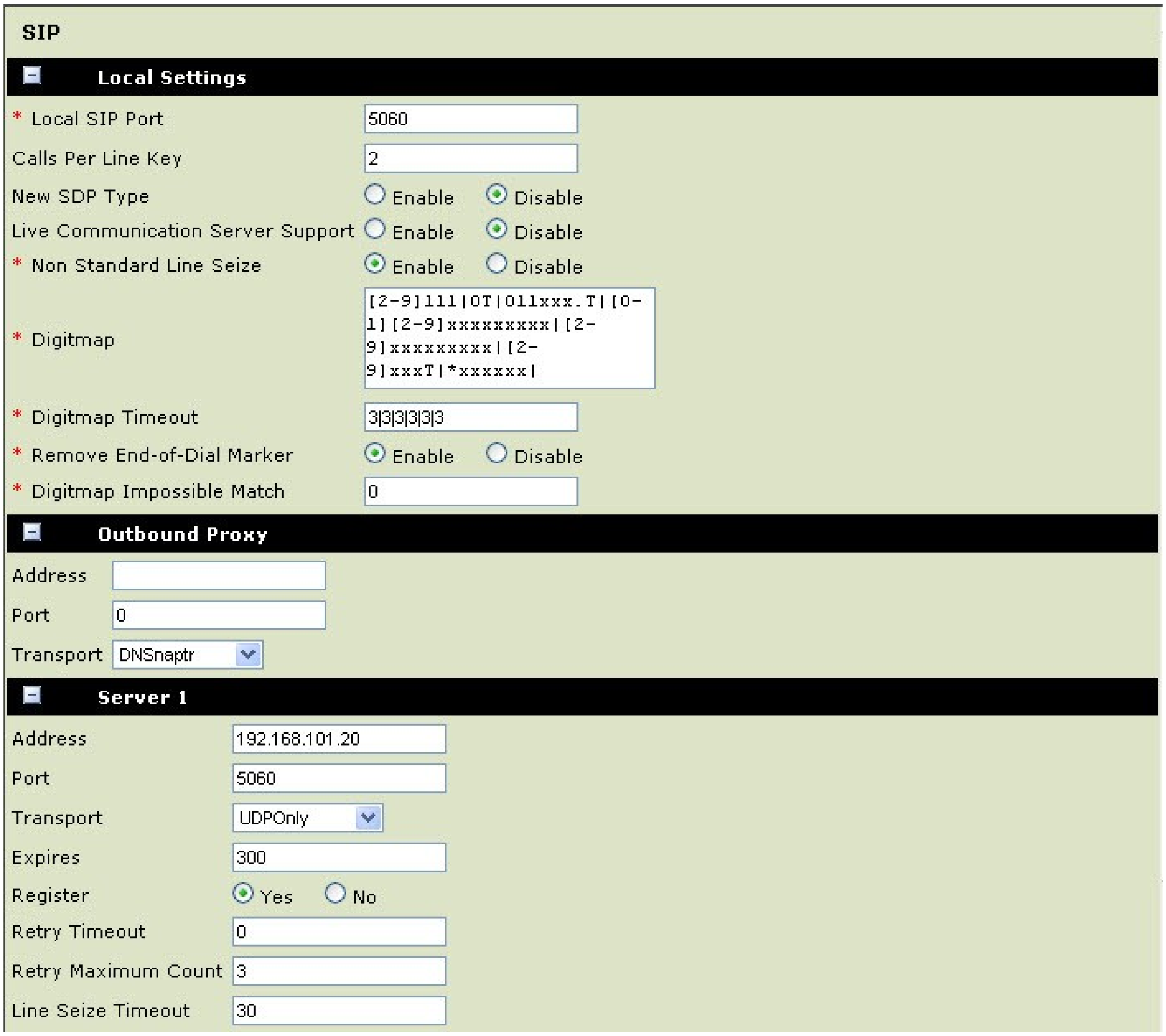

Figure 11 – SIP

Digitmap Assignment

Modifying the Digitmap for Trunk Access codes:

On the Polycom web portal, under SIP, choose local settings. In the digitmap field the default value is set to:

[2-9]11|0T|011xxx.T|[0-1][2-9]xxxxxxxxx|[2-9]xxxxxxxxx|[2-9]xxxT

This translates as follows:

[2-9]11..........................211, 311, 411... 911

0T..................................operator (dial zero after timeout)

011xxx.T.........................International Calling (dial after timeout)

[0-1][2-9]xxxxxxxxx........0(zero) + and 1 + dialing North America

[2-9]xxxxxxxxx................10 digit local calling

[2-9]xxxT........................4 digit internal dialing (after timeout)

The example below would include a Digit 9 for ARS Access:
9[2-9]11|90T|9011xxx.T|9[0-1][2-9]xxxxxxxxx|9[2-9]xxxxxxxxx|[1-8]xxx|0T
9[2-9]11..........................9211, 9311, 9411... 9911
90T.................................External operator (dial zero after
timeout)
9011xxx.T.......................International Calling (dial after timeout)
9[0-1][2-9]xxxxxxxxx........0(zero) + and 1 + dialing North America
9[2-9]xxxxxxxxx..............10 digit local calling
[1-8]xxx..........................4 digit internal dialing
0T..................................Internal Zero - Switch Board

Optionally other dialing strings can be included for Feature access codes
or Speeddials
Example below includes *8 to dial voicemail and *1xxx to dial system
speeddials that start with *1

9[2-9]11|90T|9011xxx.T|9[0-1][2-9]xxxxxxxxx|9[2-9]xxxxxxxxx|[1-8]xxx|0T|*8|*1xxx

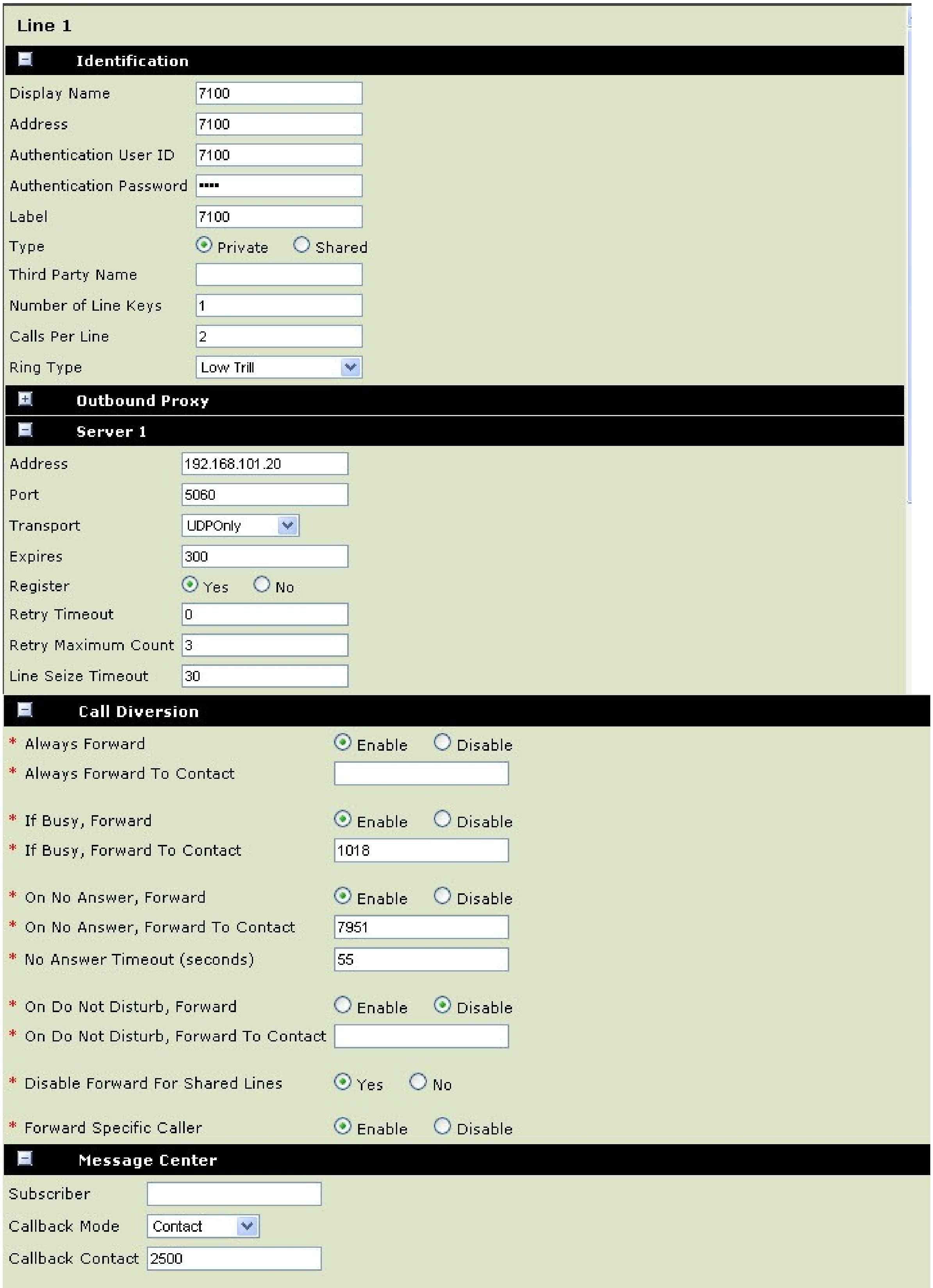

Figure 12 – Line

Appendix A

Polycom has combined their SIP stack on multiple SIP phone models and assert that these all react the same in a SIP endpoint deployed environment.

Polycom SIP Endpoints

The models that use the SIP Firmware are:

- 8440 SIP Phone
- 8400 Series SIP Phone (with Linear Image Scanner)
- 8452 SIP Phone (with Area Image Scanner)

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

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