

# MITEL – SIP CoE

# Technical

## Configuration Note



Configure Mitel MiVoice Business 7.2 SP1 PRI  
for use with IntelePeer SIP Trunking

SIP CoE 12-4940-00XXX

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Mitel Technical Configuration Notes – Configure MiVoice Business for use with IntelPeer SIP Trunking

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## Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Business to connect to IntelPeer SIP Trunking. 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	26-Aug-2015	Initial Interop with MiVoice Business Release 7.2 SP1 PR1 Software Load 13.2.1.27 and IntelPeer SIP Trunking

## Interop Status

The Interop of IntelPeer SIP Trunking has been given a Certification status. This service provider or Trunking device will be included in the SIP CoE Reference Guide. The status IntelPeer SIP Trunking achieved is:

	The most common certification which means IntelPeer SIP Trunking 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.
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## Software & Hardware Setup

This was the test setup to generate a basic SIP call between IntelPeer SIP Trunking and the MiVoice Business.

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 7.2 SP1 PR1 Active Software Load 13.2.1.27
Mitel	Minet Sets: 5320, 5360, 5312	6.03.00.12
Mitel	MiVoice Border Gateway – Teleworker	9.2.0.23
Service Provider	IntelPeer	N/A

## 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 Trunk Side Interoperability Test Plans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through IntelPeer and their PSTN gateway, call holding, call forwarding, transferring, conferencing, busy calls, DTMF RFC2833, long calls durations, variable codec, G.711 and G.729 Codec, Privacy, Loop back calling, Long Ringing	✓
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection	✓
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes as well as Embedded voicemail and DTMF detection	✓
Packetization	Forcing the MiVoice Business to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	✓
Personal Ring Groups	Receiving calls through IntelPeer and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members	✓
Teleworker	Making and receiving a call Through IntelPeer and their PSTN gateway to and from Teleworker extensions	✓
Video	Making and receiving a call through IntelPeer with video capable devices	✗
Fax	T.38 and G711Fax Calls	✓

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

## Device Limitations and Known Issues

This is a list of problems or not supported features when IntelPeer SIP Trunking is connected to the MiVoice Business.

Feature	Problem Description
Video Call	IntelPeer does not support video calls <b>Recommendation:</b> Contact IntelPeer for update on this feature

## Network Topology

This diagram shows how the testing network is configured for reference

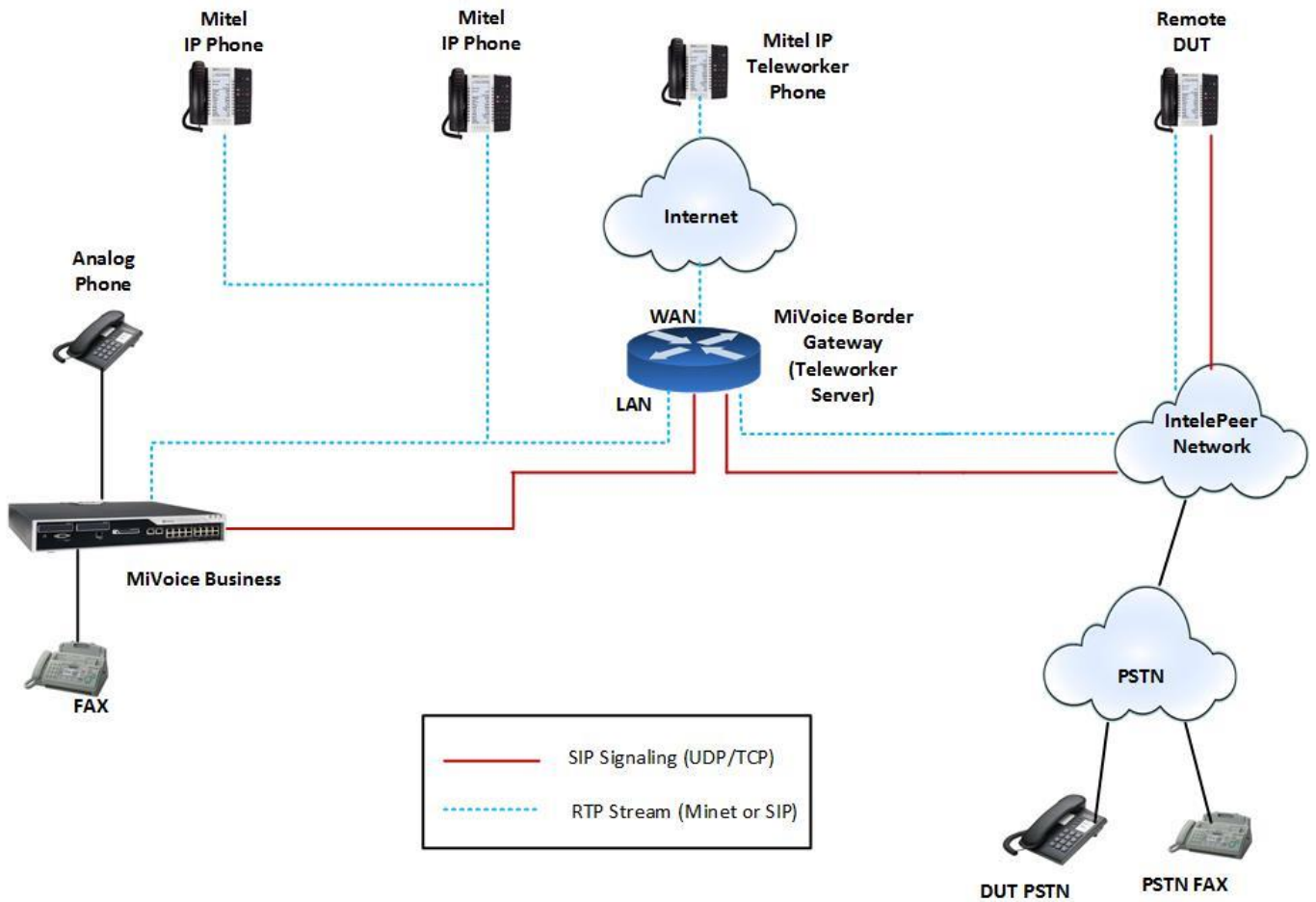


Figure 1: Network Topology

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## 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 MiVoice Business programming with IntelPeer SIP Trunking was configured in our test environment.

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**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.**

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## MiVoice Business Configuration Notes

The following steps show how to program a MiVoice Business to interconnect with IntelPeer SIP Trunking.

### Configuration Template

A configuration template can be found in the same MOL Knowledge Base article as this document. The template is a Microsoft Excel spreadsheet (.csv format) **solely** consisting of the SIP Peer profile option settings used during Interop testing. All other forms should be programmed as indicated below. Importing the template can save you considerable configuration time and reduce the likelihood of data-entry errors. Refer to the MiVoice Business documentation on how the Import functionality is used.

### Network Requirements

- There must be adequate bandwidth to support the VoIP. As a guide, the Ethernet bandwidth is approximately 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 approximately 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 MiVoice Business 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 MiVoice Business Programming

The SIP signaling connection uses UDP on Port 5060



## Licensing and Option Selection – SIP Licensing

**Navigation:** Licenses > License and Option Selection

Ensure that the MiVoice Business is equipped with enough SIP trunk licenses for the connection to IntelPeer SIP Trunking. This can be verified within the License and Option Selection form.

Enter the total number of licenses in the SIP Trunk Licences field. This is the maximum number of SIP trunk sessions that can be configured in the MiVoice Business to be used with all service providers, applications and SIP trunking devices.

**Extended Hunt Group:** Set to **YES** for NuPoint Voicemail configuration

License and Option Selection							
Online Licensing with the Application Management Center							
Application Record ID		26682859					
System Type License Sharing		Hardware Identifier					
Enterprise	No	0000003a1a4f					
<u>Local Limits</u>							
Licensed Options		Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Licenses Allowed	Can be Over Allocated
<b>Users</b>							
IP Users		9	16	0	16	Unrestricted	Yes
External Hot Desk Users		1	5	5	10	Unrestricted	Yes
ACD Active Agents		2	10	0	10	Unrestricted	No
HTML Applications		0	0	20	0	Unrestricted	Yes
Analog Lines		0	16	0	16	Unrestricted	Yes
MiVoice Business Console Active Operators		0	0	20	0	Unrestricted	No
Multi-device Users		0	5	0	5	Unrestricted	Yes
Multi-device Suites		0	0	5	5	0	No
<b>Messaging</b>							
Embedded Voice Mail		6	16	0	16	Unrestricted	Yes
Embedded Voice Mail PMS		1	Yes	0	1	Unrestricted	Yes
<b>Trunking/Networking</b>							
Digital Links		1	1	0	1	Unrestricted	Yes
Compression			8	0	8	Unrestricted	Yes
FAX Over IP (T.38)			4	0	4	Unrestricted	Yes
SIP Trunks		0	353	0	353	Unrestricted	Yes
<b>Others</b>							
IDS Connection		1	Yes	0	1	Unrestricted	Yes
MLPP		0	No	0	0	Unrestricted	No
<b>Configuration Options</b>							
Country		North America					
Extended Agent Skill Group		No					
Maximum Elements per Cluster		30					
Maximum Configurable IP Users and Devices		700					
Extended Hunt Group		Yes					
5560 IPT Device Extended Key Lines		No					

**Figure 2: License and Option Selection**

## Class of Service Assignment

**Navigation:** System Properties > System Feature Settings > Class of Service Options

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 in the Trunk Service Assignment form for SIP trunks.

The screenshot shows the Mitel system interface. The left sidebar contains a navigation menu with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties, System Settings, System Feature Settings, System Options, Shared System Options, Class of Service Options (highlighted with a red border), SIP Device Capabilities, Class of Restriction Groups, System Access Points, Feature Access Codes, and Independent Account Codes. The main area displays the 'Class of Service Options' configuration page. At the top, there is a header 'Class of Service Options on Local 2' with a 'DN to search' dropdown and buttons for 'Change', 'Copy', 'Print...', and 'Import...'. Below this is a pagination bar showing 'Page 1 of 11' and a 'Go to:' dropdown. The main content is a table with the following data:

Class Of Service Number	Comment
1	SIP trunk
2	MitelPhone
3	VM
4	
5	NuPointVM
6	ACD Annouceme
7	Ext HotDesk

Figure 3: Class of Service

## Class of Service for Trunk

### General

General	Advanced
Class Of Service Number	1
Comment	SIP trunk
<b>ACD</b>	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
ACD Make Busy on Login	No
ACD Silent Monitor Accept	No
ACD Silent Monitor Accept Monitoring Non-Prime Lines	No
ACD Silent Monitor Allowed	No
ACD Silent Monitor Notification	No
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	No
Work Timer	0
<b>Announce</b>	
Call Announce Line	No
Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack Allowed	No
<b>Busy Override</b>	
Busy Override Security	Yes
Disable Executive Busy Override Tone	No
Executive Busy Override	No
<b>Call Control Timer</b>	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10
Lockout Timer	45
<b>Call Duration</b>	
Call Duration	10
Call Duration Forced Cleardown Timer	0
Enable Call Duration Limit on External Calls	No
Enable Call Duration Limit on Internal Calls	No
<b>Call Forwarding/Rerouting</b>	
Call Forward - Delay	0
Call Forward No Answer Timer	15
Call Forward Override	No
Call Forwarding (External Destination)	Yes
Call Forwarding (Internal Destination)	Yes
Call Forwarding Accept	Yes
Call Reroute after CFFM to Busy Destination	No
Call Forwarding Reminder Ring (CFFM and CFIAH only)	No
Disable Call Reroute Chaining On Diversion	No
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No

Figure 4: Class of Service (Basic) for SIP Trunk

Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes
<b>Call Hold</b>	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	No
Call Hold Remote Retrieve	Yes
Call Hold Timer	180
Local Music On Hold source	Yes
Music on Hold on Transfer	Yes
Use Called Party Call Hold Timer	No
<b>Call Park</b>	
Call Park Timer	180
Call Park-Allowed To Park	No
<b>Call Pickup</b>	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
<b>Call Privacy</b>	
Call Privacy	No
Calling Party Name Substitution	Yes
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	Yes
<b>Call Waiting</b>	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	No
<b>Campon</b>	
Auto Campon Timer	10
Campon Recall Timer	10
<b>Direct Voice Call</b>	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
<b>Display</b>	
After Answer Display Time	
Calling Name Display - Internal - ONS	Yes
Calling Number Display - Internal - ONS	Yes
Display ANI/DNIS/ISDN Calling/Called Number	No
Display ANI/ISDN Calling Number Only	No
Display Caller ID on multicall/keylines	No
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	No
Display Dialed Digits during Outgoing Calls	No

Figure 5: Class of Service (Basic) for SIP Trunk – Cont.

Display DNIS/Called Number Before Digit Modification	No
Display Held Call ID on Transfer	No
Display Transfer Destination on Recall	No
Hot Desk External User - Display Internal Calling ID	No
Maintain Ringing Party During Recall	No
Non-Prime Public Network Identity	No
Originator's Display Update In Call Forwarding/Rerouting	No
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No
<b>Fax</b>	
Campon Tone Security	Yes
External Trunk Standard Ringback	No
Fax Capable	Yes
Return Disconnect Tone When Far End Party Clears	No
<b>HCI</b>	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
<b>Hot Desk</b>	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk External User - Allow Mid-Call Features	Yes
Hot Desk External User - Answer Confirmation	Yes
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	No
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No
<b>Miscellaneous</b>	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Force Device Busy If Any Line In Use	No
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Multi-Color LED Support - Disable	No
Phone Lock	No
Reseize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0
<b>Paging</b>	
Group Page Accept	No
Group Page Allow	No
Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes

Figure 6: Class of Service (Basic) for SIP Trunk – Cont.

Pager Access All Zones	Yes
Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
Ringing	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	180
SMDR	
SMDR External	No
SMDR Internal	No
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	No
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	No
Public Network To Public Network Connection Allowed	No
Public Trunk	No
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	No
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5

Figure 7: Class of Service (Basic) for SIP Trunk – Cont.

## Advanced

General	Advanced
<b>Account Code</b>	
Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes
<b>Attendant</b>	
Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No
<b>Conference</b>	
Conference Call	Yes
Disable Conference Join Tone	No
<b>DND</b>	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No
<b>Emergency</b>	
Emergency Call - Audio Level for Set	Ringer
Emergency Call Notification - Audio	No
Emergency Call Notification - Visual	No
<b>Group Presence</b>	
Group Presence Control	No
Group Presence Third Party Control	No
<b>Hotel</b>	
Display VIP	No
Hotel Room Monitor Setup Allowed	No
Hotel Room Monitoring Allowed	No
Hotel/Motel Room Personal Wakeup Call Allowed	No
Hotel/Motel Room Remote Wakeup Call Allowed	No
<b>Message Waiting</b>	
Message Waiting	Yes
Message Waiting - Disable Ringing Lamp Notification	No
Message Waiting Audible Tone Notification	No
Message Waiting Deactivate On Off-Hook	Yes
Message Waiting Inquire	Yes
Message Waiting Ringing Start Time Hour	
Message Waiting Ringing Start Time Minute	
Message Waiting Ringing Stop Time Hour	
Message Waiting Ringing Stop Time Minute	
Multiline Set Voice Mail Callback Message Erasure Allowed	No
ONS CLASS/CLIP: Message Waiting Activate/Deactivate	Yes

Figure 8: Class of Service (Advanced) for SIP Trunk

<b>Miscellaneous</b>	
Auto Answer Allowed	Yes
Auto Release on Key Select	No
Brokers Call	No
Called Party Features Override	No
Check COR after PSTN Dial Tone	No
Dialled Night Service	Yes
Disable Send Message	No
Flexible Answer Point	No
Individual Trunk Access	Yes
Key A	
Key B	
Key C	
Key D	
Multiline Set Loop Test	No
Multiline Set Message Center Remote Read Allowed	No
Multiline Set Music	No
Multiline Set On-hook Dialing	Yes
Multiline Set Phonebook Allowed	Yes
Non DID Extension	No
ONS CLASS/CLIP: Set	No
ONS/OPS Internal Ring Cadence for External Callers	No
Override Interconnect Restriction on Transfer	No
Recall If Transferred to Original Call Destination	No
Redial Facilities	Yes
Use Default Billable Number For Trunk Calls	No
Voice Dial Preferred	No
Voice Mail Softkey	No
<b>Phonebook</b>	
Phonebook Lookup - Default to User Location	No
Phonebook Lookup - Display User Location	No
<b>Record A Call</b>	
Record-A-Call - Save Recording on Hang-up	No
Record-A-Call - Start Automatic Incoming Call Recording	No
Record-A-Call - Start Automatic Outgoing External Call Recording	No
Record-A-Call Active	No

Figure 9: Class of Service (Advanced) for SIP Trunk – Cont.



## Class of Service for Phone

### General

General		Advanced
Class Of Service Number	2	
Comment	MitelPhone	
<b>ACD</b>		
ACD Agent Behavior on No Answer	Logout	
ACD Agent No Answer Timer	10	
ACD Make Busy on Login	Yes	
ACD Silent Monitor Accept	No	
ACD Silent Monitor Accept Monitoring Non-Prime Lines	No	
ACD Silent Monitor Allowed	No	
ACD Silent Monitor Notification	No	
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	No	
Work Timer	0	
<b>Announce</b>		
Call Announce Line	No	
Off-Hook Voice Announce Allowed	No	
Handsfree AnswerBack Allowed	No	
<b>Busy Override</b>		
Busy Override Security	No	
Disable Executive Busy Override Tone	No	
Executive Busy Override	No	
<b>Call Control Timer</b>		
Busy Tone Timer	30	
Dialing Conflict Timer	3	
First Digit Timer	15	
Inter Digit Timer	10	
Lockout Timer	45	
<b>Call Duration</b>		
Call Duration	10	
Call Duration Forced Cleardown Timer	0	
Enable Call Duration Limit on External Calls	No	
Enable Call Duration Limit on Internal Calls	No	
<b>Call Forwarding/Rerouting</b>		
Call Forward - Delay	0	
Call Forward No Answer Timer	15	
Call Forward Override	Yes	
Call Forwarding (External Destination)	Yes	
Call Forwarding (Internal Destination)	Yes	
Call Forwarding Accept	Yes	
Call Reroute after CFFM to Busy Destination	No	
Call Forwarding Reminder Ring (CFFM and CFIAH only)	No	
Disable Call Reroute Chaining On Diversion	No	
Group Call Forward Follow Me Accept	No	
Group Call Forward Follow Me Allow	No	
Third Party Call Forward Follow Me Accept	No	

Figure 10: Class of Service (Basic) for Phone

Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes
<b>Call Hold</b>	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	No
Call Hold Remote Retrieve	Yes
Call Hold Timer	30
Local Music On Hold source	Yes
Music on Hold on Transfer	Yes
Use Called Party Call Hold Timer	No
<b>Call Park</b>	
Call Park Timer	180
Call Park-Allowed To Park	No
<b>Call Pickup</b>	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
<b>Call Privacy</b>	
Call Privacy	No
Calling Party Name Substitution	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	Yes
<b>Call Waiting</b>	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes
<b>Campon</b>	
Auto Campon Timer	0
Campon Recall Timer	10
<b>Direct Voice Call</b>	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
<b>Display</b>	
After Answer Display Time	
Calling Name Display - Internal - ONS	Yes
Calling Number Display - Internal - ONS	Yes
Display ANI/DNIS/ISDN Calling/Called Number	Yes
Display ANI/ISDN Calling Number Only	Yes
Display Caller ID on multicall/keylines	Yes
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	No
Display Dialed Digits during Outgoing Calls	Yes

Figure 11: Class of Service (Basic) for Phone – Cont.

Display DNIS/Called Number Before Digit Modification	Yes
Display Held Call ID on Transfer	No
Display Transfer Destination on Recall	No
Hot Desk External User - Display Internal Calling ID	Yes
Maintain Ringing Party During Recall	No
Non-Prime Public Network Identity	No
Originator's Display Update In Call Forwarding/Rerouting	Yes
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No
<b>Fax</b>	
Campan Tone Security	No
External Trunk Standard Ringback	No
Fax Capable	No
Return Disconnect Tone When Far End Party Clears	No
<b>HCI</b>	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
<b>Hot Desk</b>	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk External User - Allow Mid-Call Features	Yes
Hot Desk External User - Answer Confirmation	No
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	Yes
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No
<b>Miscellaneous</b>	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Force Device Busy If Any Line In Use	No
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Multi-Color LED Support - Disable	No
Phone Lock	No
Resize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0
<b>Paging</b>	
Group Page Accept	No
Group Page Allow	No
Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes

Figure 12: Class of Service (Basic) for Phone – Cont.

Pager Access All Zones	Yes
Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
Ringing	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	180
SMDR	
SMDR External	Yes
SMDR Internal	No
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	Yes
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	Yes
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	Yes
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5

Figure 13: Class of Service (Basic) for Phone – Cont.

## Advanced

General		Advanced
<b>Account Code</b>		
Account Code Length		12
Account Code Verified		No
Forced Non-Verified Account Code		No
Forced Verified Account Code		No
Non Verified Account Code		Yes
<b>Attendant</b>		
Attendant Busy Out Timer		10
SC1000 Attendant Basic Function Key		No
<b>Conference</b>		
Conference Call		Yes
Disable Conference Join Tone		No
<b>DND</b>		
Do Not Disturb		Yes
Do Not Disturb - Access to Remote Phones		Yes
Do Not Disturb Permanent		No
<b>Emergency</b>		
Emergency Call - Audio Level for Set		Ringer
Emergency Call Notification - Audio		No
Emergency Call Notification - Visual		No
<b>Group Presence</b>		
Group Presence Control		No
Group Presence Third Party Control		No
<b>Hotel</b>		
Display VIP		No
Hotel Room Monitor Setup Allowed		No
Hotel Room Monitoring Allowed		No
Hotel/Motel Room Personal Wakeup Call Allowed		No
Hotel/Motel Room Remote Wakeup Call Allowed		No
<b>Message Waiting</b>		
Message Waiting		Yes
Message Waiting - Disable Ringing Lamp Notification		No
Message Waiting Audible Tone Notification		No
Message Waiting Deactivate On Off-Hook		Yes
Message Waiting Inquire		Yes
Message Waiting Ringing Start Time Hour		
Message Waiting Ringing Start Time Minute		
Message Waiting Ringing Stop Time Hour		
Message Waiting Ringing Stop Time Minute		
Multiline Set Voice Mail Callback Message Erasure Allowed		No
ONS CLASS/CLIP: Message Waiting Activate/Deactivate		No

Figure 14: Class of Service (Advanced) for Phone

<b>Miscellaneous</b>	
Auto Answer Allowed	Yes
Auto Release on Key Select	No
Brokers Call	No
Called Party Features Override	No
Check COR after PSTN Dial Tone	No
Dialled Night Service	Yes
Disable Send Message	No
Flexible Answer Point	No
Individual Trunk Access	Yes
Key A	
Key B	
Key C	
Key D	
Multiline Set Loop Test	No
Multiline Set Message Center Remote Read Allowed	No
Multiline Set Music	No
Multiline Set On-hook Dialing	Yes
Multiline Set Phonebook Allowed	Yes
Non DID Extension	No
ONS CLASS/CLIP: Set	No
ONS/OPS Internal Ring Cadence for External Callers	No
Override Interconnect Restriction on Transfer	No
Recall If Transferred to Original Call Destination	No
Redial Facilities	Yes
Use Default Billable Number For Trunk Calls	No
Voice Dial Preferred	No
Voice Mail Softkey	No
<b>Phonebook</b>	
Phonebook Lookup - Default to User Location	No
Phonebook Lookup - Display User Location	No
<b>Record A Call</b>	
Record-A-Call - Save Recording on Hang-up	No
Record-A-Call - Start Automatic Incoming Call Recording	No
Record-A-Call - Start Automatic Outgoing External Call Recording	No
Record-A-Call Active	No

Figure 15: Class of Service (Advanced) for Phone – Cont.

## Network Element Assignment

**Navigation:** Voice Network > Network Elements

Create a network element for IntelPeer SIP Trunking. In this example, the soft switch is reachable by an IP Address and is defined as “IntelPee” in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy and Registrar are provided by your service provider.**

If your service provider trusts your network connection by asking for your gateway external IP address, then programming the IP address for the SIP Peer, Outbound Proxy and Registrar is not required for SIP trunk integration. This will need to be verified with your service provider. Set the **SIP Peer Transport** to UDP and **Port** to 5060.

The screenshot shows a web form titled "Change Network Elements". The form contains the following fields and values:


Change Network Elements	
Name	Intelpeep x
Type	Other v
FQDN or IP Address	68.68.123.174
Local Version	False
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	UDP v
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	default v
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default v
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal v
<div>Save Cancel</div>	

Figure 16: Network Element Assignment

## Network Element Assignment (Proxy)

In addition, depending in your configuration, a Proxy may need to be configured to route SIP data to the service provider. If you have a Proxy server installed in your network, MiVoice Business will require knowledge of this by programming the Proxy as a network element then referencing this proxy in the SIP Peer profile assignment (later in this document).

**Change**

 **Network Elements**

<b>Name</b>	InteleGW x
<b>Type</b>	Outbound Proxy v
<b>FQDN or IP Address</b>	10.64.3.2
<b>Local Version</b>	False
<b>Zone</b>	1
<b>ARID</b>	
<b>Outbound Proxy Specific</b>	
<b>Outbound Proxy Transport Type</b>	UDP v
<b>Outbound Proxy Port</b>	5060

**Save** **Cancel**

Figure 17: Network Element Assignment (Proxy)



## Trunk Attributes

This is configured in the Trunk Attributes form. In this example, the Trunk Attributes is defined for Trunk Service Number 1 which will be used to direct incoming calls to an answer point in the MiVoice Business.

1. Program the Non-dial In or Dial In Trunks (DID) according to the site requirements and what type of service was ordered from your service provider
2. **Dial in Trunks Incoming Digit Modification- Absorb** is set to 0





Change	
 <b>Trunk Attributes</b>	
Trunk Service Number	1
Release Link Trunk	No 
Call Recognition Service	Off 
Direct Inward Dialing Service	<input type="radio"/> Off <input checked="" type="radio"/> On
Class of Service	1
Class of Restriction	1
Baud Rate	300 
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Label	Intelepeer
<div>Save Cancel</div>	

Figure 18: Trunk Attributes

---

## SIP Peer Profile

**Navigation: Trunks > SIP > SIP Peer Profile**

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base MiVoice Business Platform. The SIP Peer Profile should be configured with the following options:

### Basic

1. **Network Element:** Select the SIP Peer Profile that needs to be associated with previously created "IntelePee" Network Element
2. **Address Type:** Select IP address of your Mitel 3300ICP
3. **Maximum Simultaneous Calls:** This entry should be configured to maximum number of SIP trunks provided by IntelePeer
4. **Outbound Proxy Server:** Select the Network Element previously configured for the Outbound Proxy Server
5. **SMDR:** If Call Detail Records are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank)
6. **Trunk Service:** Enter the trunk service number that was previously configured. **1** is used in this configuration.
7. **Subscription User Name/Password:** Enter user name and password which will be matched in later MBG configuration for KPML credentials under Configuration > Settings > Service Parameter. This is part of configuration for Mid Call features to function with KPML such as pressing 5 to handoff from the EHDU in the PRG (Personal Ring Groups).

**NOTE: Ensure the remaining SIP Peer profile policy options are similar to the figure below**

SIP Peer Profile				
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
SIP Peer Profile Label	Inteleper			
Network Element	Intelepee ▼			
<b>Local Account Information</b>				
Registration User Name				
Address Type	<input type="radio"/> FQDN <input checked="" type="radio"/> IP Address: 10.35.32.2			
<b>Administration Options</b>				
Interconnect Restriction	1			
Maximum Simultaneous Calls	4			
Minimum Reserved Call Licenses	0			
<b>Administration Options</b>				
Outbound Proxy Server	InteleGW ▼			
SMDR Tag	0			
Trunk Service	1			
Zone	1			
User Name				
Password				
Confirm Password				
Authentication Option for Incoming Calls	No Authentication ▼			
Subscription User Name	administrator			
Subscription Password	••••••••			
Subscription Confirm Password	••••••••			

Figure 19: SIP Peer Profile – Basic

## Call Routing

All the parameters are configured as shown

SIP Peer Profile				
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
Alternate Destination Domain Enabled	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Alternate Destination Domain FQDN or IP Address				
Enable Special Re-invite Collision Handling	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Only Allow Outgoing Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Private SIP Trunk	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Reject Incoming Anonymous Calls	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Route Call Using P-Called-Party-ID (if present)	<input checked="" type="radio"/> No <input type="radio"/> Yes			
Route Call Using To Header	<input checked="" type="radio"/> No <input type="radio"/> Yes			

Figure 20: SIP Peer Profile Assignment – Call Routing

### Calling Line ID

All the parameters are configured as shown

SIP Peer Profile	
Basic	Call Routing
Default CPN	<input type="text"/>
Default CPN Name	<input type="text"/>
CPN Restriction	<input checked="" type="radio"/> No <input type="radio"/> Yes
Public Calling Party Number Passthrough	<input type="radio"/> No <input checked="" type="radio"/> Yes
Strip PNI	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Diverting Party Number as Calling Party Number	<input checked="" type="radio"/> No <input type="radio"/> Yes
Use Original Calling Party Number If Available	<input checked="" type="radio"/> No <input type="radio"/> Yes

Figure 21: SIP Peer Profile Assignment – Calling Line ID

### SDP Options

All the parameters are configured as shown

SIP Peer Profile	
Basic	Call Routing
Allow Peer To Use Multiple Active M-Lines	<input checked="" type="radio"/> No <input type="radio"/> Yes
Allow Using UPDATE For Early Media Renegotiation	<input checked="" type="radio"/> No <input type="radio"/> Yes
Avoid Signaling Hold to the Peer	<input type="radio"/> No <input checked="" type="radio"/> Yes
AVP Only Peer	<input type="radio"/> No <input checked="" type="radio"/> Yes
Enable Mitel Proprietary SDP	<input checked="" type="radio"/> No <input type="radio"/> Yes
Force sending SDP in initial Invite message	<input type="radio"/> No <input checked="" type="radio"/> Yes
Force sending SDP in initial Invite - Early Answer	<input checked="" type="radio"/> No <input type="radio"/> Yes
Ignore SDP Answers in Provisional Responses	<input checked="" type="radio"/> No <input type="radio"/> Yes
Limit to one Offer/Answer per INVITE	<input checked="" type="radio"/> No <input type="radio"/> Yes
NAT Keepalive	<input checked="" type="radio"/> No <input type="radio"/> Yes
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	<input type="radio"/> No <input checked="" type="radio"/> Yes
Renegotiate SDP To Enforce Symmetric Codec	<input checked="" type="radio"/> No <input type="radio"/> Yes
Repeat SDP Answer If Duplicate Offer Is Received	<input checked="" type="radio"/> No <input type="radio"/> Yes
Restrict Audio Codec	No Restriction <input type="button" value="v"/>
RTP Packetization Rate Override	<input checked="" type="radio"/> No <input type="radio"/> Yes
RTP Packetization Rate	20ms <input type="button" value="v"/>
Special handling of Offers in 2XX responses (INVITE)	<input checked="" type="radio"/> No <input type="radio"/> Yes
Suppress Use of SDP Inactive Media Streams	<input checked="" type="radio"/> No <input type="radio"/> Yes

Figure 22: SIP Peer Profile Assignment – SDP Options

## Signaling and Header Manipulation

All the parameters are configured as shown

SIP Peer Profile				
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
Trunk Group Label		<input type="text"/>		
Allow Display Update		<input type="radio"/> No <input checked="" type="radio"/> Yes		
Build Contact Using Request URI Address		<input checked="" type="radio"/> No <input type="radio"/> Yes		
De-register Using Contact Address not *		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Disable Reliable Provisional Responses		<input type="radio"/> No <input checked="" type="radio"/> Yes		
Disable Use of User-Agent and Server Headers		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Domain for Trunk Context		<input type="text"/>		
E.164: Enable sending '+'		<input checked="" type="radio"/> No <input type="radio"/> Yes		
E.164: Add '+' if digit length > N digits		<input type="text" value="0"/>		
E.164: Do not add '+' to Emergency Called Party		<input checked="" type="radio"/> No <input type="radio"/> Yes		
E.164: Do not add '+' to Called Party		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Force Max-Forward: 70 on Outgoing Calls		<input checked="" type="radio"/> No <input type="radio"/> Yes		
If TLS use 'sips:' Scheme		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Ignore Incoming Loose Routing Indication		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Include Diversion Header for EHDU		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Multilingual Name Display		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Only use SDP to decide 180 or 183		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Prefer From Header for Caller ID		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Require Reliable Provisional Responses on Outgoing Calls		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Signal Privacy (if enabled) on Emergency Calls		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Suppress Redirection Headers		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use Fixed Retry Time for 491		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use Privacy: none		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use P-Asserted Identity Header		<input type="radio"/> No <input checked="" type="radio"/> Yes		
Use P-Asserted Identity for Billing		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use P-Call-Leg-ID Header		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use P-Preferred Identity Header		<input type="text" value="No"/>		
Use Restricted Character Set For Authentication		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use To Address in From Header on Outgoing Calls		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use user=phone		<input checked="" type="radio"/> No <input type="radio"/> Yes		
Use user=phone for Diversion Header		<input checked="" type="radio"/> No <input type="radio"/> Yes		

Figure 23: SIP Peer Profile Assignment – Signaling and Header Manipulation

## Timers

All the parameters are configured as shown

SIP Peer Profile	
Basic	Call Routing
Calling Line ID	SDP Options
Signaling and Header Manipulation	Timers
Key Press Event	
Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	1800
Session Timer: Local as Refresher	<input checked="" type="radio"/> No <input type="radio"/> Yes
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

Figure 24: SIP Peer Profile Assignment - Timers

## Key Press Event

1. Set Yes for:
  - a. Allow Inc Subscriptions for Local Digit Monitoring
  - b. Allow Out Subscriptions for Remote Digit Monitoring
  - c. Request Outbound Proxy to Handle Out Subscriptions
2. Set **KPML Transport** to *UDP*
3. Set **KPML Port** to *5060*

SIP Peer Profile	
Basic	Call Routing
Calling Line ID	SDP Options
Signaling and Header Manipulation	Timers
Key Press Event	
Profile Info	
Allow Inc Subscriptions for Local Digit Monitoring	<input type="radio"/> No <input checked="" type="radio"/> Yes
Allow Out Subscriptions for Remote Digit Monitoring	<input type="radio"/> No <input checked="" type="radio"/> Yes
Force Out Subscriptions for Remote Digit Monitoring	<input checked="" type="radio"/> No <input type="radio"/> Yes
Request Outbound Proxy to Handle Out Subscriptions	<input type="radio"/> No <input checked="" type="radio"/> Yes
KPML Transport	UDP
KPML Port	5060

Figure 25: SIP Peer Profile Assignment – Outgoing DID Ranges

SIP Peer Profile	
Basic	Call Routing
Calling Line ID	SDP Options
Key Press Event	Profile Information
Creator	
Date Created	
Created with Version	
Service Provider	
Vendor Notes	

Figure 26: SIP Peer Profile Assignment – Profile Information

## ARS Digit Modification Plans

**Navigation:** Call Routing > Automatic Route Selection (ARS) > ARS Digit Modification Plans

1. Ensure that Digit Modification for outgoing calls on the SIP trunk to IntelPeer absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 1 digit (in this case will be 9 to dial out).
2. Click **Save**

Change	
ARS Digit Modification Plans	
Digit Modification Number	1
Number of Digits to Absorb	1
Digits to be Inserted	
Final Tone Plan/Information Marker	
<div> <div>Save</div> <div>Cancel</div> </div>	

Figure 27: Digit Modification Assignment

## ARS Routes

**Navigation:** Call Routing > Automatic Route Selection > ARS Routes

1. Create a route for SIP Trunks connecting a trunk to IntelPeer. In this example, the SIP trunk is assigned to Route Number 1.
2. Choose **SIP Trunk** as a **Routing Medium**
3. Choose the **SIP Peer Profile** and **Digit Modification** entry created earlier
4. Click **Save**

Change	
ARS Routes	
Route Number	1
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	Inteleper
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	1
Digits Before Outpulsing	
Route Type	
Compression	Off
<div>Save Cancel</div>	

Figure 28: SIP Trunk Route Assignment



## ARS Digits Dialed

**Navigation:** Call Routing > Automatic Route Selection > ARS Digits Dialed

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this example, when a user dials 9, the call will be routed to IntelPeer via route 1 configured in the previous step.

1. Click **Save**

**Change**

**Change Range Programming - ARS Digits Dialed** [Help](#)

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

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
9	10	Route	1

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed	Change to <input type="button" value="v"/>	<input type="text" value="9"/>	<input type="text"/>
Number of Digits to Follow	Change to <input type="button" value="v"/>	<input type="text" value="10"/> <input type="button" value="v"/>	-
Termination Type	Change to <input type="button" value="v"/>	<input type="text" value="Route"/> <input type="button" value="v"/>	-
Termination Number	Change to <input type="button" value="v"/>	<input type="text" value="1"/>	<input type="text"/>

Figure 29: ARS Digit Dialed Assignment

## Personal Ring Groups Configuration

**Navigation:** Users and Devices > Group Programming > Personal Ring Groups

Mitel phone extension 1000 and an EHDU (External Hot Desk User) 1111 are added as members of Personal Ring Group. EHDU 1111 targets an external PSTN number

Personal Ring Groups on **Local\_2**

< **Page 1 of 1** > **Go to:**   **value:**

Personal Ring Groups

Personal Ring Group	One Busy All Busy	Prime Member Name	Home Element	Secondary Element
1111	No	Twin,ext	Local_2	Not Assigned

Personal Ring Group 1111  
Local-only DN False  
One Busy All Busy No  
Prime Member Name Twin,ext  
Home Element Local\_2  
Secondary Element Not Assigned

Personal Ring Group Members

Member Index	Number	Presence	Name	Home Element	Secondary Element
1	1111	Present	Twin,ext	Local_2	Not Assigned
2	1000	Present	user1	Local_2	Not Assigned

Figure 30: Personal Ring Groups

Multiline IP sets 1111 and 1000 are configured as follows:

**Navigation:** Users and Devices > Advanced Configuration > IP Telephones > Multiline IP Sets

Multiline IP Sets										
Device Id	Hot Desk User	Device Type	Auxiliary Module	Number	ACD Enabled	Line Type	Interconnect Number	Hot Desk User External Dialing Prefix	Hot Desk User External Number	Max Call History Record
1	No	5340e IP	None	1001	Yes	Single Line	1			0
2	No	5312 IP	None	1003	No	Single Line	1			0
3	No	5330e IP	None	0415	Yes	Single Line	1			0
4	No	5360 IP	None	1004	Yes	Single Line	1			0
5	No	5360 IP	None	1000	No	Single Line	1			0
6	No	5020 IP	None	2910	No	Single Line	1			0
7	No	5020 IP	None	2911	No	Single Line	1			0
8	No	5224 dual mode	None	1005	Yes	Single Line	1			0
10	Yes		None	1111	No	Single Line	1	91	2142425916	0

Figure 31: Multiline IP Sets

**Change**

**Change Range Programming - Multiline IP Sets**

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	SIP Password	ACD Enabled	Line Type	Interconnect Number
10	Yes		None	1111	False	*****	*****	No	Single Line	1

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Device Id	-	10	-
Hot Desk User	Change to <input type="button" value="v"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes	-
Device Type	Change to <input type="button" value="v"/>	5005 IP <input type="button" value="v"/>	-
Auxiliary Module	Change to <input type="button" value="v"/>	None <input type="button" value="v"/>	-
Number	Change to <input type="button" value="v"/>	1111 <input type="button" value="v"/>	<input style="width: 50px;" type="text"/>
Local-only DN	Change to <input type="button" value="v"/>	<input type="checkbox"/>	-
User PIN	Change to <input type="button" value="v"/>	<input style="width: 100px;" type="text"/>	-
Confirm User PIN	Change to <input type="button" value="v"/>	<input style="width: 100px;" type="text"/>	-
SIP Password	Change to <input type="button" value="v"/>	<input style="width: 100px;" type="text"/>	-
Confirm SIP Password	Change to <input type="button" value="v"/>	<input style="width: 100px;" type="text"/>	-
ACD Enabled	Change to <input type="button" value="v"/>	<input checked="" type="radio"/> No <input type="radio"/> Yes	-

**Figure 32: Programming Multiline IP Sets**

Line Type	-	Single Line	-
Interconnect Number	Change to <input type="button" value="v"/>	<input type="text" value="1"/>	<input type="button" value="v"/>
External Hot Desk User License	Change to <input type="button" value="v"/>	<input type="radio"/> No <input checked="" type="radio"/> Yes	-
Hot Desk User External Dialing Prefix	Change to <input type="button" value="v"/>	<input type="text" value="91"/>	-
Hot Desk User External Number	Change to <input type="button" value="v"/>	<input type="text" value="2142425916"/>	-
Language	-	English	-
Max Call History Records	Change to <input type="button" value="v"/>	<input type="text" value="0"/>	<input type="button" value="v"/>
MAC Address	Change to <input type="button" value="v"/>	<input type="text"/>	-
Tenant Number	Change to <input type="button" value="v"/>	<input type="text" value="1"/>	<input type="button" value="v"/>
Lock Default Configuration	Change to <input type="button" value="v"/>	<input checked="" type="radio"/> No <input type="radio"/> Yes	-
HTML Infrastructure License	Change to <input type="button" value="v"/>	<input checked="" type="radio"/> No <input type="radio"/> Yes	-
HTML GUI Application	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
New Page Application1	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
New Page Application2	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
New Page Application3	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
Notification Application1	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
Notification Application2	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
Notification Application3	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
Branding Application	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
Screen Saver Application	Change to <input type="button" value="v"/>	<input type="button" value="v"/>	-
Service Level	-	Full	-
Pin Security Status	Change to <input type="button" value="v"/>	Weak or Expired	-
<div style="text-align: right;"> <input type="button" value="Preview"/> <input checked="" type="button" value="Save"/> <input type="button" value="Cancel"/> </div>			

Figure 33: Programming Multiline IP Sets

## NuPoint Configuration

### MiVB Setup for Connecting NuPoint

#### Licensing and Option Selection – SIP Licensing

The first step in setting up the MiVB for connecting to NuPoint is checking the **Extended Hunt Group option** to see if it is enabled. Refer to [Figure 2](#).

#### System Options

The ports that are used by NuPoint to connect to the MiVB are programmed as 5020 IP endpoints on the MiVB. NuPoint needs to be able to register these IP Endpoints in order to create the ports. Thus the Registration Access Code and Replacement Access Code need to be set on the MiVB. Set \*\*\* for the **Registration Access Code** and ### for the **Replacement Access Code**.

DTRX Autobaud Timeout	60
DTRX DSA Response Format	Comment
DTRX Herald Message	SX2000:
DTRX Inactivity Timeout	60
Email Server	
Email - Sender's Address	
External Hot Desking - Single Digit Mid Call Features	Yes
Feature Active Dial Tone - Call Forwarding	No
IDS sync maximum sets of results	5
Interconnect Checking for Conference Calls	No
Last Number Redial Source	All Trunks
Loop Signalling Trunks - Invalid DN Handling	Immediate
Maximum CO Trunks In A Conference	4
Maximum Parties In A Conference	5
Maximum Trunks In A Conference	4
Multiline Set Display 24 Hour Format	No
Multilingual Name Display	No
Music On Hold	Yes
Night Answer Prompt for Network Configuration	No
Number Of Forward Hops	10
Outgoing External Call Prefix For Applications	
Remote Help Server	
Resource Tuning Threshold	0
Ringing Cadence for Tie Line Calls	External
Route Optimization Attempts	3
Route Optimization Establishment Timer	10
Route Optimization Network Id	
Route Optimization Trailing Digits	2
Send Travelling Class Marks	No
Set Registration Access Code	***
Set Registration Auto DN Selection - Begin	
Set Registration Auto DN Selection - End	
Set Registration Auto DN Selection - Secondary	Not Assigned
Set Registration Security	
Set Replacement Access Code	###
Site Preference for Hot Desk Device	5020 IP
Speed Call Pause Duration	3
SUPERSET Callback Message Cancel Timer	
System Data Synchronization	Yes
System Name	Local_2
UK only - Standard for CLIP	
Voice Encryption Enabled	Yes
Voice/Video SRTP Encryption Enabled	No

Figure 34: System Option

---

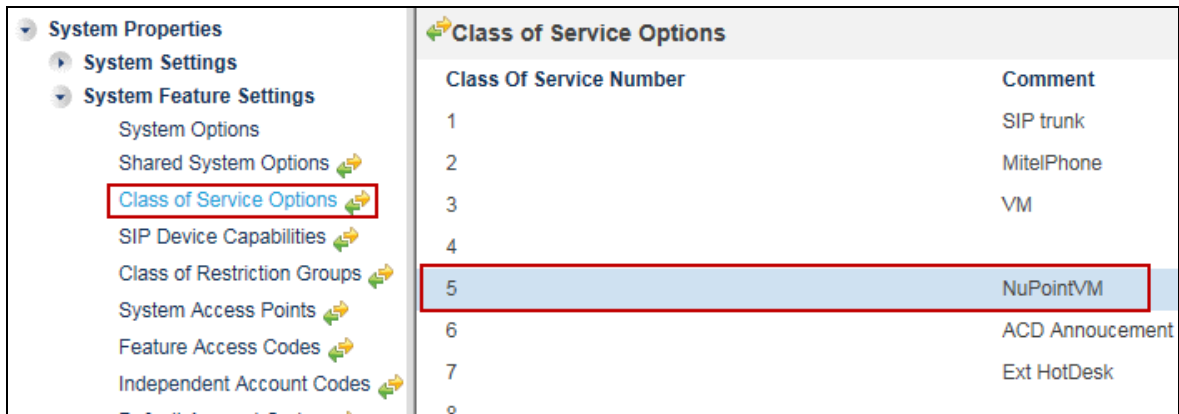
## Class of Service Options

**Navigation:** System Properties > System Feature Settings > Class of Service Options

The next step is to setup a Class of Services for NuPoint's inbound ports such as voicemail

In **Class of Service Options** for NuPoint Voicemail, enable the following:

- COV/ONS/E&M Voicemail Port
- HCI/CTI/TAPI Call Control Allowed
- HCI/CTI/TAPI Monitor Allowed
- Public Network Access via DPNSS



Class Of Service Number	Comment
1	SIP trunk
2	MitelPhone
3	VM
4	
5	NuPointVM
6	ACD Annoucement
7	Ext HotDesk
8	

**Figure 35: Class of Service Option for NuPoint Voice Ports**

## IP Endpoints used for NuPoint Ports

**Navigation:** Users and Devices > User and Services Configuration

5020 IP end points are created to be mapped to the incoming NuPoint Voice Ports.  
The numbers 2910~2911 are configured as NuPoint Voice Ports for this test.

The screenshot displays the 'User and Services Configuration' web interface. On the left, a search bar is set to 'Last Name'. Below it, a list of search results (9 matches) is shown. The entry 'Phone Service (2910)' is highlighted with a red box. The right pane shows the configuration for this service, with the 'Service Profile' tab selected. The configuration includes fields for 'Number' (2910), 'Service Label' (Phone Service), 'Directory Name' (port1,NuPoint), 'Prime Name' (No), 'Privacy' (No), 'Hot Desking User' (No), 'Device Type' (5020 IP, highlighted with a red box), 'Service Level' (Full), 'Home Element' (Local\_2), 'Secondary Element' (Not Assigned), and 'Local-only DN' (unchecked). A 'Save Changes' button is visible in the top right corner.

Field	Value
Number	2910
Service Label	Phone Service
Directory Name	port1,NuPoint
Prime Name	<input checked="" type="radio"/> No <input type="radio"/> Yes
Privacy	<input checked="" type="radio"/> No <input type="radio"/> Yes
Hot Desking User	<input checked="" type="radio"/> No <input type="radio"/> Yes
Device Type	5020 IP
Service Level	Full
Home Element	Local_2
Secondary Element	Not Assigned
Local-only DN	<input type="checkbox"/>

Figure 36: IP Endpoints Configuration



Class of Service value for Day, Night 1 and Night 2 of the IP end point should be given the Class of Service of incoming ports created earlier, which is 5 for this setup.

User Profile	Service Profile	Device Details	Service Details
Access and Authentication	Phone Applications	Keys	
	Day	Night 1	Night 2
Class of Service	5	5	5
Class of Restriction	1	1	1
External Hot Desking Enabled	<input checked="" type="radio"/> No <input type="radio"/> Yes		
External Hot Desking Dialing Prefix			
External Hot Desking Number			
DID Service Number			

Figure 37: IP Endpoints Class of Service

## Voicemail Hunt Group

**Navigation:** Users and Devices > Group Programming > Hunt Group

Create a Voicemail Hunt Group that will be used to call voicemail. All of the endpoints created in the section above will be added to this hunt group. Enter the hunt group number that will be used for voicemail and change the Hunt Group type to Voicemail. Here, Hunt Group 2900 is created.

### Hunt Groups

Hunt Group	Hunt Group Mode	Hunt Group Name	Hunt Group Priority	Hunt Group Type	Home Element	Secondary Element
2000	Circular		64	VoiceMail	Local_2	Not Assigned
2900	Circular		64	VoiceMail	Local_2	Not Assigned
2999	Circular		64	HCIReroute	Local_2	Not Assigned

Hunt Group

Local-only DN

Hunt Group Mode

Hunt Group Name

Class of Service - Day

Class of Service - Night1

Class of Service - Night2

Home Element

Secondary Element

2900

False

Circular

Local\_2

Not Assigned

< Page 1 of 1 >

Go to:

Add Member

### Hunt Group Members

Member Index	Number	Presence	Name	Home Element
1	2910	Present	port1,NuPoint	Local_2
2	2911	Present	port2,Nupoint	Local_2

Figure 38: Voicemail Hunt Group Configuration

### HCIReroute Hunt Group

Program the HCIReroute Hunt Group and set it to always route to the NuPoint Voicemail Hunt Group. The primary reason for setting up an HCIReroute is to enable MiTAI for MWI. 2999 is configured as HCIReroute Hunt Group in this test and Call Rerouting Always Alternative number 2 was modified to reroute everything to the Voicemail Hunt Group.

**Navigation:** Users and Devices > Group Programming > Hunt Group

👉 Hunt Groups						
Hunt Group	Hunt Group Mode	Hunt Group Name	Hunt Group Priority	Hunt Group Type	Home Element	Secondary Element
2000	Circular		64	VoiceMail	Local_2	Not Assigned
2900	Circular		64	VoiceMail	Local_2	Not Assigned
2999	Circular		64	HCIReroute	Local_2	Not Assigned

Hunt Group	2999
Local-only DN	False
Hunt Group Mode	Circular
Hunt Group Name	
Class of Service - Day	
Class of Service - Night1	
Class of Service - Night2	
Home Element	Local_2
Secondary Element	Not Assigned
First RAD	
Second RAD	
Night Answer RAD	
Hunt Group Priority	64
Hunt Group Type	HCIReroute
Phase Timer Ring	

Figure 39: HCIReroute Hunt Group

**Navigation:** Call Routing > Call Handling > Call Rerouting Always Alternatives

Call Rerouting Always Alternatives					
Always Alternative Number	Originating Device DID	Originating Device TIE	Originating Device CO	Originating Device INT	Directory Number
1	No Reroute	No Reroute	No Reroute	No Reroute	
2	Reroute	Reroute	Reroute	Reroute	2900
3	No Reroute	No Reroute	No Reroute	No Reroute	
4	No Reroute	No Reroute	No Reroute	No Reroute	
5	No Reroute	No Reroute	No Reroute	No Reroute	
6	No Reroute	No Reroute	No Reroute	No Reroute	
7	No Reroute	No Reroute	No Reroute	No Reroute	
8	No Reroute	No Reroute	No Reroute	No Reroute	

**Figure 40: Call Rerouting Always Alternatives**

**Navigation:** Call Routing > Call Handling > Call Rerouting

Call Rerouting						
Number	Call Rerouting - Day	Call Rerouting - Night1	Call Rerouting - Night2	Call Rerouting DND Type	Call Rerouting - 1st Alt.	Call Rerouting Alt.
0415	1	1	1	All	1	1
1000	1	1	1	All	1	1
1001	1	1	1	All	1	1
1003	1	1	1	All	1	1
1004	1	1	1	All	1	1
1005	1	1	1	All	1	1
1006	1	1	1	All	1	1
1007	1	1	1	All	1	1
1111	1	1	1	All	1	1
2000	1	1	1	All	1	1
2001	1	1	1	All	1	1
2002	1	1	1	All	1	1
2003	1	1	1	All	1	1
2004	1	1	1	All	1	1
2900	1	1	1	All	1	1
2910	1	1	1	All	1	1
2911	1	1	1	All	1	1
2999	2	2	2	All	2	2

**Figure 41: Call Rerouting**

---

## MiCollab NuPoint Configuration

### Network Elements

**Navigation:** From Server Manager, Applications > Users and Services

1. Select the Network Element tab
2. Click **Add**

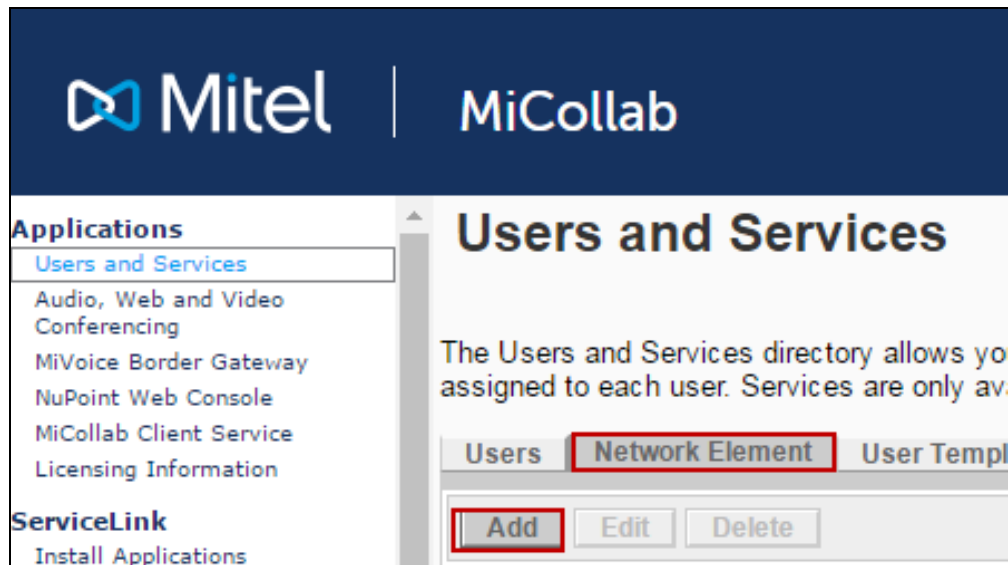


Figure 42: Add Network Element

1. Set **System Name**: **MitelPBX** is given in this test
2. Set **Network Address**: Enter the MiVB ICP IP address
3. Set **Credentials**: Enter the MiVB ICP administration credentials
4. Set **Registration Code**: **\*\*\*** is given which should match the **Set Registration Access Code** in the [System Options](#) section
5. Set **Replacement Code**: **###** is given which should match the **Set Replacement Access Code** in the [System Options](#) section
6. Set **Standard Phone COS**: **5** is given for all fields to match the Class of Service for Nupoint Voicemail port created in the [Class of Service Option](#) section
7. Set **Default COR**: **1** is given to all fields in this setup
8. Set **Call Forward Destination Directory Number**: **2900** is given, which is the Hunt Group Number for NuPoint Voicemail
9. Set **HCI Reroute Hunt Group Number for Mitai MWI**: **2999** is given to match the previous configuration
10. Click **Save**

**Edit Network Element**

Save Cancel

**Element Identification**

Type: Mitel 3300 ICP

\*System Name: 3300

\*Network Address: 10.35.32.2 Ping Test

Release:

Version:

**Credentials**

\*System Login: system

\*Password: .....

\*Confirm Password: .....

**System Properties**

\*Set Registration Code: \*\*\*

\*Set Replacement Code: ###

	Day	Night 1	Night 2
Standard Phone COS:	5	5	5
Record-A-Call COS:			
Default COR:	1	1	1

☐ Release NuPoint UM IP Integration License  
(Selecting this checkbox will remove this ICP from NuPoint UM when the save button is clicked. Leaving this checkbox unselected will apply any ICP changes to NuPoint UM when the save button is clicked.)

☒ Single Point Provisioning Enabled

**Voicemail**

Call Reroute First Alternative Number: 1

Call Forward Destination Directory Number: 2900

HCI Reroute Hunt Group Number for Mitai MWI: 2999

Save Cancel

**Figure 43: Network Element – Cont.**

## Voicemail Line Group

1. Click **NuPoint Web Console**

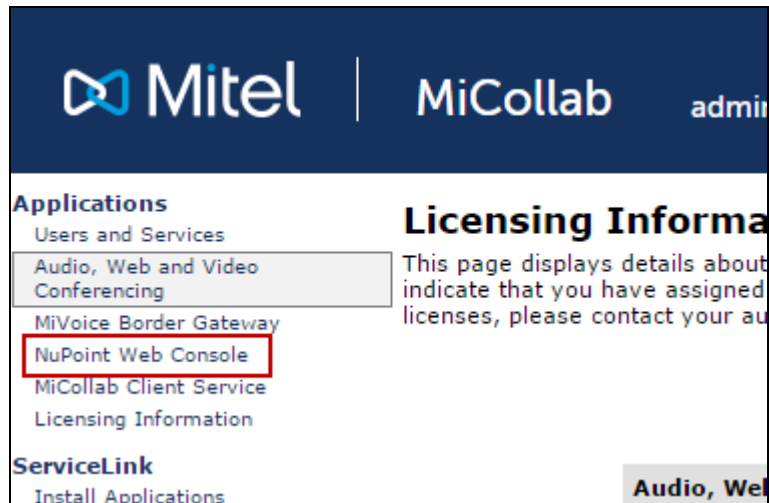


Figure 44: Voicemail Line Group Configuration

**Navigation:** Offline Configuration > Line Groups

2. Click **Add**

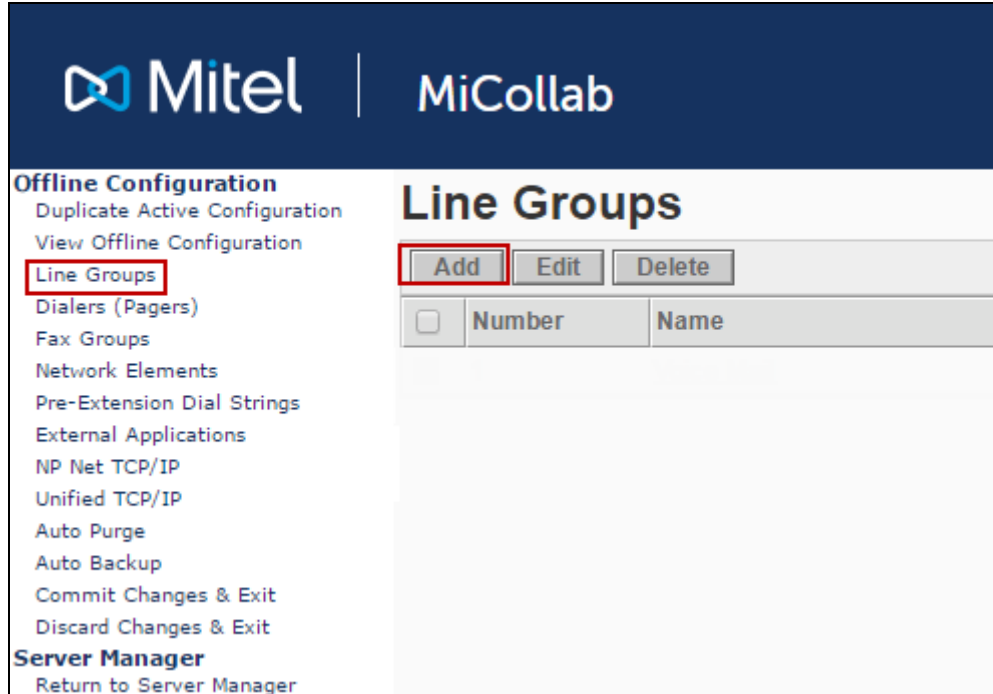
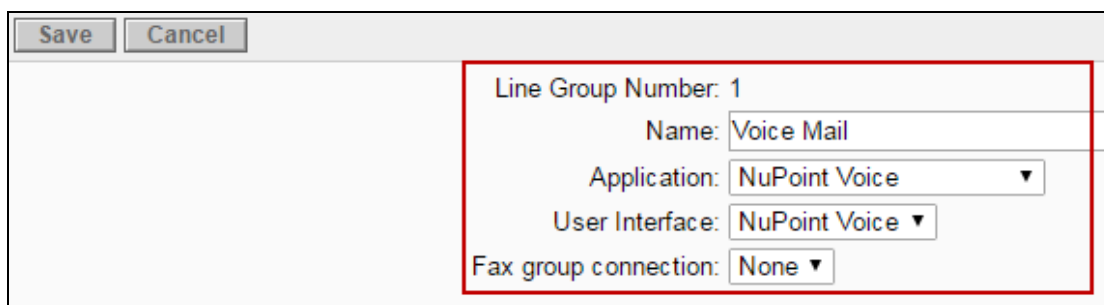


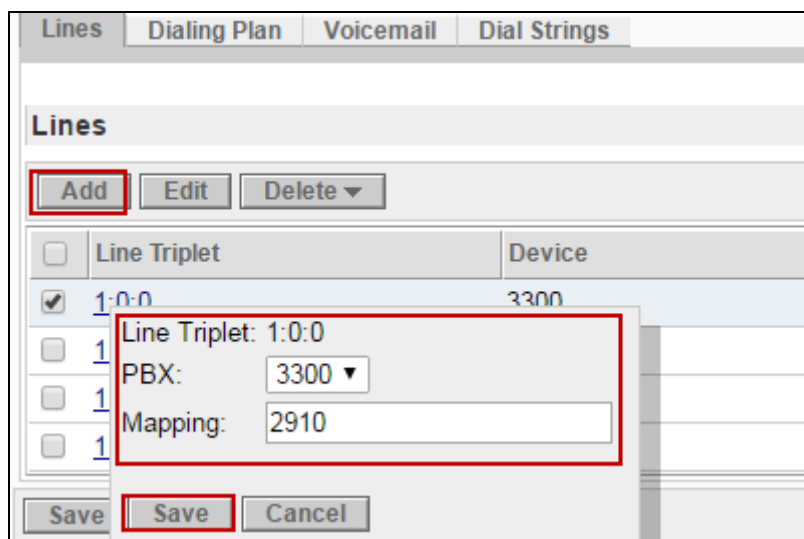
Figure 45: Voicemail Line Group Configuration – Cont.

2. On the Add Line Group web page, click **Next Available** to fill in the **Line Group Number** (the value should be 1 as this is the first line group being created)
3. Enter a **Name** such as Voicemail to describe for what the line group will be used
4. Choose NuPoint Voice for the **Application**
5. Choose NuPoint Voice for the **User Interface**



**Figure 46: Adding Line Group**

1. Click **Add** under the Lines heading. This will bring up the Line Triplet dialogue box
2. Click **Next Available** to get the next available Line Triplet (1:0:0 should come up since this is the first time line triplets are being assigned).
3. Select PBX: MitelPBX. Enter the first extension number that was created in the section [IP Endpoints](#) used for NuPoint Ports in the Mapping field.
4. Click **Save**



**Figure 47: Adding Lines**

1. Click on the **Dialing Plan** tab on the Add Line Group page. The dialing plan consists of nine numbers separated by commas and Length of extensions



are configured as Variable except 9 for which 3 is configured, this was the default setting. Mailboxes 999 and 998 are created, 998 is the default administrative mailbox and 999 is the default attendant mailbox.

2. Click **Save**

Length of extensions starting with...		
1 :	Variable	Standard
2 :	Variable	Standard
3 :	Variable	Standard
4 :	Variable	Standard
5 :	Variable	Standard
6 :	Variable	Standard
7 :	Variable	Standard
8 :	Variable	Standard
9 :	3 digits	Standard

☐ Classic Mode

Dialing Plan: v,v,v,v,v,v,v,v,3

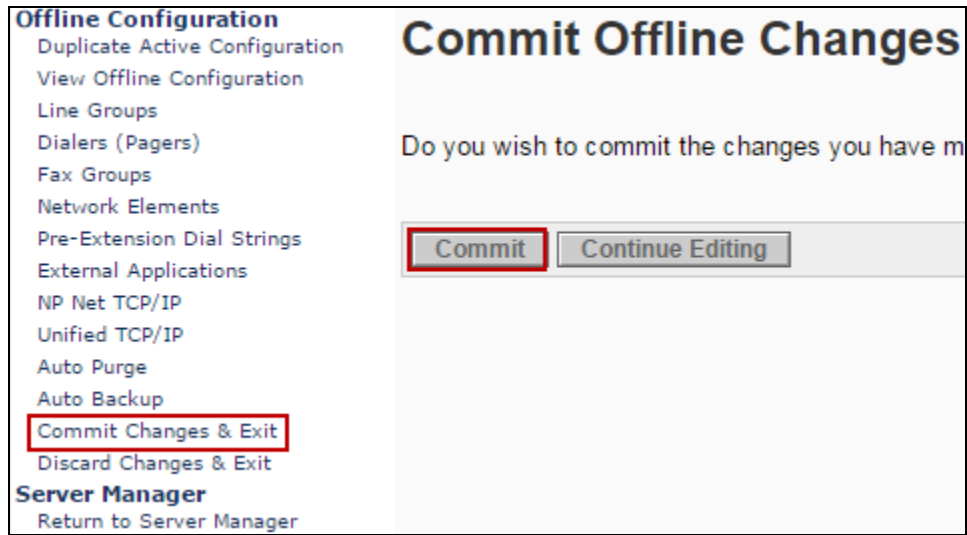
**Save** Cancel

Figure 48: Adding Dial Plan

1. Click the **Commit changes & Exit** link under the Offline Configuration heading

---

2. Click **Commit**



**Figure 49: Committing Offline Changes**

3. Click the **Activate** link at the top of the page
4. On the Activate Offline Configuration page, deselect the check boxes for:
  - a. Wait for MWI queue to empty
  - b. Wait for Pager queue to empty
5. Click **Activate**

## **Adding Mailboxes**

NuPoint with MAS and Single Point Provisioning allows for programming MiVB phones, users and NuPoint Mailboxes from the MAS interface. It is assumed MiVB phones and users were configured in the [MiVoice Business Configuration Notes](#) Section and this chapter only covers adding mailboxes.

**Navigation:** Mailbox Maintenance > Mailboxes

1. Click **Add**

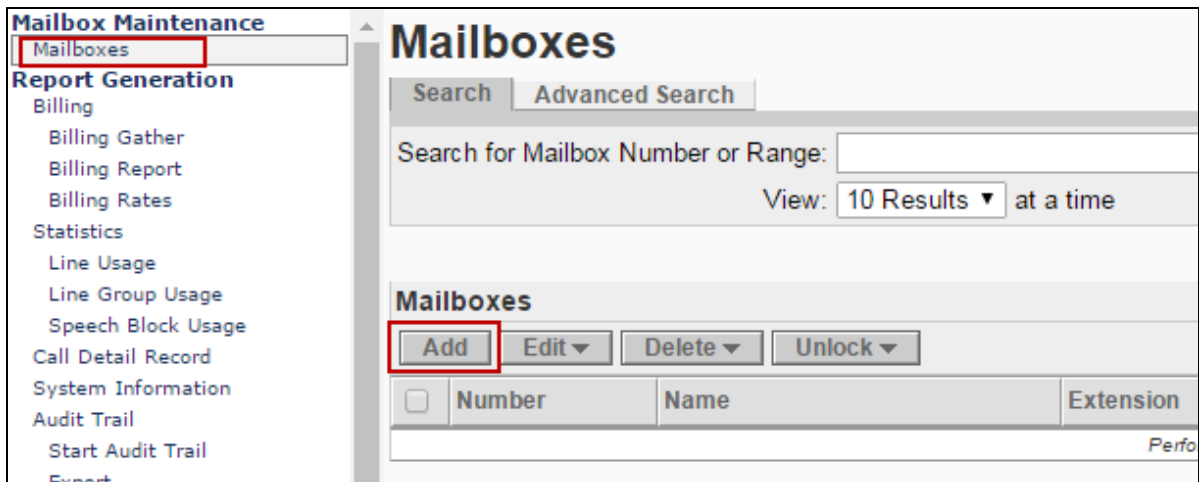


Figure 50: Add Mailbox

2. **Mailbox Number** 1000 is created for this test
3. Under the **General** tab, set the proper **Name**, **Passcode** and associated MiVB phone/user as **Extension**.

## Add Mailbox(es)

Create Mailbox(es)

Mailbox Number(s): 1000

Copy from another mailbox:

**Basic** [Advanced](#)

**General**

### Personal Information

Name: 1000

**IMPORTANT NOTE**  
If you expect your callers to use "Dial By Name " with First Name then  
Enter the name in following format: <First Name > <Last Name >.  
If you expect your callers to use "Dial By Name" with Last Name then  
Enter the name in following format: <Last Name >, <First Name >.  
Note that the comma is ESSENTIAL in this case.

Passcode:  The user will be asked to change the pas

Extension: 1000

Attendant Extension:

### Unified Messaging Information

UM Audio Encoding: ADPCM (Smallest files, default value)

UM-SMTP Email Address:

UM-Web View Email Address:

**Basic** [Advanced](#)

Figure 51: Add Mailbox – Cont.

4. Select the **Message Waiting** tab
5. Select **Mitai Messaging** as **Type**
6. Click **Save**
7. Click **Done** when pop-up window shows the mailbox was added successfully

**Edit Mailbox - 1000**

Save Cancel Basic [Advanced](#)

General Class of Service **Message Waiting**

**Message Waiting #1**

Type: Mitai Messaging ▾  
▸ Details

**Message Waiting #2**

Type: None ▾  
▸ Details

**Message Waiting #3**

Type: None ▾

Save Cancel Basic [Advanced](#)

Figure 52: Add Mailbox – Cont.

## MiVoice Border Gateway Configuration Notes

When configuring MiVoice Border Gateway (MIVoice BORDER GATEWAY), specify the Network profile, gateway mode used in this setup

**Navigation:** Applications > MiVoice Border Gateway > System Configuration > Network Profiles

1. Click the “→” next to **Server-gateway configuration on the network edge**
2. Click **Apply**

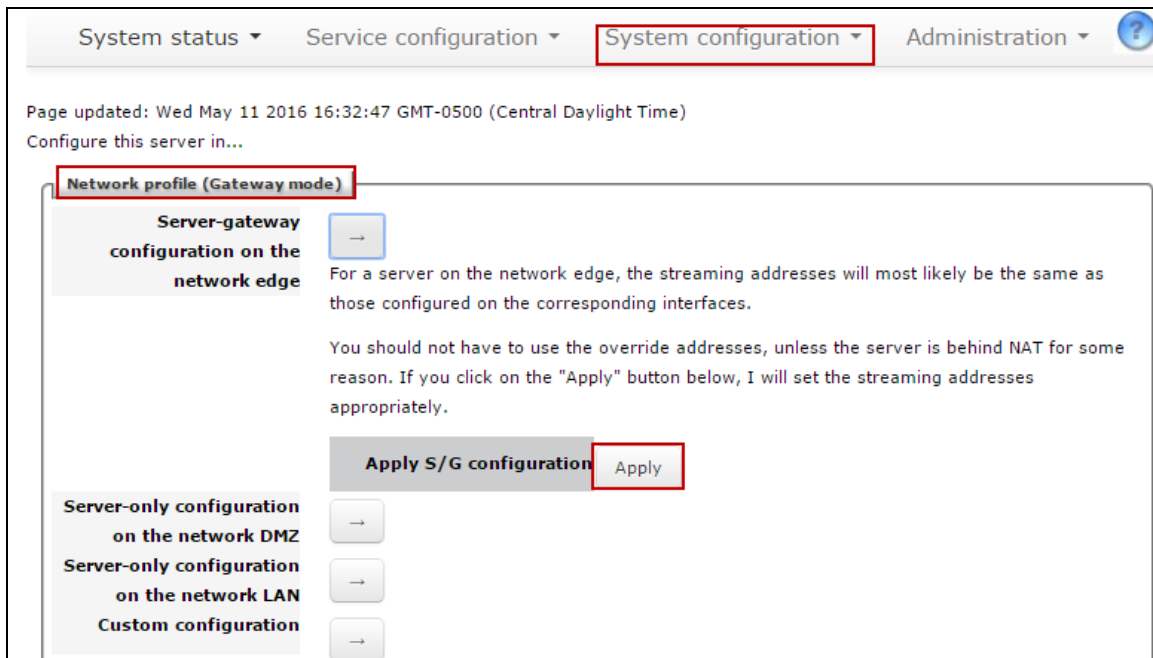


Figure 53: Network Profiles

In order to make the mid-call feature works for External Hot Desk User, setup KPML username and password

**Navigation:** Service Configuration > Settings

1. Click **Edit**
2. Set **KPML Username:** administrator is given which is the same as **Subscription User Name** created in the section [SIP Peer Profile](#)
3. Set **KPML Password:** Enter the same password as **Subscription Password** created in the section [SIP Peer Profile](#)

The screenshot shows the 'SIP options' configuration page. It is organized into several sections:

- SIP support:** Includes checkboxes for UDP (checked), TCP, and TCP/TLS (disabled).
- Registration Mode:** A dropdown menu set to 'Max Set-Side'.
- Set-side registration expiry time:** A text input field containing '240'.
- ICP-side registration expiry time:** A text input field.
- Allowed URI names:** A text input field with an 'Add another' button below it.
- PRACK support:** A checkbox that is unchecked.
- Send options keepalives:** A dropdown menu set to 'Only behind NAT'.
- Options interval:** A text input field containing '20'.
- Challenge methods:** A list box containing 'Invite', 'Subscribe', 'Refer', and 'Prack', with 'Invite' selected.
- Local streaming:** A checkbox that is unchecked.
- Codec support:** A dropdown menu set to 'Unrestricted'.
- RTP framesize:** A dropdown menu set to '20ms'.
- Set-side RTP security:** A dropdown menu set to 'Allow'.
- Icp-side RTP security:** A dropdown menu set to 'Disable'.
- KPML username:** A text input field containing 'administrator'.
- KPML password:** A text input field with masked characters (dots).
- Confirm KPML password:** A text input field with masked characters (dots).
- Permit weak passwords:** A checkbox that is unchecked.

A red rectangle highlights the 'KPML username', 'KPML password', and 'Confirm KPML password' fields.

**Figure 54: MBG Settings**

Then identify the working MiVoice Business ICP where to forward SIP messages to and then to configure the SIP trunk.

**Navigation:** MiVoice Border Gateway > Service Configuration > ICPs

The screenshot shows the 'Service configuration' tab selected in the top navigation bar. Below the navigation bar, there is a status bar with the text 'Page updated: Wed May 11 2016 16:35:06 GMT-0500 (Central Daylight Time)' and a link to the 'Diagnostics' page. The main content area is titled 'ICP Information' and contains a table with the following data:

Default for MiNet	Default for SIP	Name	Hostname or IP address	Type	Installer password	SIP capabilities	Indirect call recording capable		
<input checked="" type="radio"/>	<input checked="" type="radio"/>	3300 32.2	10.35.32.2	MiVoice Business		UDP			

**Figure 55: MIVoice BORDER GATEWAY Configuration**

1. On **ICPs** page, ensure that the “working” MiVoice Business is configured
2. If needed, click **Add ICP** link and add a new Mitel switch
3. Click **Update** Default ICPs
4. To add a new SIP trunk:
  - a. Select the **Service Configuration** tab and then select **SIP trunking**
  - b. Click **Add a SIP trunk** link

The screenshot shows the 'Service configuration' tab selected in the top navigation bar. A dropdown menu is open, showing the following options: ICPs, MiNet devices, SIP devices, **SIP trunking** (highlighted with a red box), WebRTC, and Application integration. Below the dropdown menu, there is a section titled 'SIP trunk information' with a red box around the '+' icon. The table below has the following data:

Name	Remote endpoint	Number of routing rules	PRACK support	Remote RTP (ms)
------	-----------------	-------------------------	---------------	-----------------

**Figure 56: SIP Trunking Configuration**



Enter the SIP trunk details as follows:

1. Set **Name**: IntelPeer is given in this setup
2. Set **Remote Trunk Endpoint Address**: Enter the IP address / FQDN for your deployment
3. Set **Remote Trunk Endpoint Port**: 5060 is used
4. Set **Remote RTP Framesize (ms)**: This is the Packetization rate you want to set on this trunk. Set to Auto.
5. Set **PRACK Support**: Disabled for this configuration
6. Set **Routing rules**: This allows routing of calls with certain range of dialed digits to the selected MiVoice Business ICP
7. The remaining settings are optional and could be configured as required
8. Click **Save**

Manage SIP trunk

Name: IntelPeer

Remote trunk endpoint address: 68.68.123.174

Remote trunk endpoint port: 5060

Options keepalives: Never

Options interval: 60

Rewrite host in PAI: ☒

Idle timeout (s): 3600

Local streaming: ☐

Log verbosity: Use master setting

Authentication password:

Set-side RTP security: Allow

Accept traffic from any port: ☐

Remote RTP framesize (ms): 20ms

RTP address override: ---

PRACK support: Disabled

Authentication username:

Confirm authentication password:

Icp-side RTP security: Disable

Search routing rules:

Next Previous

Note, if you modify your routing rules, you must save them before changing pages or navigating elsewhere, or those changes will be lost.

Page: 1 of 1

Jump to page: 1

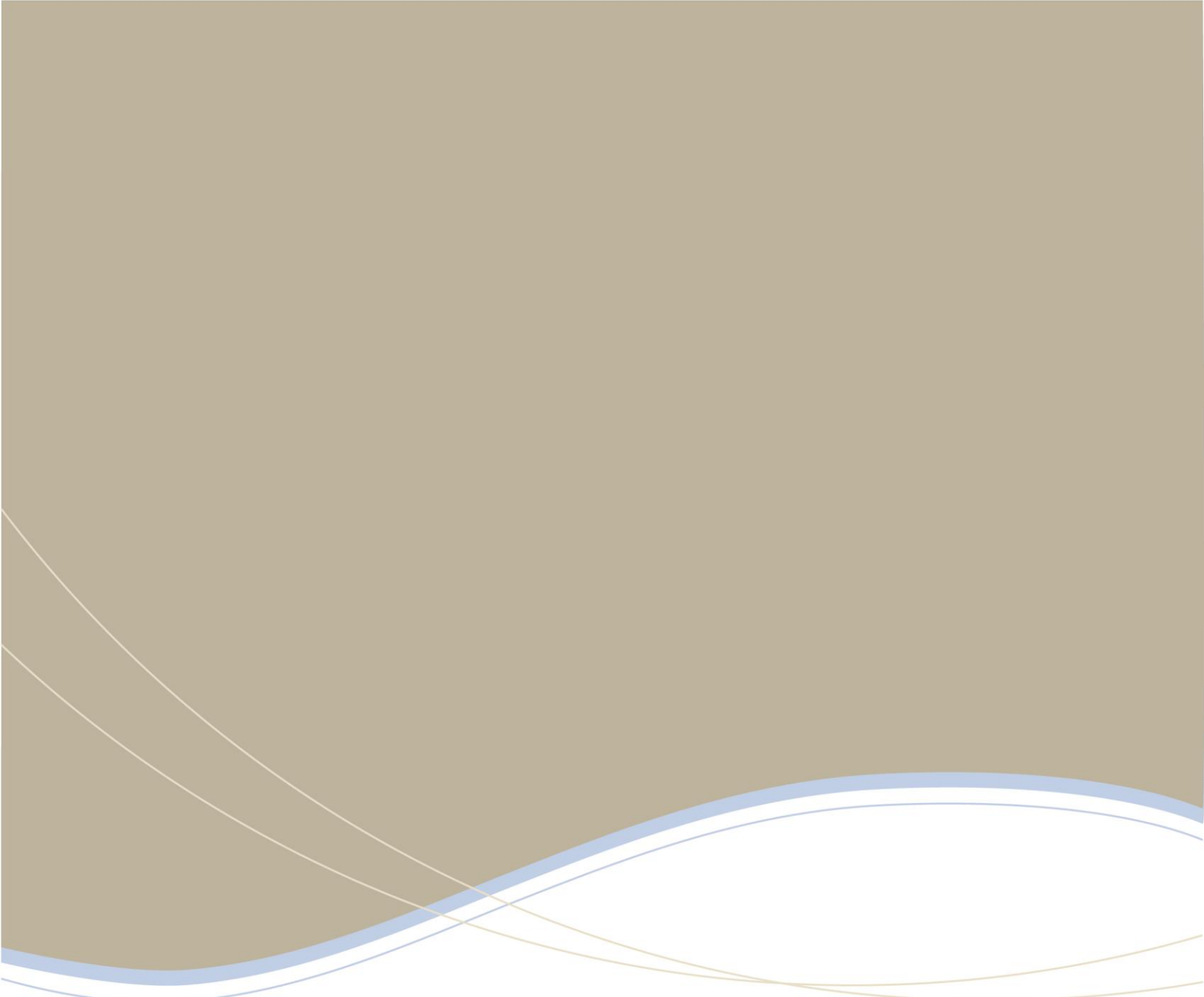
Rules per page: 10

First Prev Next Last

Match	Rule	Primary	Secondary	
1 Request URI	917512XXXX	3300 32.2	-----	Raise Prepend Delete Lower Append

Save

Figure 57: SIP Trunk Configuration Settings



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