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

Simply Communicating®

NOTICE

TRADEMARKS

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

May 2016 12-4940-00XXX

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Business to connect to IntelePeer 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 IntelePeer SIP Trunking

Interop Status

The Interop of IntelePeer 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 IntelePeer SIP Trunking achieved is:

COMPATIBLE	The most common certification which means IntelePeer 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.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between IntelePeer 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	IntelePeer	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 Pans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through IntelePeer 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 IntelePeer 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 IntelePeer and their PSTN gateway to and from Teleworker extensions	☑
Video	Making and receiving a call through IntelePeer 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 IntelePeer SIP Trunking is connected to the MiVoice Business.

Feature	Problem Description
Video Call	IntelePeer does not support video calls **Recommendation**: Contact IntelePeer for update on this feature

Network Topology

This diagram shows how the testing network is configured for reference

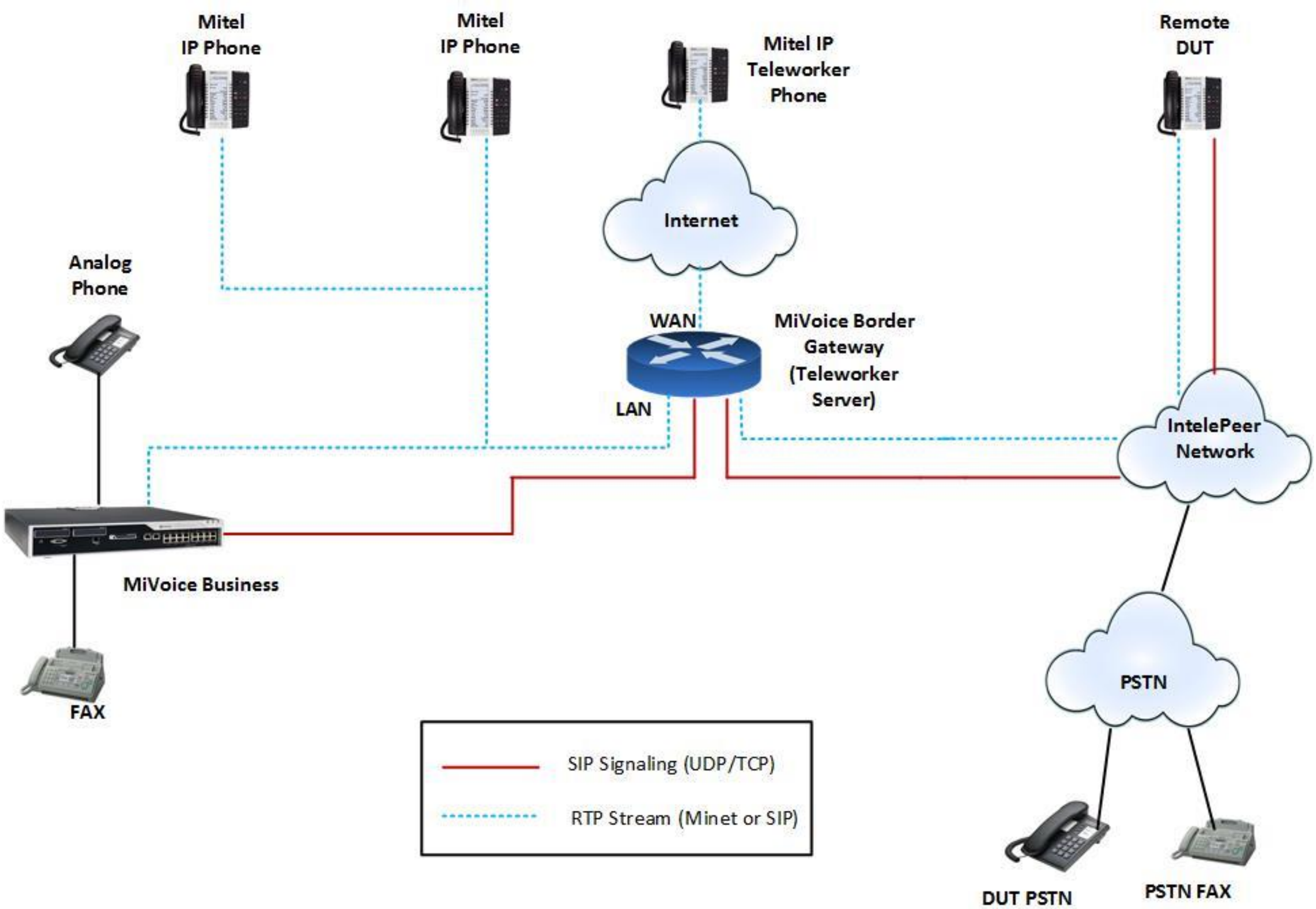

Figure 1: Network Topology

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 IntelePeer SIP Trunking was configured in our test environment.

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.

MiVoice Business Configuration Notes

The following steps show how to program a MiVoice Business to interconnect with IntelePeer 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 IntelePeer 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

Local Limits

Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Licenses Allowed	Can be Over Allocated
Users						
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
Messaging						
Embedded Voice Mail	6	16	0	16	Unrestricted	Yes
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	Yes
Trunking/Networking						
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
Others						
IDS Connection	1	Yes	0	1	Unrestricted	Yes
MLPP	0	No	0	0	Unrestricted	No

Configuration Options

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.

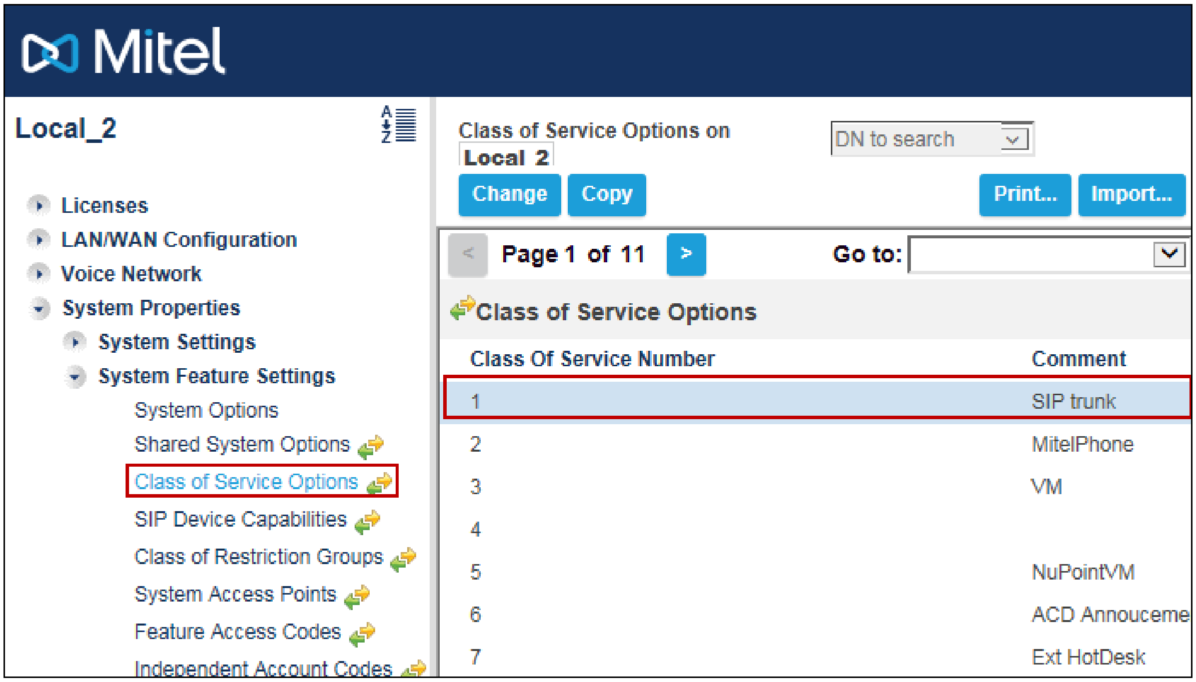

Figure 3: Class of Service

Class of Service for Trunk

General

General | Advanced

Class Of Service Number	1
Comment	SIP trunk
ACD	
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
Announce	
Call Announce Line	No
Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack Allowed	No
Busy Override	
Busy Override Security	Yes
Disable Executive Busy Override Tone	No
Executive Busy Override	No
Call Control Timer	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10
Lockout Timer	45
Call Duration	
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
Call Forwarding/Rerouting	
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
Call Hold	
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
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	No
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Privacy	
Call Privacy	No
Calling Party Name Substitution	Yes
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	Yes
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	No
Campon	
Auto Campon Timer	10
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
Display	
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
Fax	
Campon Tone Security	Yes
External Trunk Standard Ringback	No
Fax Capable	Yes
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
Hot Desk	
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
Miscellaneous	
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
Paging	
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

Account Code	
Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes
Attendant	
Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No
Conference	
Conference Call	Yes
Disable Conference Join Tone	No
DND	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No
Emergency	
Emergency Call - Audio Level for Set	Ringer
Emergency Call Notification - Audio	No
Emergency Call Notification - Visual	No
Group Presence	
Group Presence Control	No
Group Presence Third Party Control	No
Hotel	
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
Message Waiting	
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

Miscellaneous	
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
Phonebook	
Phonebook Lookup - Default to User Location	No
Phonebook Lookup - Display User Location	No
Record A Call	
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
ACD	
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
Announce	
Call Announce Line	No
Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack Allowed	No
Busy Override	
Busy Override Security	No
Disable Executive Busy Override Tone	No
Executive Busy Override	No
Call Control Timer	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10
Lockout Timer	45
Call Duration	
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
Call Forwarding/Rerouting	
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
Call Hold	
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
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	No
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Privacy	
Call Privacy	No
Calling Party Name Substitution	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	Yes
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes
Campon	
Auto Campon Timer	0
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
Display	
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
Fax	
Campon Tone Security	No
External Trunk Standard Ringback	No
Fax Capable	No
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
Hot Desk	
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
Miscellaneous	
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
Paging	
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

Account Code	
Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes
Attendant	
Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No
Conference	
Conference Call	Yes
Disable Conference Join Tone	No
DND	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No
Emergency	
Emergency Call - Audio Level for Set	Ringer
Emergency Call Notification - Audio	No
Emergency Call Notification - Visual	No
Group Presence	
Group Presence Control	No
Group Presence Third Party Control	No
Hotel	
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
Message Waiting	
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

Miscellaneous	
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
Phonebook	
Phonebook Lookup - Default to User Location	No
Phonebook Lookup - Display User Location	No
Record A Call	
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 IntelePeer SIP Trunking. In this example, the soft switch is reachable by an IP Address and is defined as “IntelePee” 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.

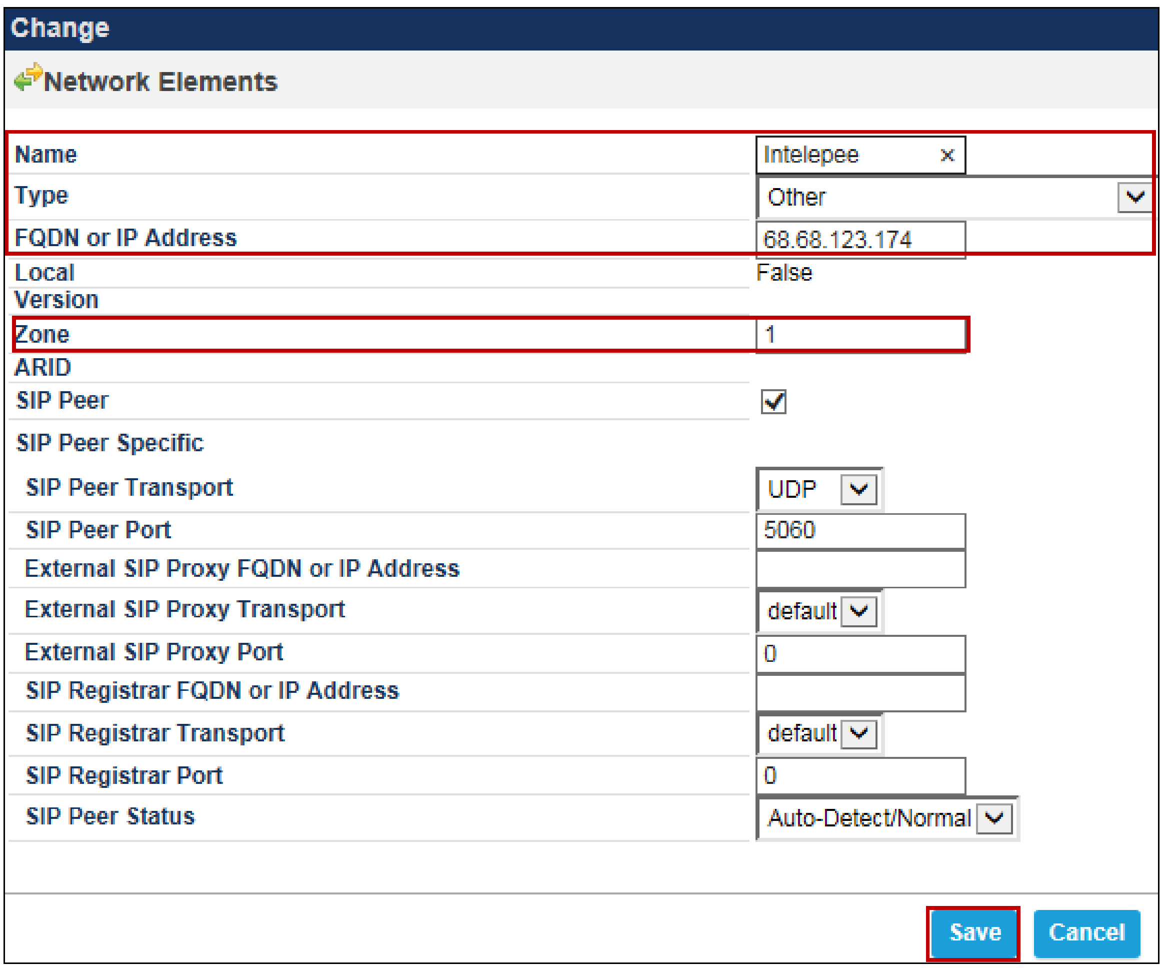

Figure 16: Network Element Assignment

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).

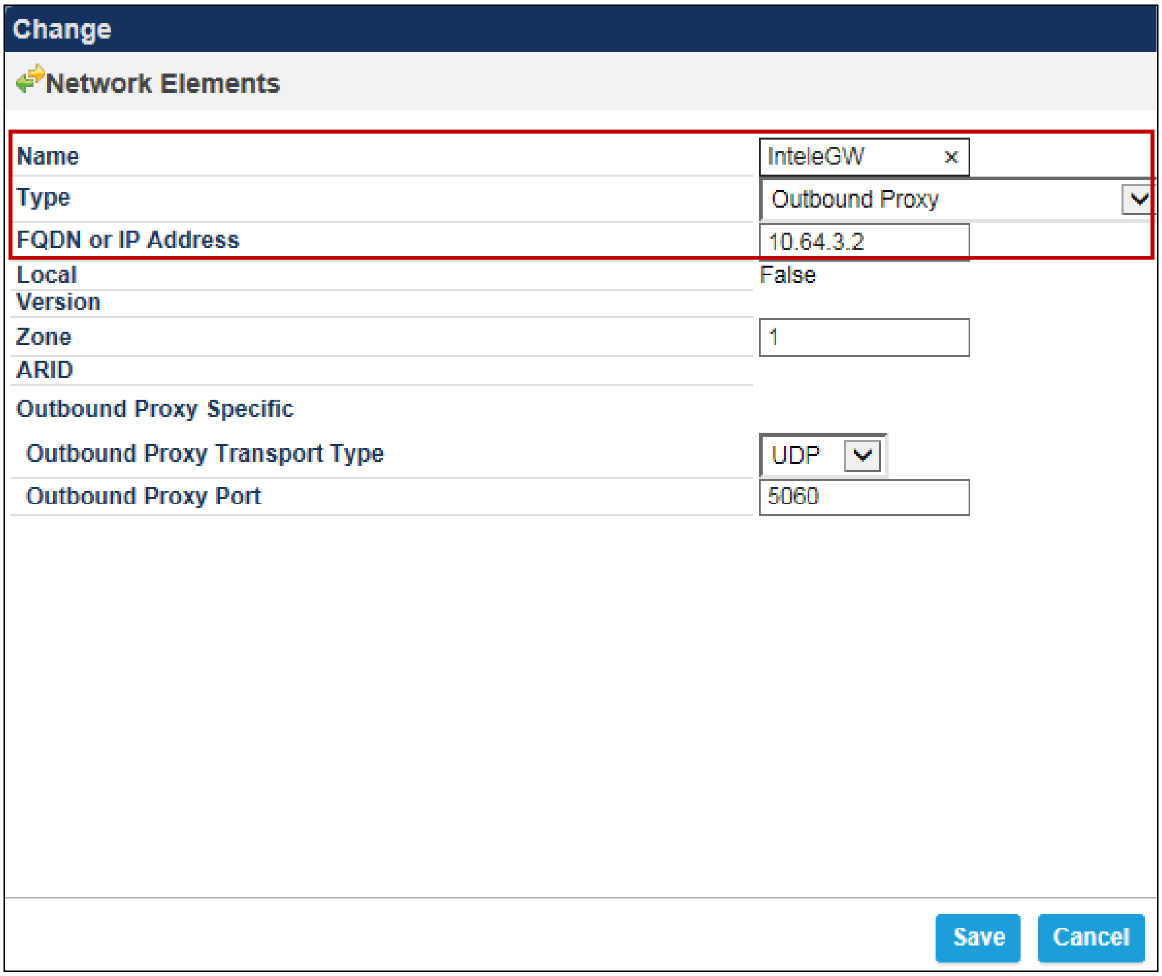

Figure 17: Network Element Assignment (Proxy)

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

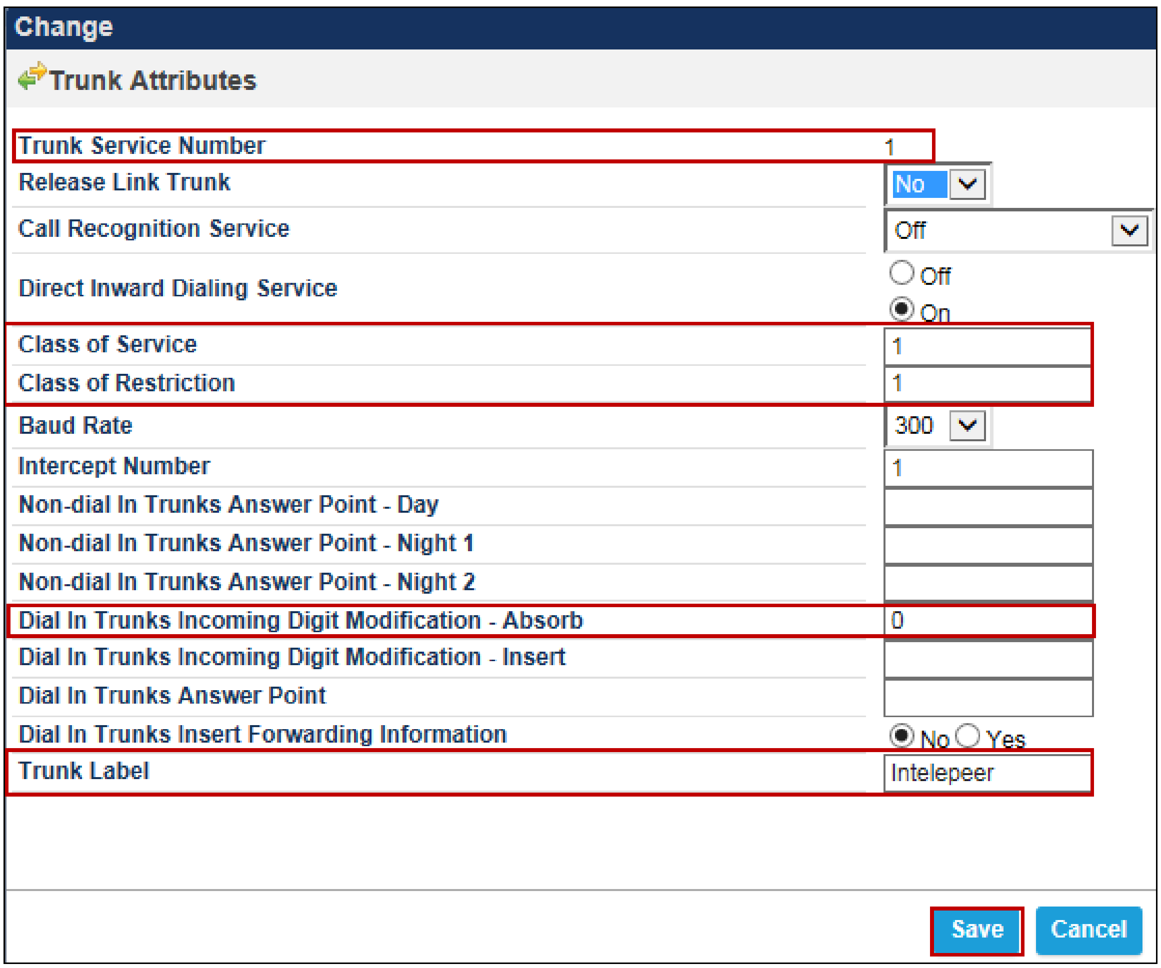

Figure 18: Trunk Attributes

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

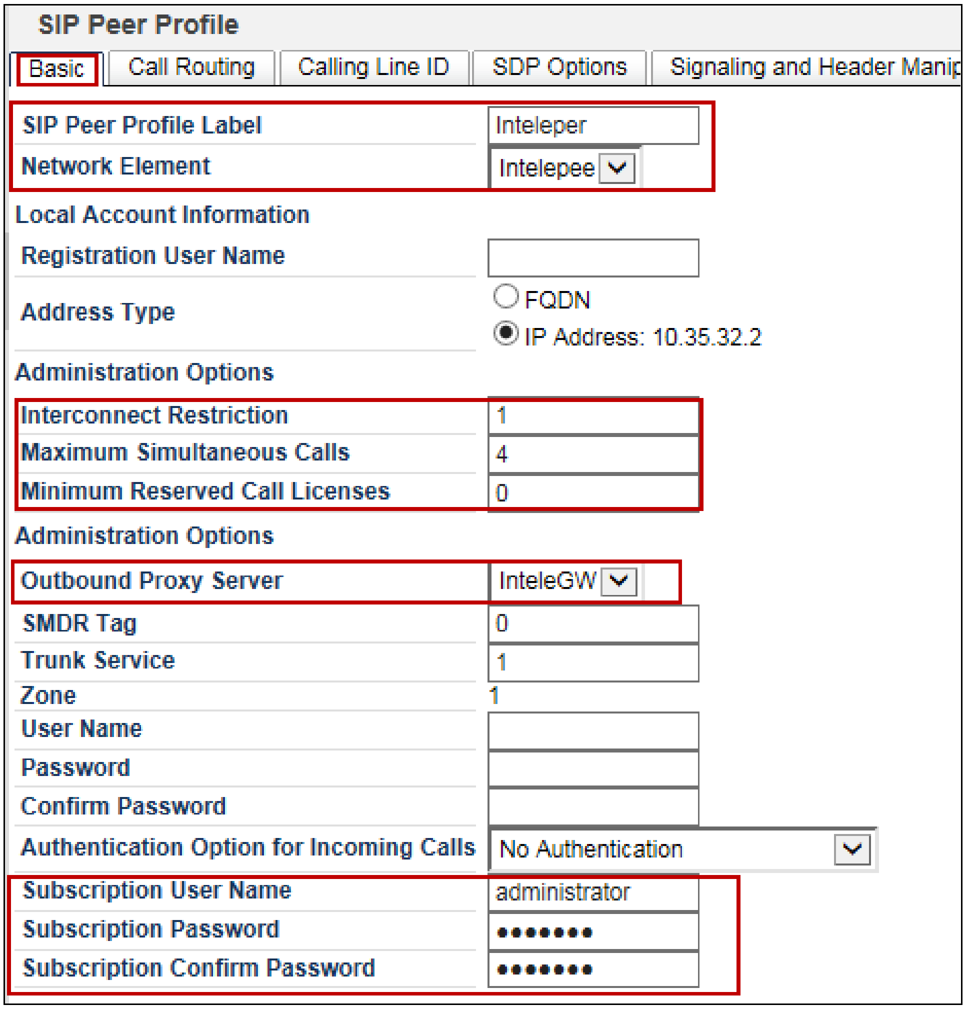

Figure 19: SIP Peer Profile – Basic

Call Routing

All the parameters are configured as shown

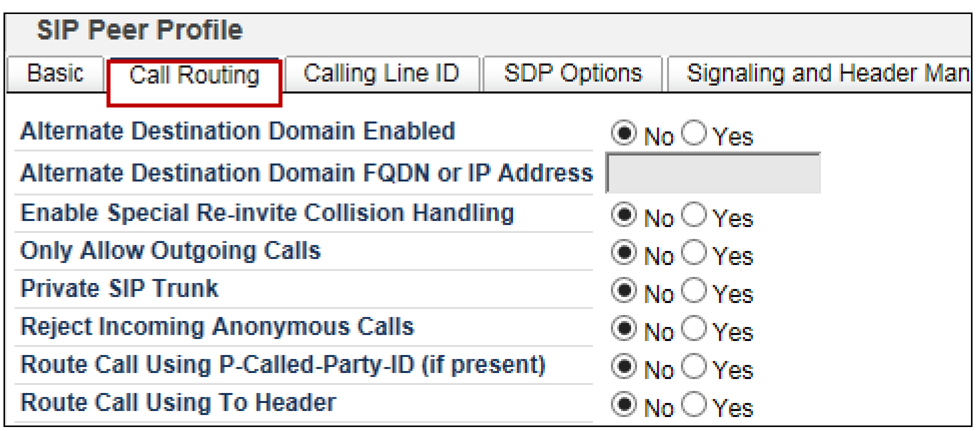

Figure 20: SIP Peer Profile Assignment – Call Routing

Calling Line ID

All the parameters are configured as shown

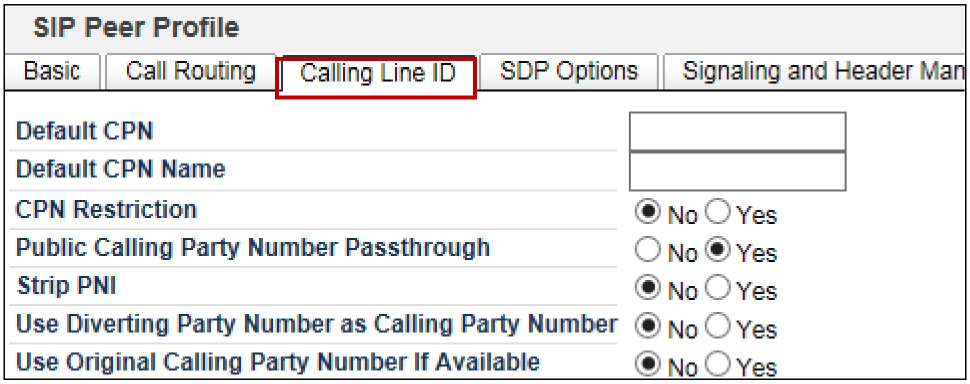

Figure 21: SIP Peer Profile Assignment – Calling Line ID

SDP Options

All the parameters are configured as shown

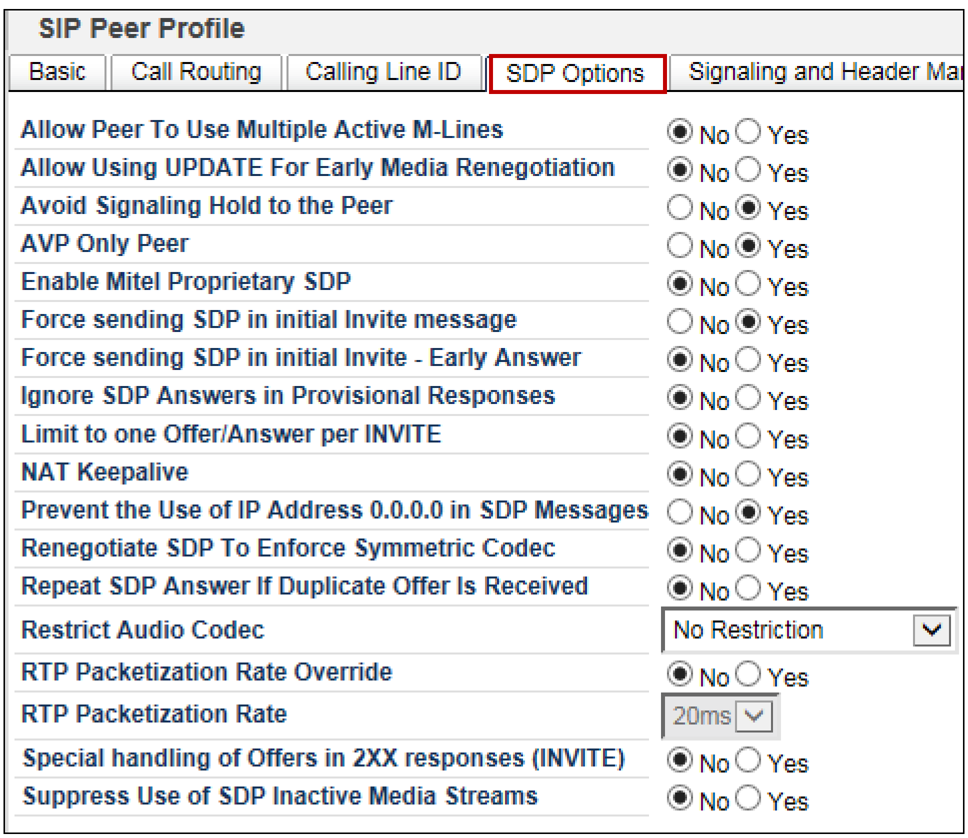

Figure 22: SIP Peer Profile Assignment – SDP Options

Signaling and Header Manipulation

All the parameters are configured as shown

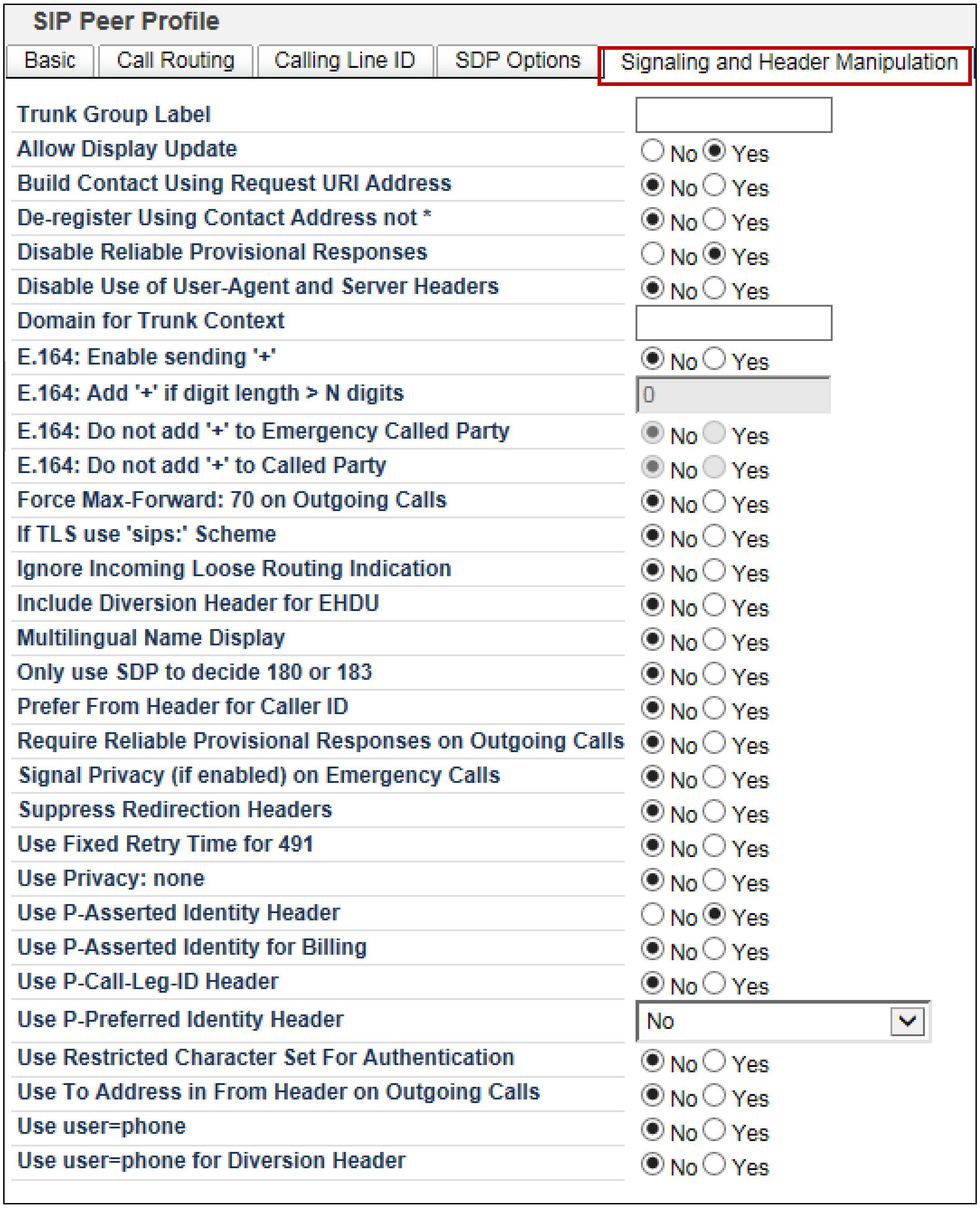

Figure 23: SIP Peer Profile Assignment – Signaling and Header Manipulation

Timers

All the parameters are configured as shown

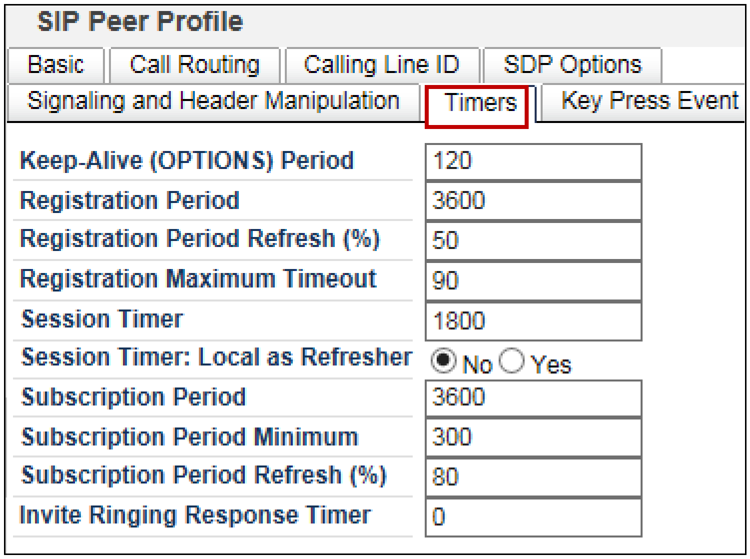

Figure 24: SIP Peer Profile Assignment - Timers

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*

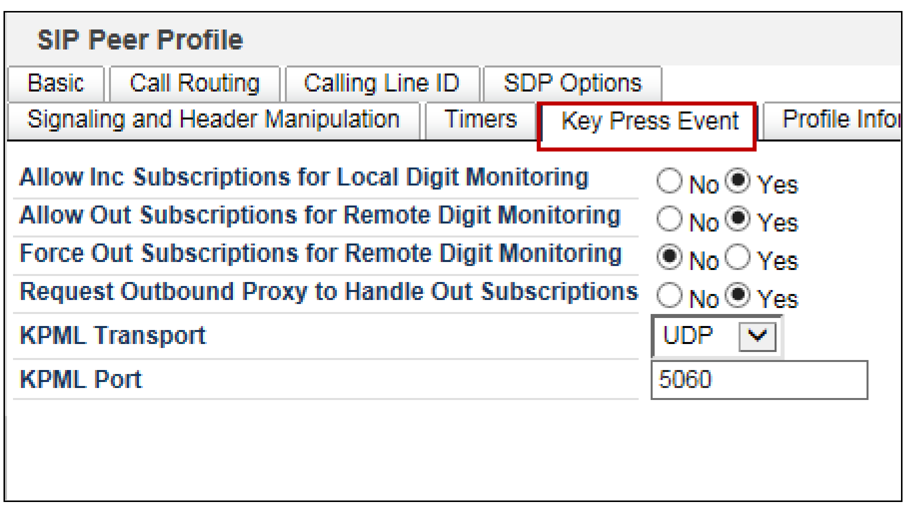

Figure 25: SIP Peer Profile Assignment – Outgoing DID Ranges

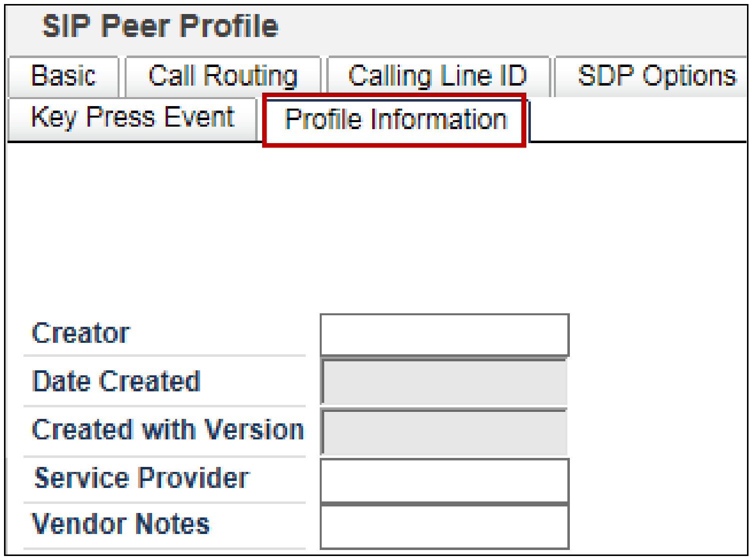

Figure 26: SIP Peer Profile Assignment – Profile Information

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 IntelePeer 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**

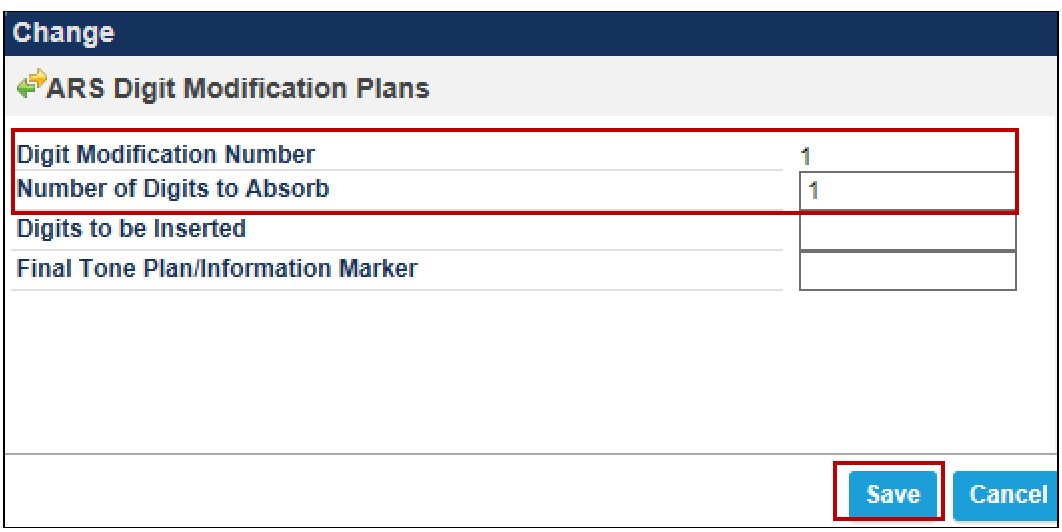

Figure 27: Digit Modification Assignment

ARS Routes

Navigation: Call Routing > Automatic Route Selection > ARS Routes

1. Create a route for SIP Trunks connecting a trunk to IntelePeer. 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**

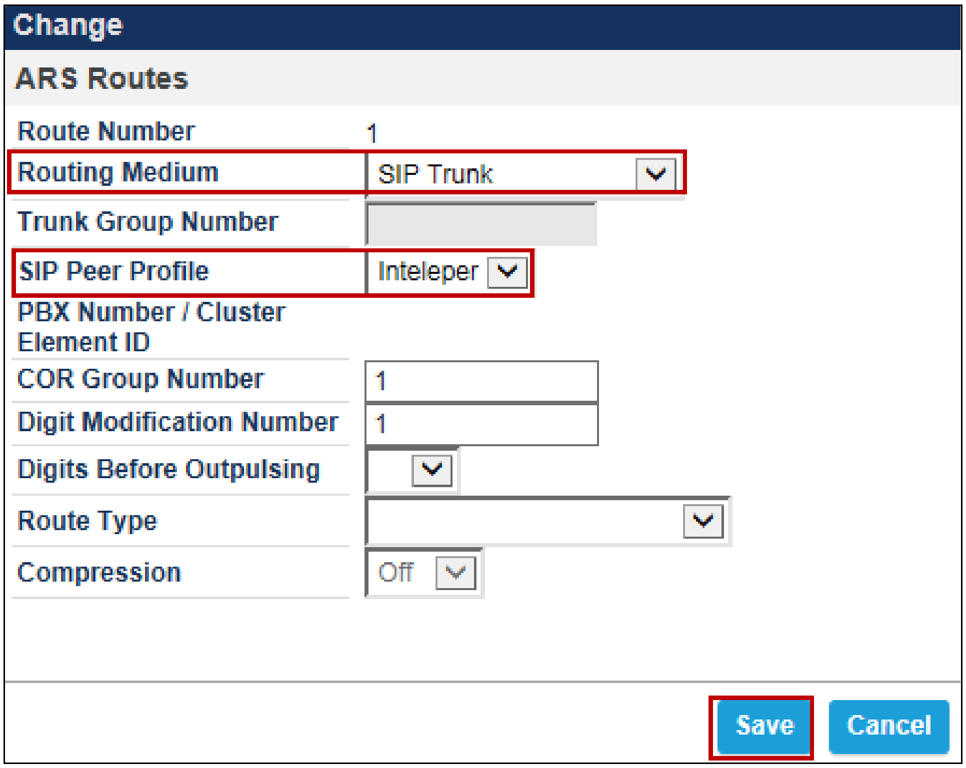

Figure 28: SIP Trunk Route Assignment

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 IntelePeer via route 1 configured in the previous step.

1. Click **Save**

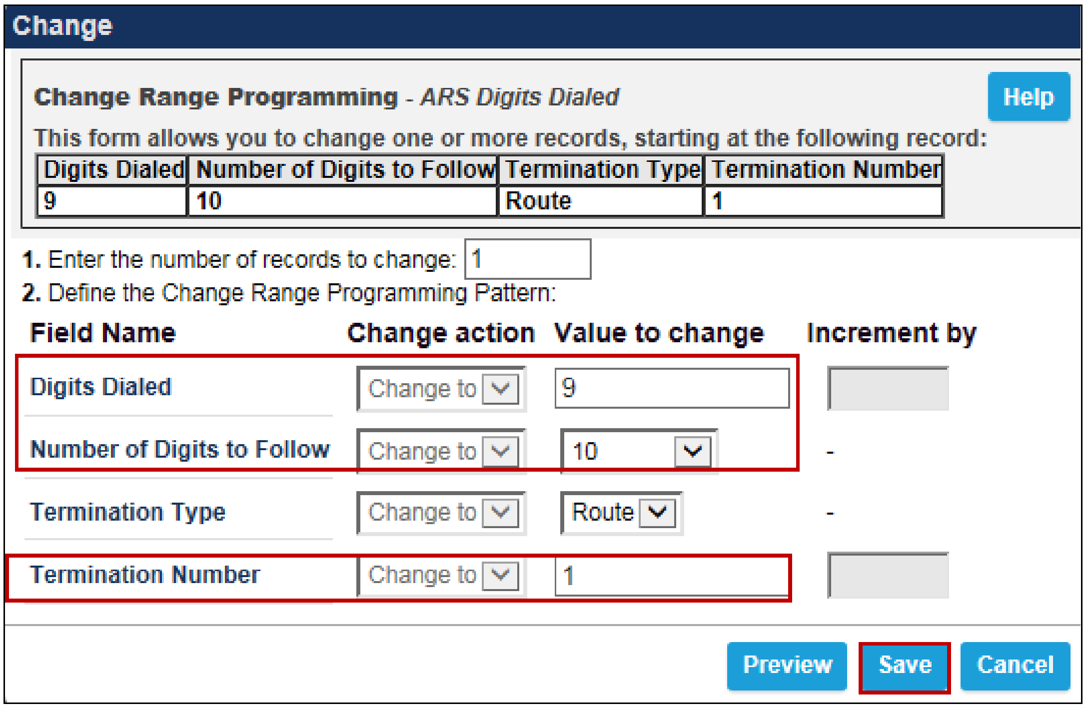

Figure 29: ARS Digit Dialed Assignment

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

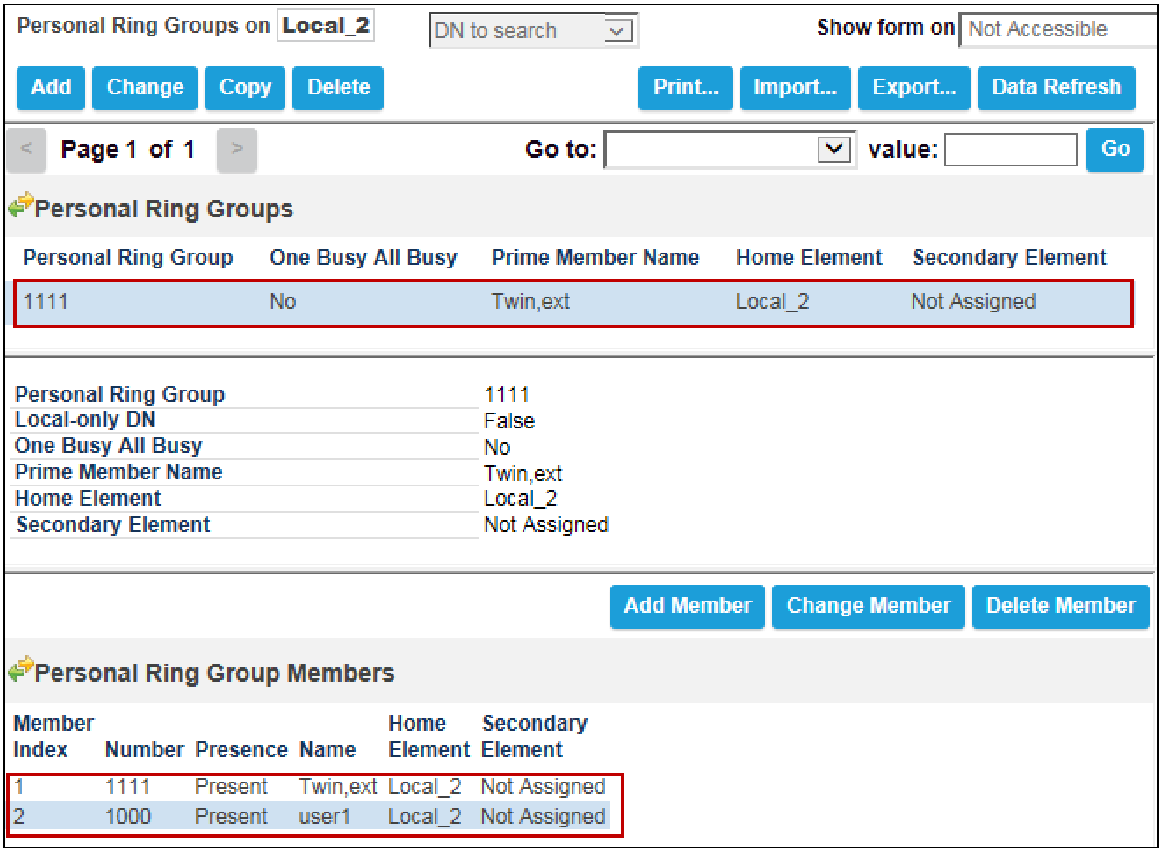

Figure 30: Personal Ring Groups

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 Histor Recor
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

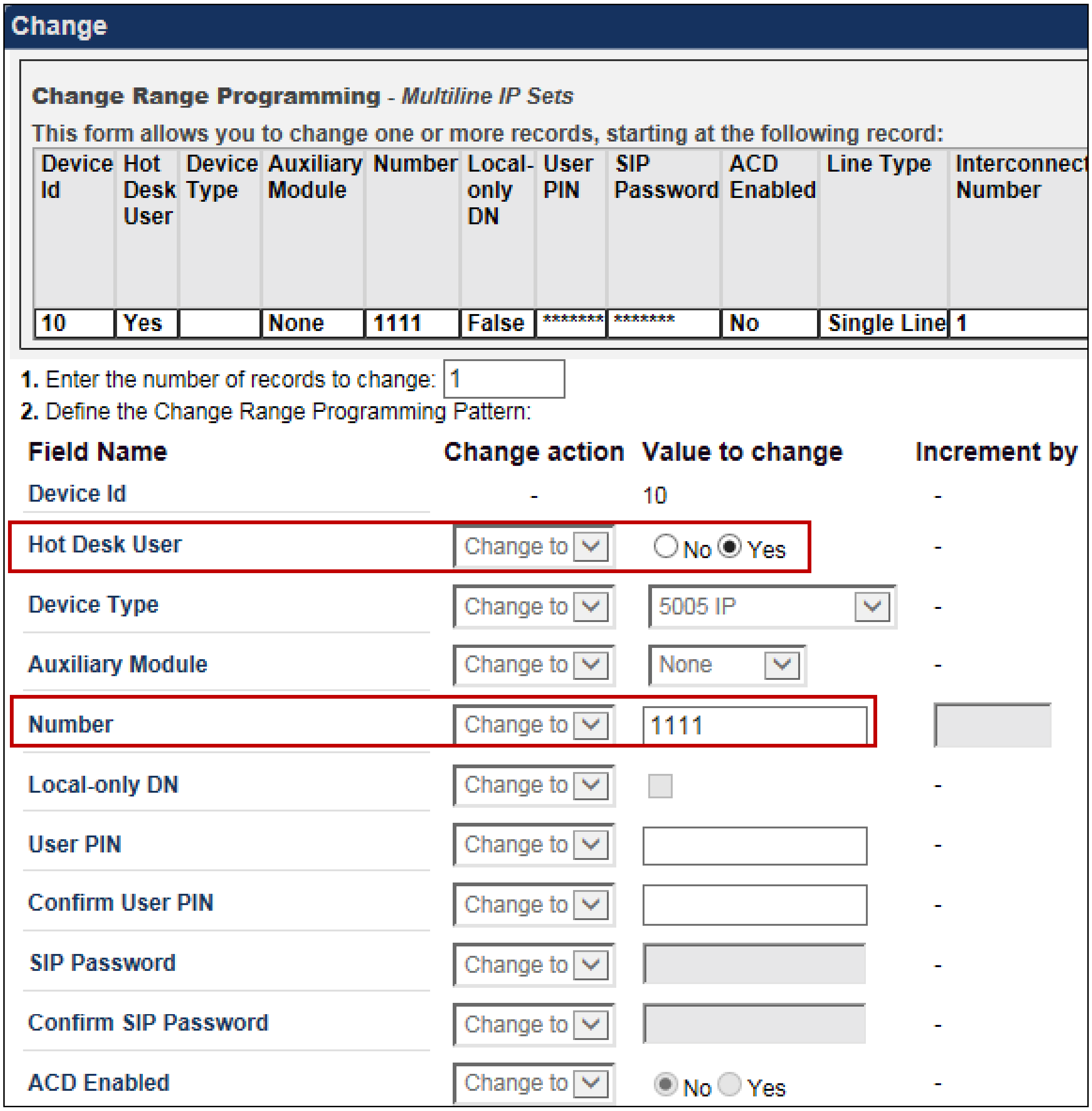

Figure 32: Programming Multiline IP Sets

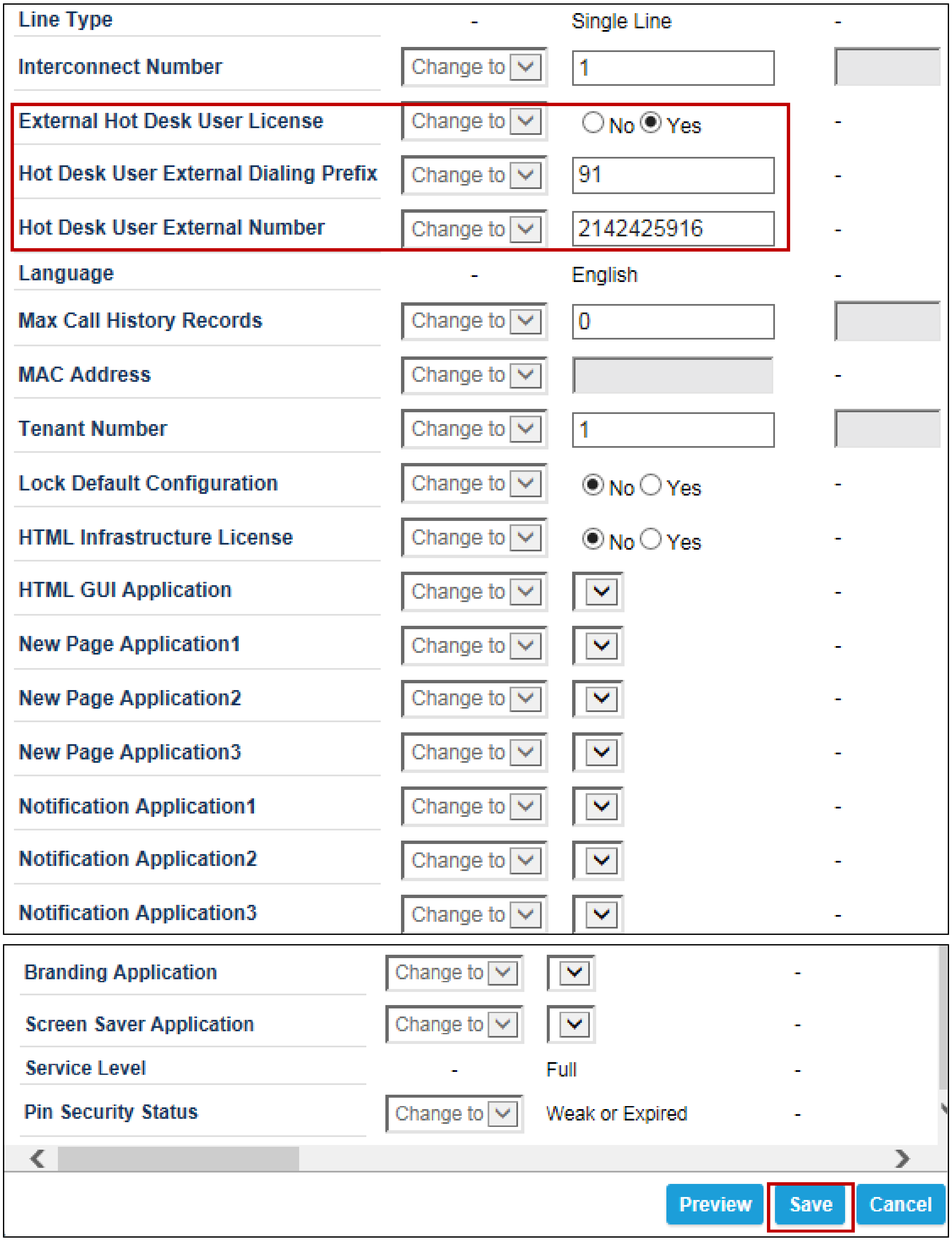

Figure 33: Programming Multiline IP Sets

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

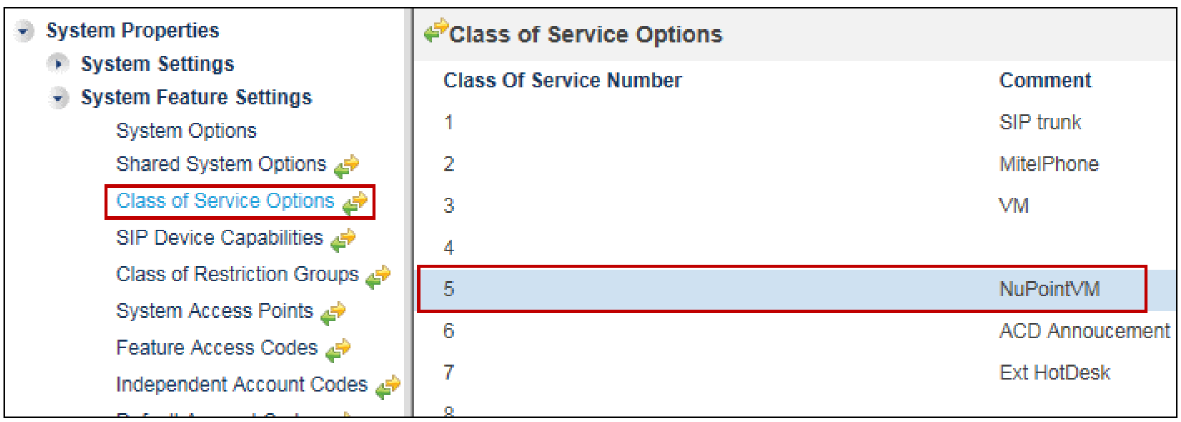

Figure 35: Class of Service Option for NuPoint Voice Ports

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.

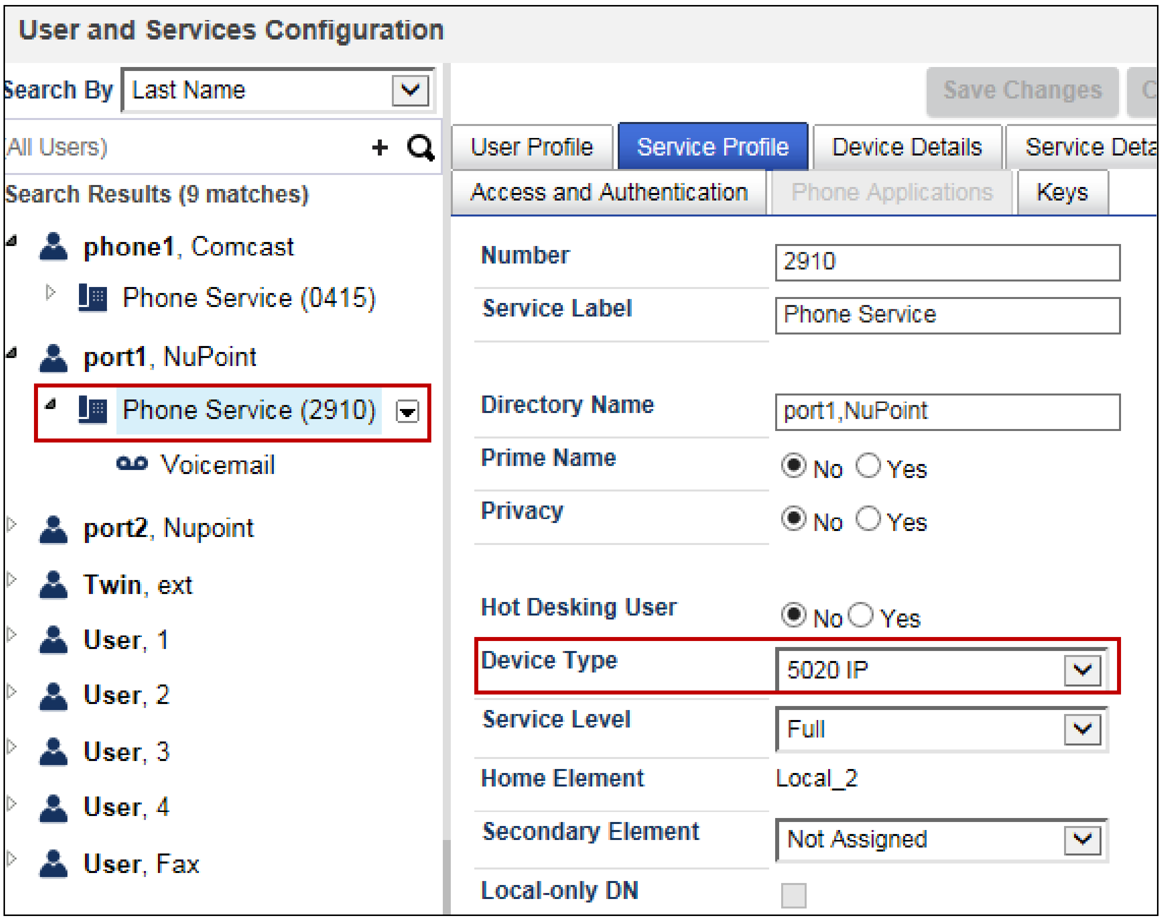

Figure 36: IP Endpoints Configuration

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.

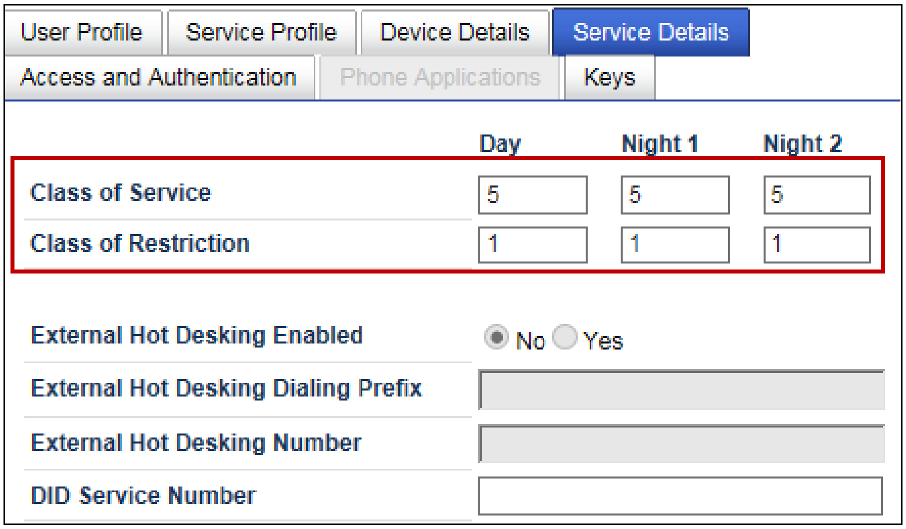

Figure 37: IP Endpoints Class of Service

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.

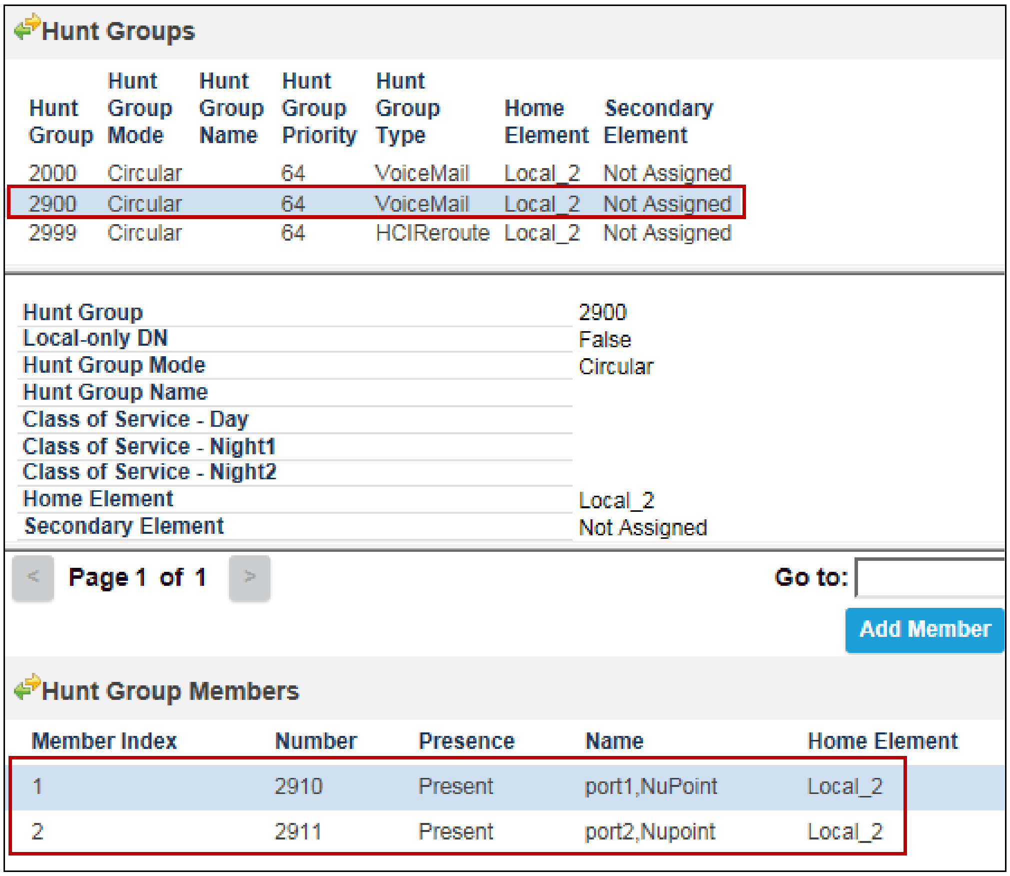

Figure 38: Voicemail Hunt Group Configuration

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

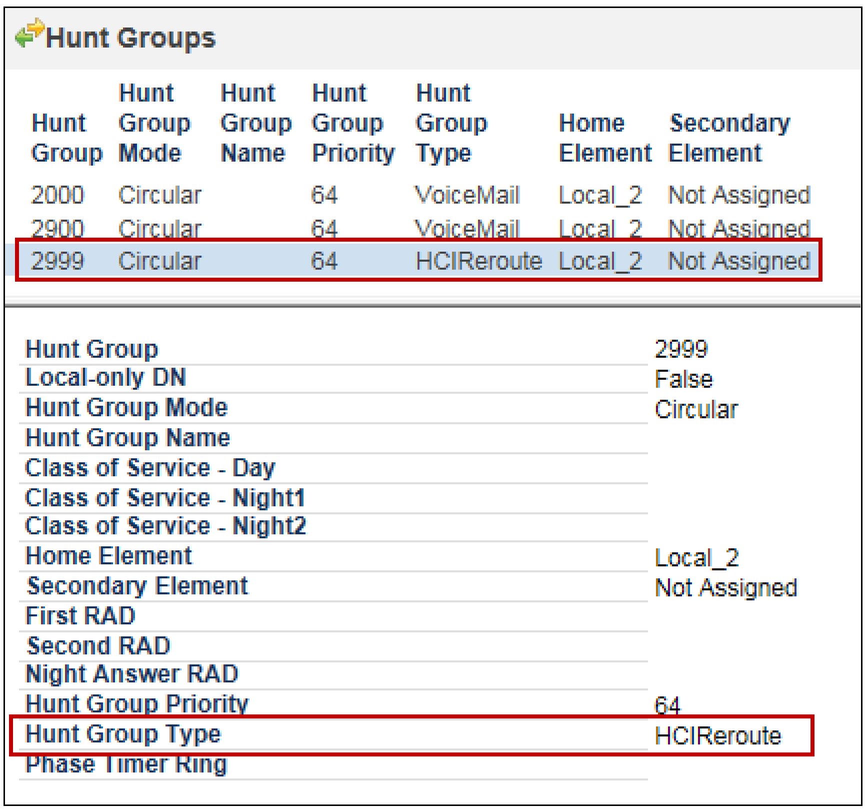

Figure 39: HCIReroute Hunt Group

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

Network Elements

Navigation: From Server Manager, Applications > Users and Services

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

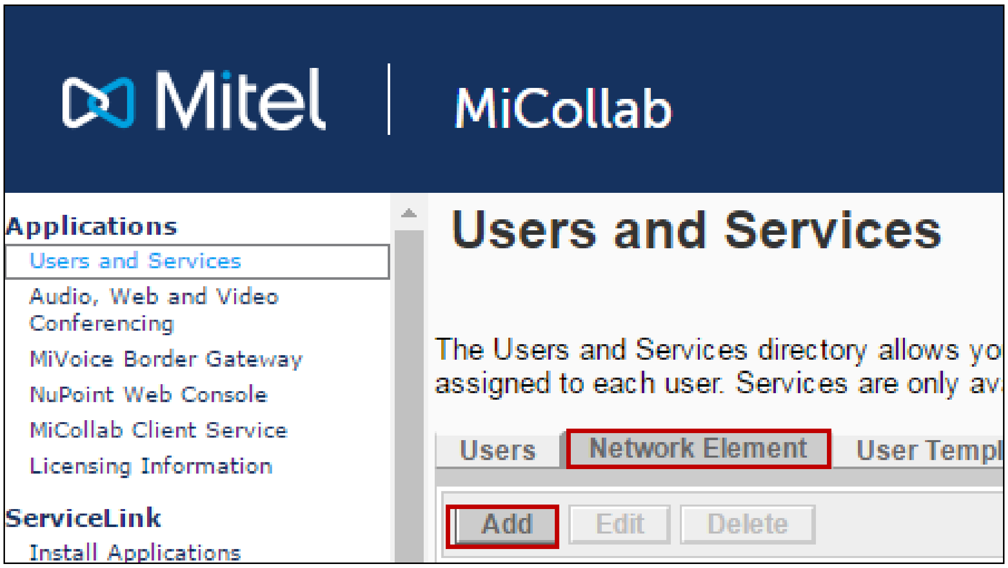

Figure 42: Add Network Element

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**

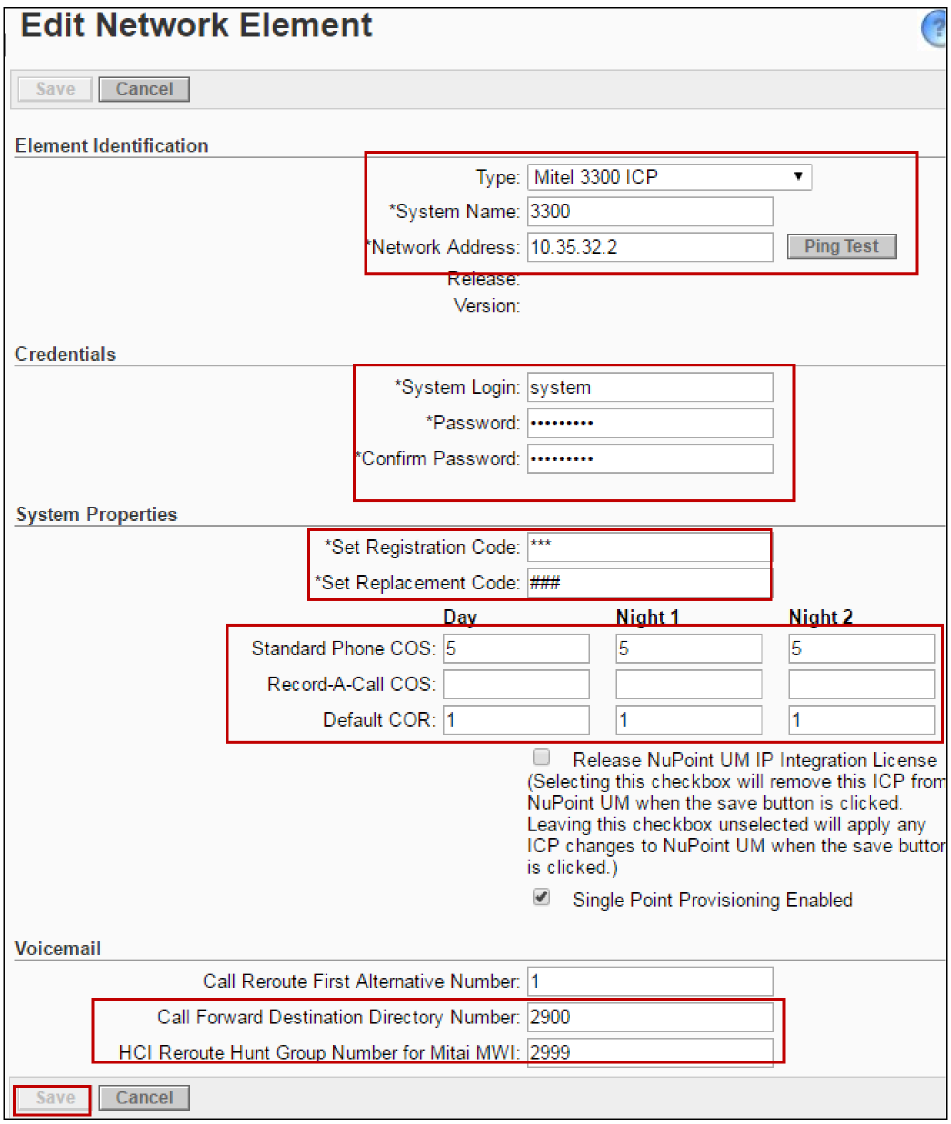

Figure 43: Network Element – Cont.

Voicemail Line Group

1. Click **NuPoint Web Console**

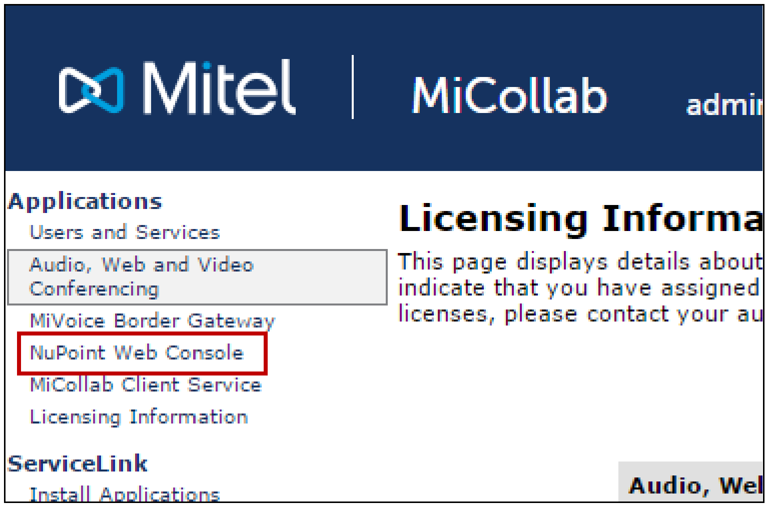

Figure 44: Voicemail Line Group Configuration

Navigation: Offline Configuration > Line Groups

2. Click **Add**

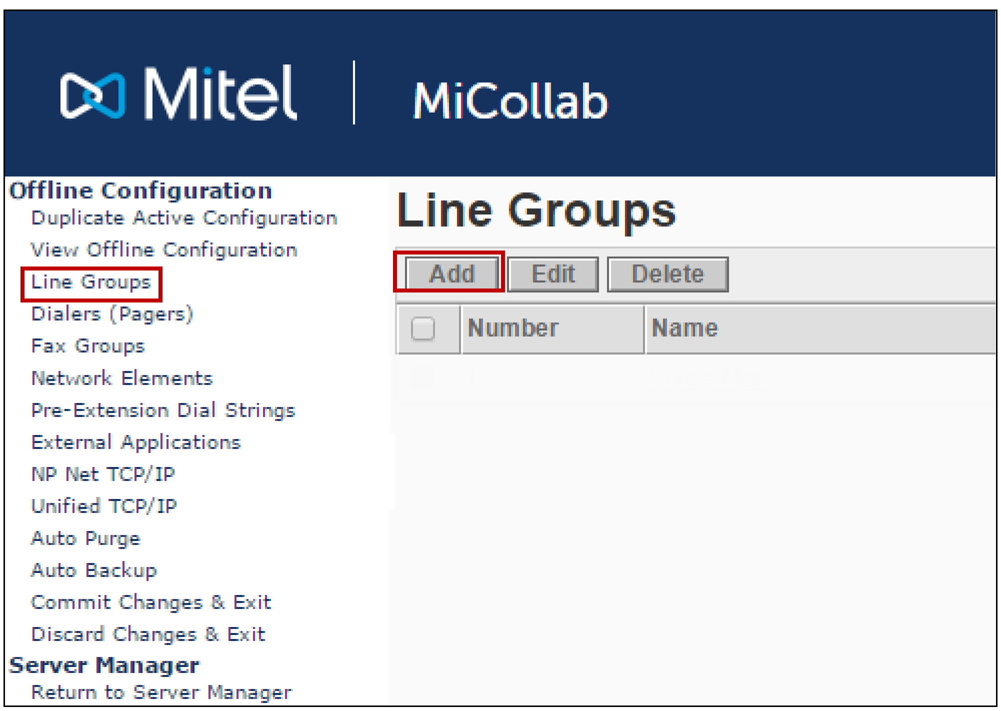

Figure 45: Voicemail Line Group Configuration – Cont.

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**

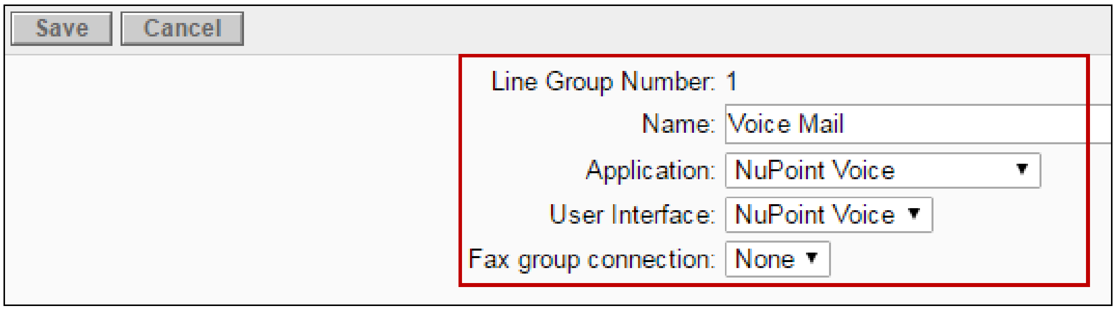

Figure 46: Adding Line Group

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**

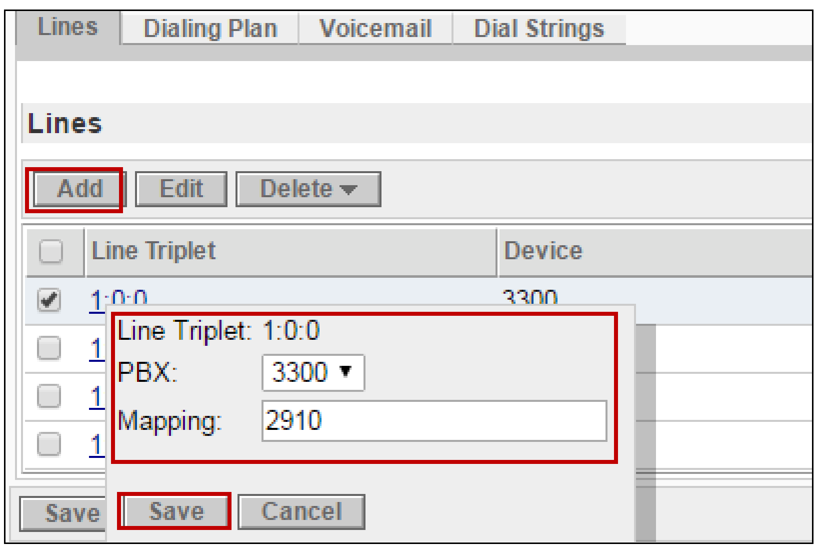

Figure 47: Adding Lines

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**

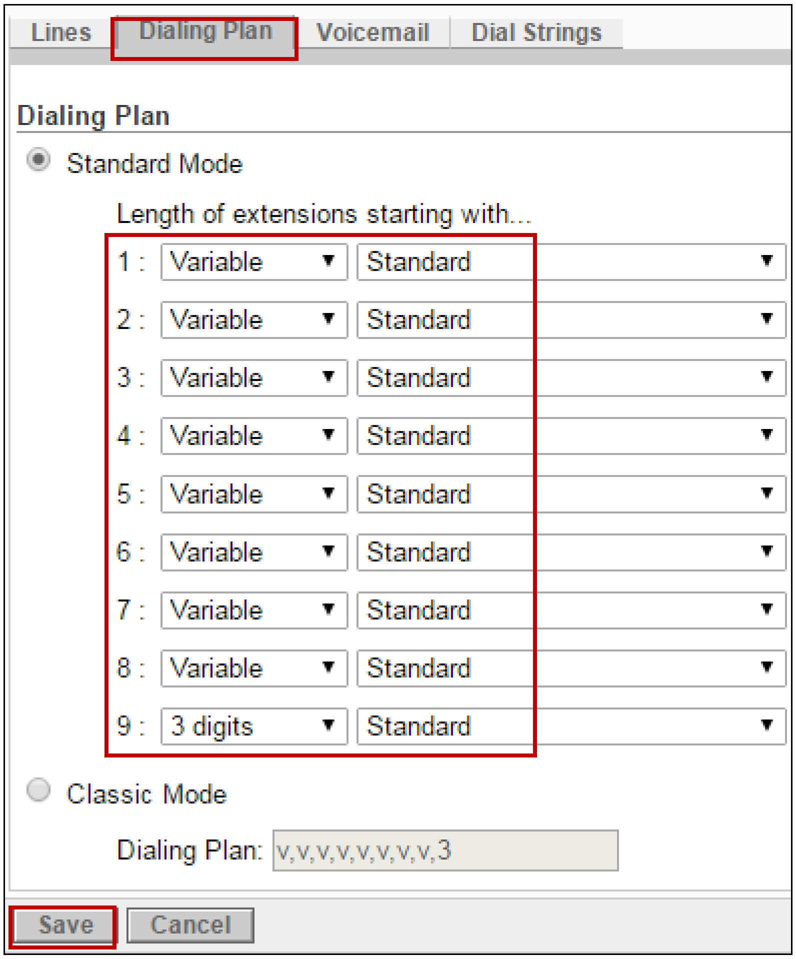

Figure 48: Adding Dial Plan

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

2. Click **Commit**

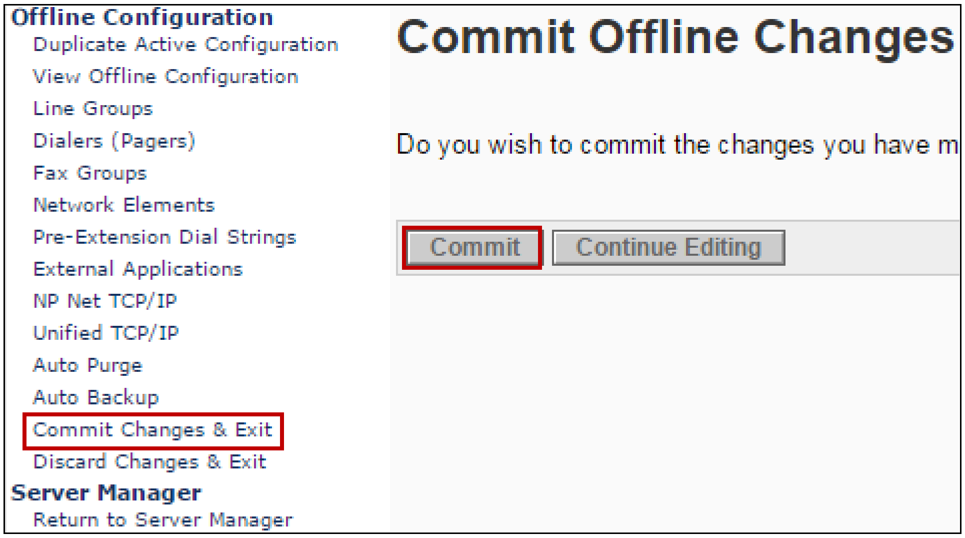

Figure 49: Committing Offline Changes

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**

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**

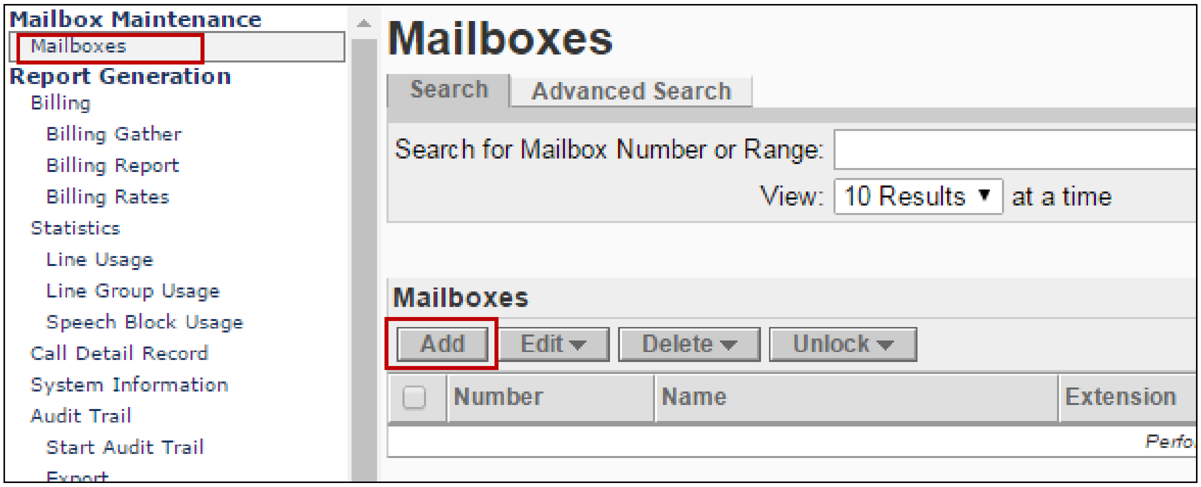

Figure 50: Add Mailbox

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**.

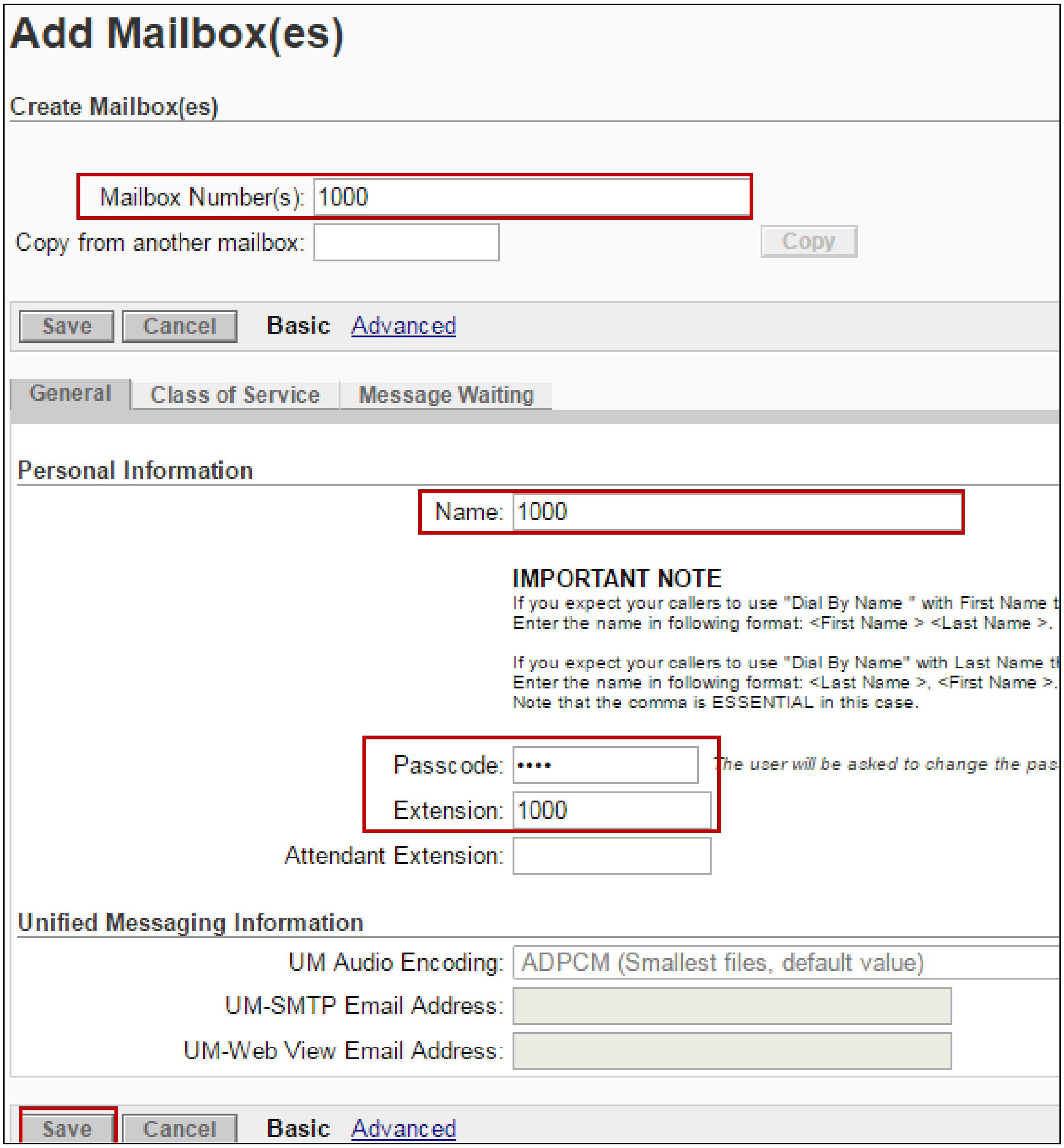

Figure 51: Add Mailbox – Cont.

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

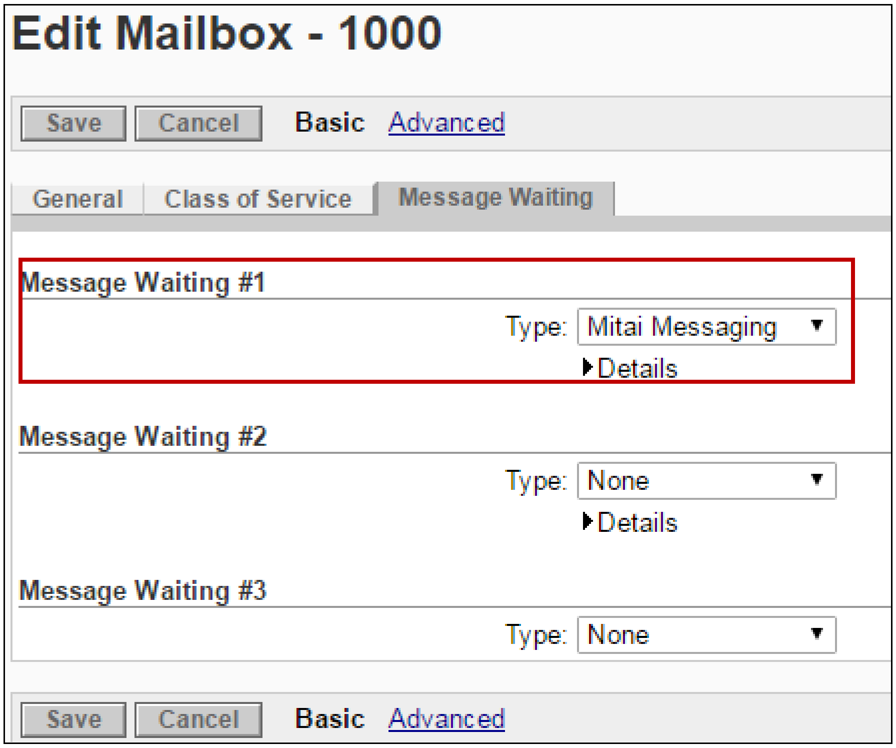

Figure 52: Add Mailbox – Cont.

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**

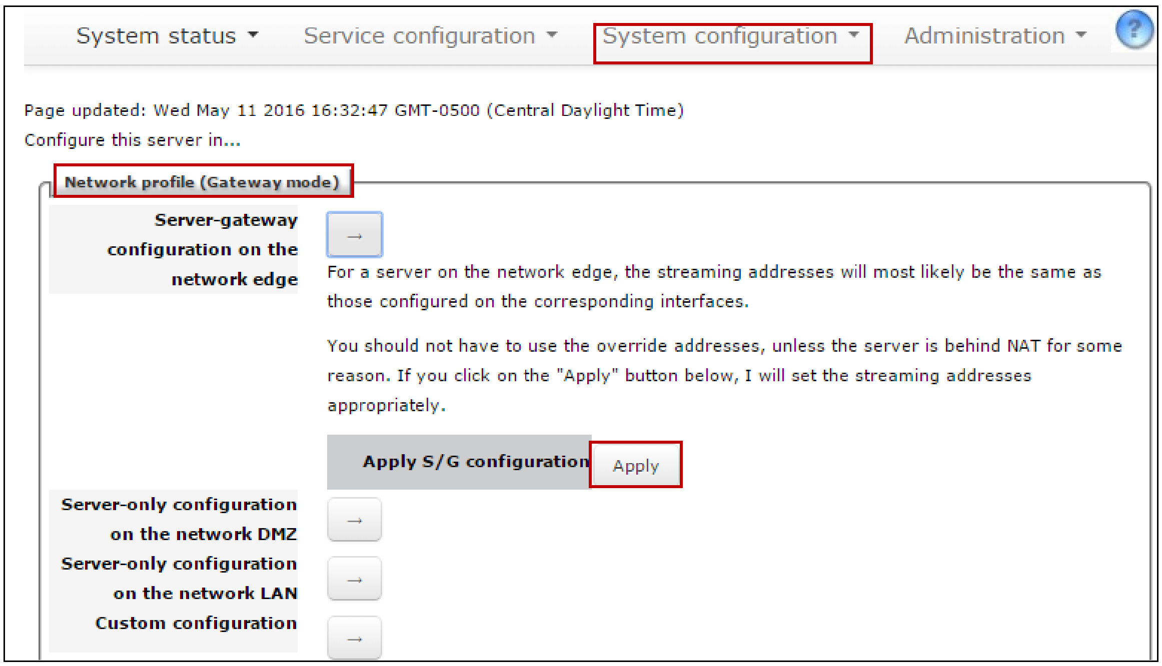

Figure 53: Network Profiles

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

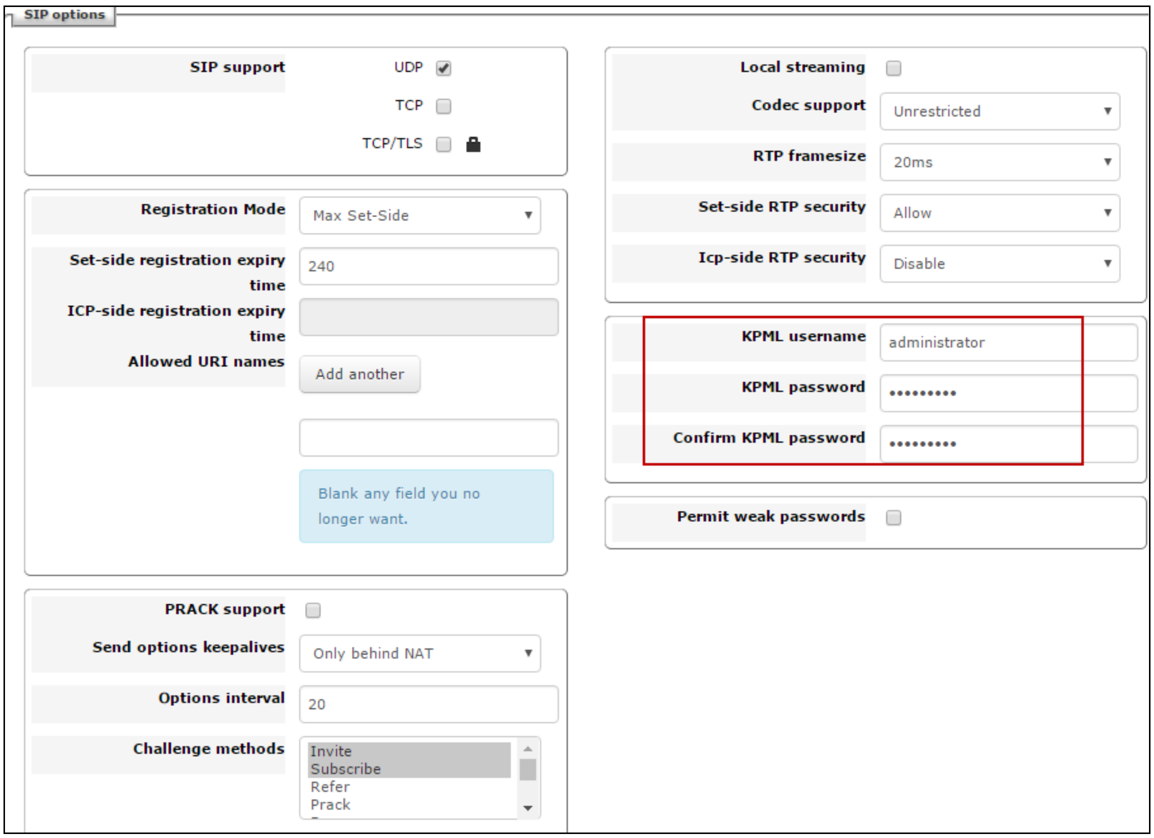

Figure 54: MBG Settings

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

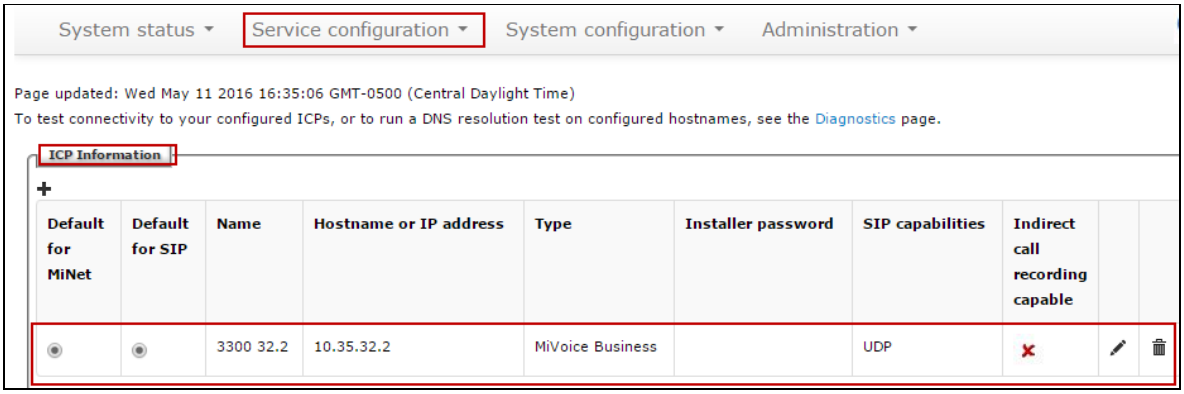

Figure 55: MIVOICE BORDER GATEWAY Configuration

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

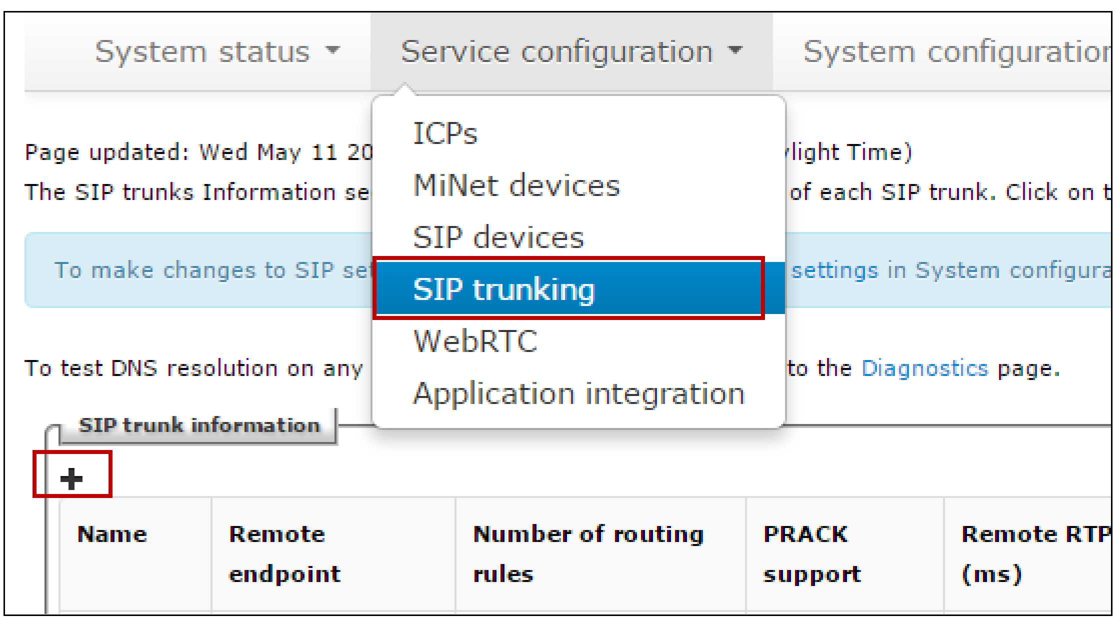

Figure 56: SIP Trunking Configuration

Enter the SIP trunk details as follows:

1. Set **Name**: IntelePeer 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**

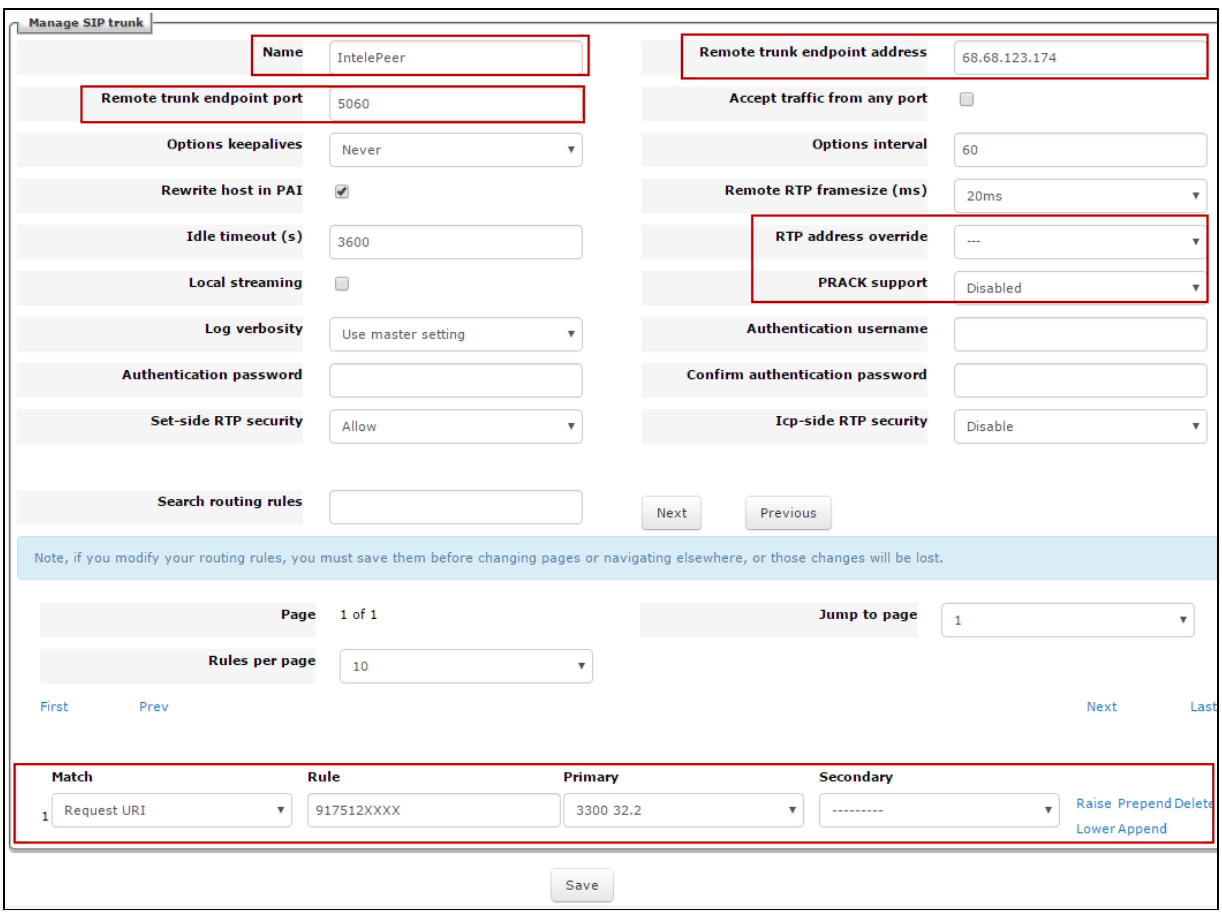

Figure 57: SIP Trunk Configuration Settings

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