Mitel Technical Configuration Notes- HO2849

October 24, 2018

Configure MiVoice Business 8.0 SP3 PR1 with MBG for use with First Communications SIP Trunking

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVoice Business to connect to First Communications SIP Trunking.

Environment: MiVoice Business (Platform: MXe-III) with Software load 14.0.3.22, MiVoice Border Gateway (Virtual) with Version 10.0.2.14

NOTICE

TRADEMARKS

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

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVoice Business to connect to First Communications 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	05-September-2018	Initial Interop with Mitel MiVoice Business Release 8.0 SP3 PR1 Software Load 14.0.3.22 and First Communications SIP Trunking

Interop Status

The Interop of First Communications 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 First Communications SIP Trunking achieved is:

COMPATIBLE	The most common certification which means First Communications 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 First Communications SIP Trunking and the MiVoice Business.

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 8.0 SP3 PR1 Software Load 14.0.3.22
Mitel	MBG – Teleworker	v10.0.2.14
Mitel	Minet Sets: 5312, 5320, 5330	Minet 5312 (06.05.00.11) Minet 5320 (06.05.00.11) Minet 5330 (06.03.03.08)
Service Provider	First Communications	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 First Communications and their PSTN gateway, call holding, transferring, conferencing, busy calls, DTMF RFC 2833, In-band, 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 and DTMF detection.	☑
Packetization	Forcing the Mitel MiVoice Business to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	⚠
Personal Ring Groups	Receiving calls through First Communications and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑
External Hot Desking	Receiving calls through First Communications and their PSTN gateway to PRG with EHDU. Including moving calls to/from the prime member of the PRG with the EHDU. Also placing calls from the EHDU and using mid call features with EHDU.	☑
Teleworker	Making and receiving a call First Communications and their PSTN gateway to and from Teleworker extensions.	☑
Fax	T.38 and G711 Fax 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 First Communications SIP Trunking is connected to the MiVoice Business.

Features	Problem Description
Codec	First Communications does not support G722, G722.1 and G721
Registration & Authentication	First Communications does not support Register & digest based authentication
Packetization	First Communications does not support 30ms, 40ms, 50ms, 60ms Packetization rate
Video	Not supported by First Communications Network.
Fax	First Communications does not support Multiple M-Lines for Fax
TLS – SRTP-Basic Calls	MiVoice Border Gateway does not support TLS/SRTP with the current release.

Network Topology

This diagram shows how the testing network is configured for reference

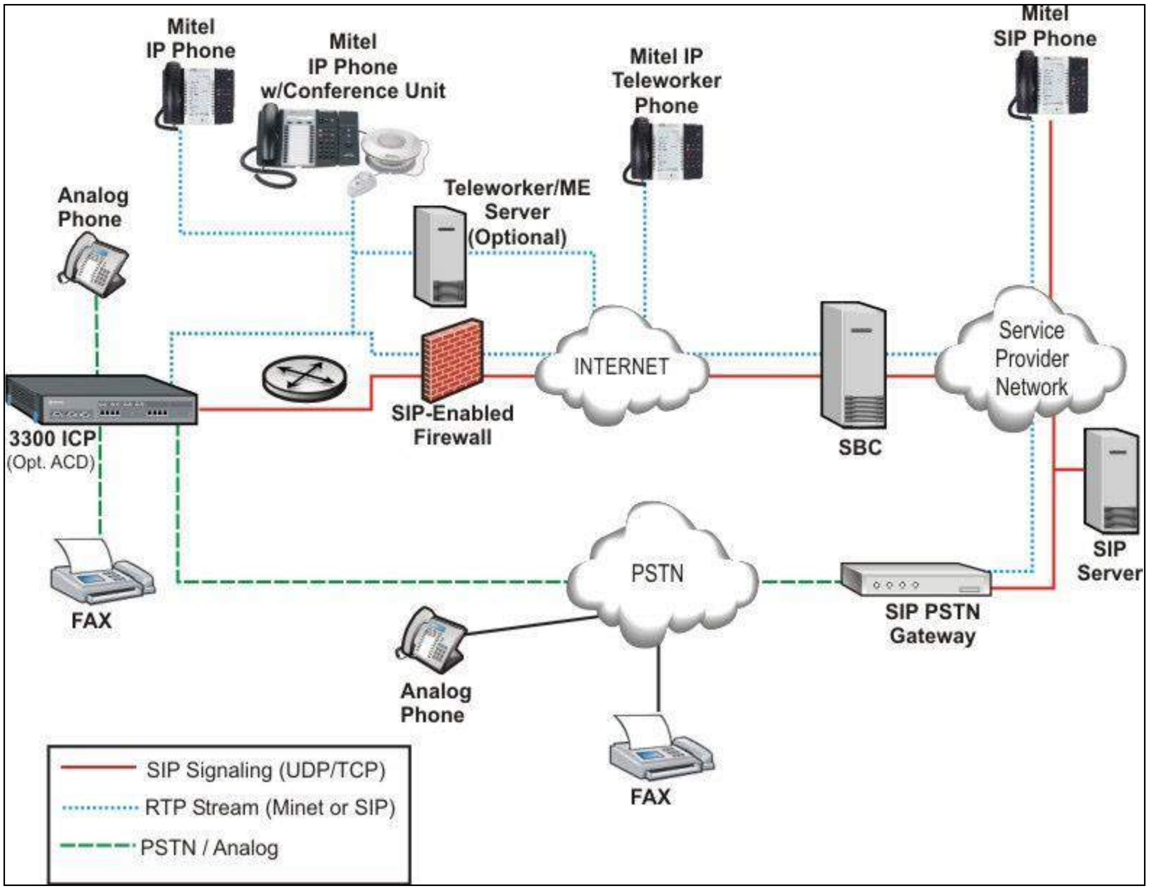


Figure 1: Network Topology

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 First Communications SIP Trunking MiVoice Business programming 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 First Communications 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 voice over IP. As a guide, the Ethernet bandwidth is approx. 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms Packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx. 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the 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

Ensure that the MiVoice Business is equipped with enough SIP Trunking licenses for the connection to First Communications 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.

License and Option Selection

Application Record ID 26682859

System Type	License Sharing	Hardware Identifier
Enterprise	No	0000003a1a4f

Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Local Limits: Licenses Allowed	Local Limits: Can be Over Allocated
Users						
IP Users	16	16	0	16	Unrestricted	Yes
External Hot Desk Users	3	7	3	10	Unrestricted	Yes
ACD Active Agents	2	10	0	10	Unrestricted	No
HTML Applications	0	0	20	0	Unrestricted	Yes
Single Line Users	0	16	0	16	Unrestricted	Yes
MiVoice Business Console Active Operators	0	0	20	0	Unrestricted	No
Multi-device Users	3	5	0	5	Unrestricted	Yes
Multi-device Suites	0	0	5	5	0	No
Messaging						
Embedded Voice Mail	2	16	0	16	Unrestricted	Yes
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	Yes
Trunking / Networking						
Digital Links	0	1	0	1	Unrestricted	Yes
Compression		8	0	8	Unrestricted	Yes
FAX Over IP (T.38)		4	0	4	Unrestricted	Yes
SIP Trunks	5	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				

Figure 2 – License and Option Selection

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 3 - License and Option Selection – Contd.

Class of Service Assignment

The Class of Service Options Assignment form is used to create or edit a Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced in the Trunk Service Assignment form for SIP trunks.

Many different options may be required for your site deployment, but eensure that “Public Network Access via DPNSS” Class of Service Option is configured for all devices that make outgoing calls through the SIP trunks in the MiVoice Business.

- Public Network Access via DPNSS set to Yes
- Campon Tone Security/FAX Machine set to Yes

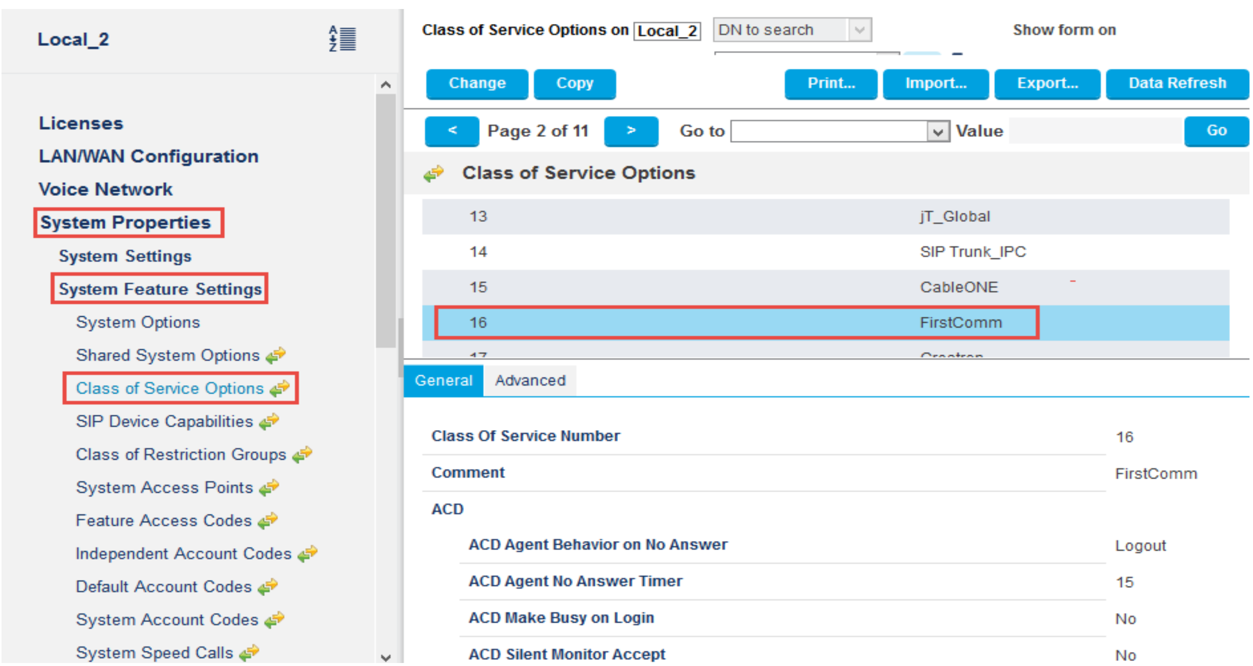


Figure 4 – Class of Service

Class of Service for SIP Trunk

General | Advanced

Class Of Service Number	16
Comment	FirstComm
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	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

Figure 5: Class of Service (General) – Contd.

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
Follow Reroute on Disabled Forwarding	No
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No
Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes

Figure 6: Class of Service (General) – Contd.

Call Hold	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	Yes
Call Hold Remote Retrieve	Yes
Call Hold Timer	
Local Music On Hold source	Yes
Music on Hold on Transfer	No
Use Called Party Call Hold Timer	No
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	Yes
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Pickup Display	No
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	Yes

Figure 7: Class of Service (General) – Contd.

Campon	
Auto Campon Timer	
Campon Recall Timer	0
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
Display DNIS/Called Number Before Digit Modification	No
Display DNIS on Key Label	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

Figure 8: Class of Service (General) – Contd.

Originator's Display Update In Call Forwarding/Rerouting	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	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 Auto Logout Timer	0
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

Figure 9: Class of Service (General) – Contd.

Miscellaneous	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Enbloc Dialing - Enabled	No
Force Device Busy If Any Line In Use	No
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Integrated DECT High Power - Enabled	Yes
Integrated DECT Wideband - Enabled	Yes
Long Key Press Timer	0
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
Pager Access All Zones	Yes
Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No

Figure 10: Class of Service (General) – Contd.

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	Yes
DASS II OLI/TLI Provided	Yes
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
R2 Call Progress Tone	Yes
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes

Figure 11: Class of Service (General) – Contd.

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 12: Class of Service (General) – Contd.

Figure 13: Class of Service (Advanced) – Contd.

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
Miscellaneous	
Auto Answer Allowed	Yes
Auto Release on Key Select	No
Brokers Call	No
Called Party Features Override	No

Figure 14: Class of Service (Advanced) – Contd.

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

Figure 15: Class of Service (Advanced) – Contd.

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 16: Class of Service (Advanced) – Contd.

Class of Service for Phones

General | Advanced

Class Of Service Number	11
Comment	MitelPhone_FC
ACD	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
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

Figure 17: Class of Service (General)

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

Figure 18: Class of Service (General) – Contd.

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
Follow Reroute on Disabled Forwarding	Yes
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No
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	Yes
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	Yes
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Pickup Display	No

Figure 19: Class of Service (General) – Contd.

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	
Campon Recall Timer	0
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 20: Class of Service (General) – Contd.

Display DNIS/Called Number Before Digit Modification	Yes
Display DNIS on Key Label	No
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	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	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	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 Auto Logout Timer	0

Figure 21: Class of Service (General) – Contd.

Hot Desk External User - Allow Mid-Call Features	No
Hot Desk External User - Answer Confirmation	No
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
Enbloc Dialing - Enabled	No
Force Device Busy If Any Line In Use	Yes
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Integrated DECT High Power - Enabled	Yes
Integrated DECT Wideband - Enabled	Yes
Long Key Press Timer	0
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

Figure 22: Class of Service (General) – Contd.

Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes
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

Figure 23: Class of Service (General) – Contd.

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	Yes
ONS VMail-Delay Dial Tone Timer	5

Figure 24: Class of Service (General) – Contd.

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
Call Screening	
BLF Screening Allow	No
BLF Screening Accept	No
Conference	
Conference Call	Yes

Figure 25: Class of Service (Advanced) – Contd.

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	

Figure 26: Class of Service (Advanced) – Contd.

Message Waiting Ringing Stop Time Hour	
Message Waiting Ringing Stop Time Minute	
Multiline Set Voice Mail Callback Message Erasure Allowed	Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate	Yes
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

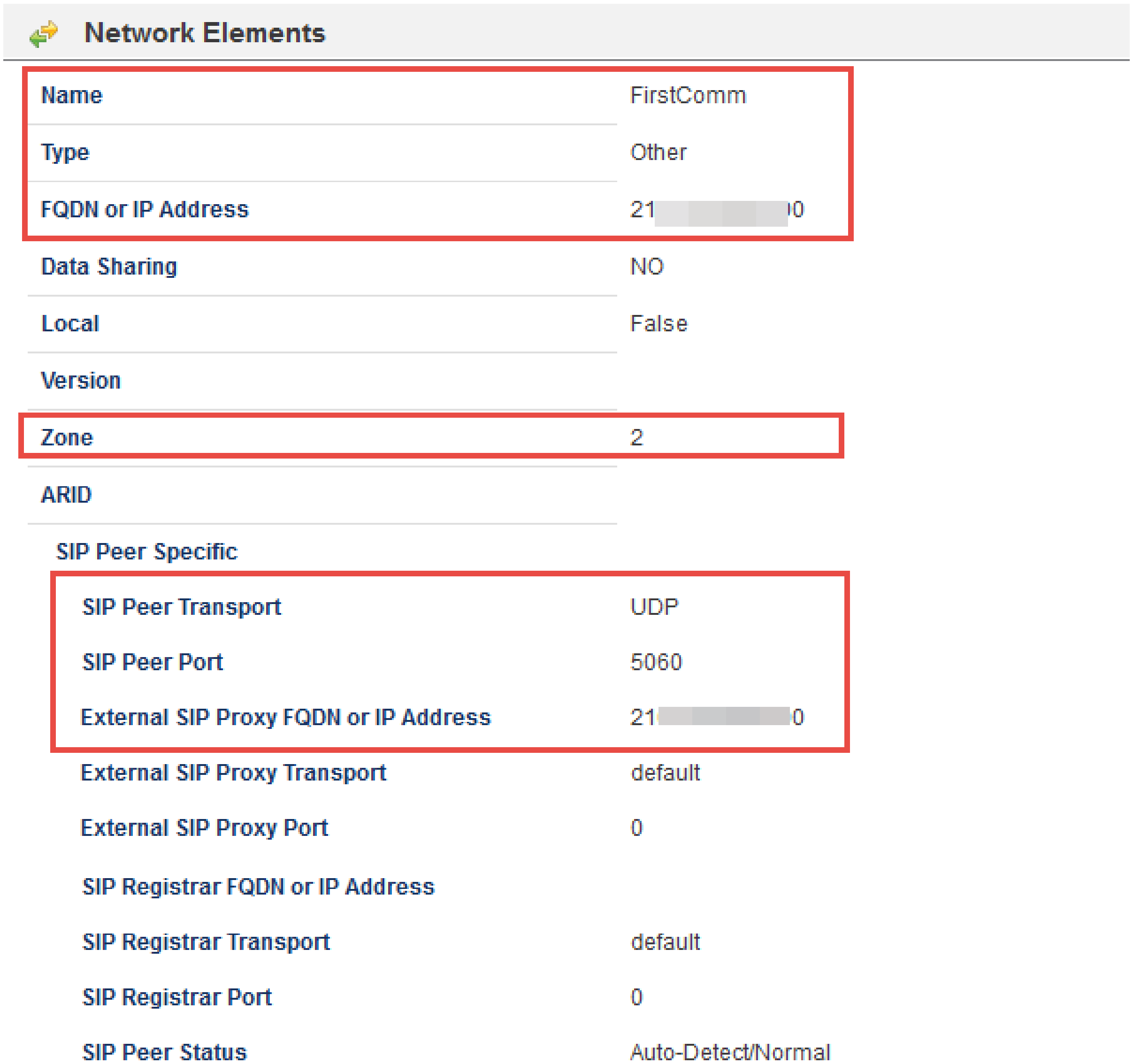
Figure 27: Class of Service (Advanced) – Contd.

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	Yes
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 28: Class of Service (Advanced) – Contd.

Network Element Assignment

Create a network element for First Communications SIP Trunking. In this example, the soft switch is reachable by an IP Address and is defined as “First Communications” 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 transport to UDP and port to 5060.



Network Elements

Name	FirstComm
Type	Other
FQDN or IP Address	21[illegible]0
Data Sharing	NO
Local	False
Version	
Zone	2
ARID	
SIP Peer Specific	
SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	21[illegible]0
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

Figure 29 – Network Element Assignment

Network Element Assignment (Proxy)

In addition, depending on your configuration, a Proxy may need to be configured to route SIP data to the service provider. If a Proxy server is installed in the network, the MiVoice Business should be configured with the Proxy as a network element then referencing this proxy in the SIP Peer profile assignment (later in this document).

Name	FC_MBG
Type	Outbound Proxy
FQDN or IP Address	10.64.4.9
Data Sharing	NO
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	UDP
Outbound Proxy Port	5060

Figure 30 – 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 6 which will be used to direct incoming calls to an answer point in the Mitel MiVoice Business.

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.

The example below shows configuration for incoming DID calls. The Mitel MiVoice Business will not absorb any digits of the DID number from First Communications leaving 10 digits for the MiVoice Business to translate and ring the remaining 4 digit extension. For example, First Communications delivers 224-717-5112 through the SIP trunk to the MiVoice Business. The MiVoice Business will not absorb any digits and leaving the MiVoice Business to ring extension 5112. Extension 5112 must be programmed as a valid dial able number in the MiVoice

Business. Please refer to the Mitel MiVoice Business System Administration documentation for further programming information.

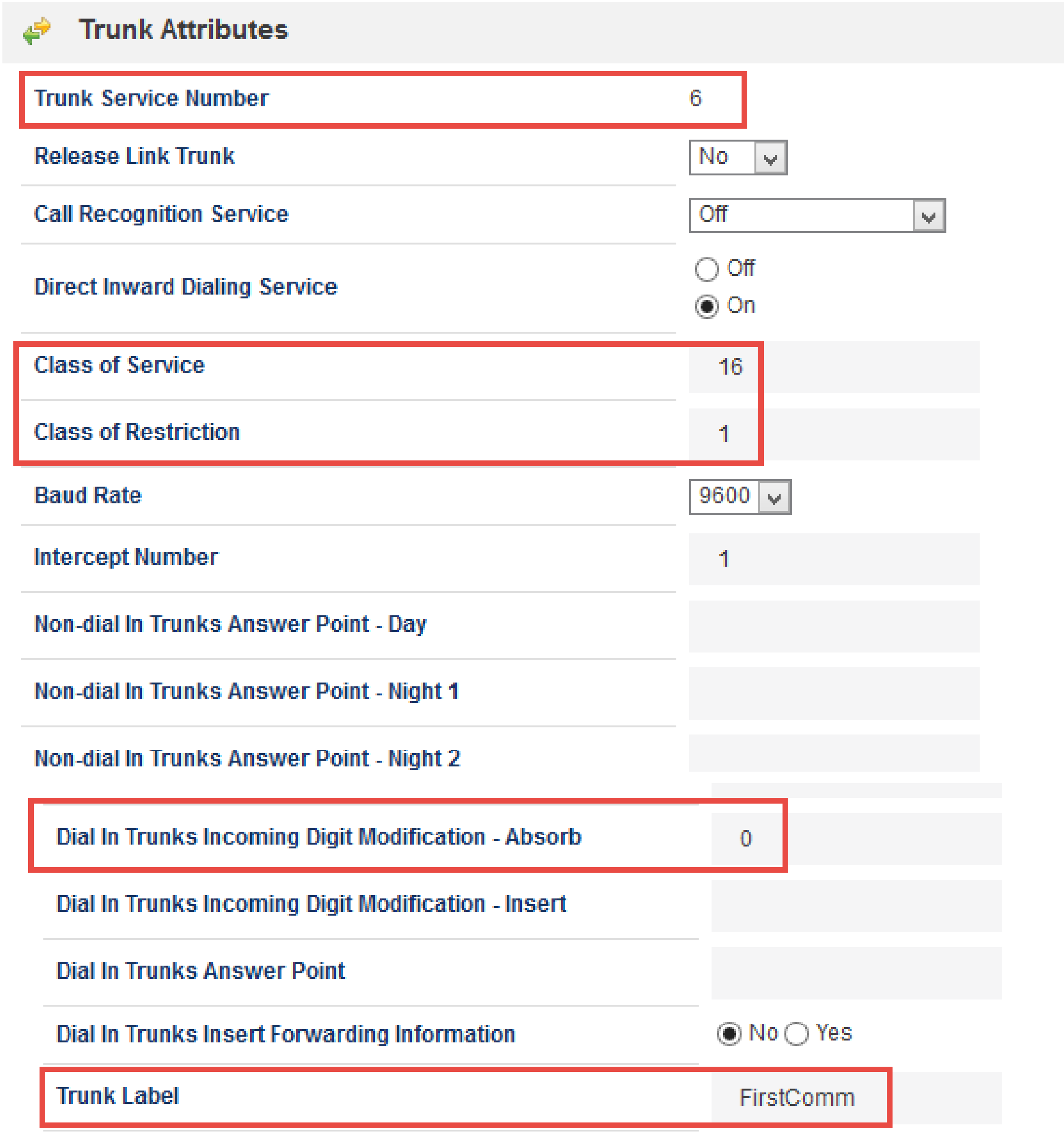


Figure 31 – Trunk Attributes

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:

Network Element: The selected SIP Peer Profile needs to be associated with previously created “First Communications” Network Element.

Registration User Name: The Mitel MiVoice Business does not support Bulk Registration; therefore trunks will have to be registered individually. Enter the DIDs assigned by First Communications. Enter one or more numbers. The field has a maximum of 60 characters. The maximum number of digits per number is 26. You can enter a mix of ranges and single numbers (for example, "2247175110-2247175113, 2247175114"). Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash.

Address Type: Select IP address.

Outbound Proxy Server: Select the Network Element previously configured for the Outbound Proxy Server.

Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided by First Communication**. Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the DID ranges for CPN Substitution form (see DID Ranges for CPN Substitution). Then select the appropriate numbers in the Outgoing DID Ranges in this form (SIP Peer Profile).

Trunk Service Assignment: Enter the trunk service assignment previously configured.

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).

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).

Maximum Simultaneous Calls: This entry should be configured to maximum number of SIP trunks provided by First Communications.

NOTE: Ensure the remaining SIP Peer profile policy options are similar the screen capture below.

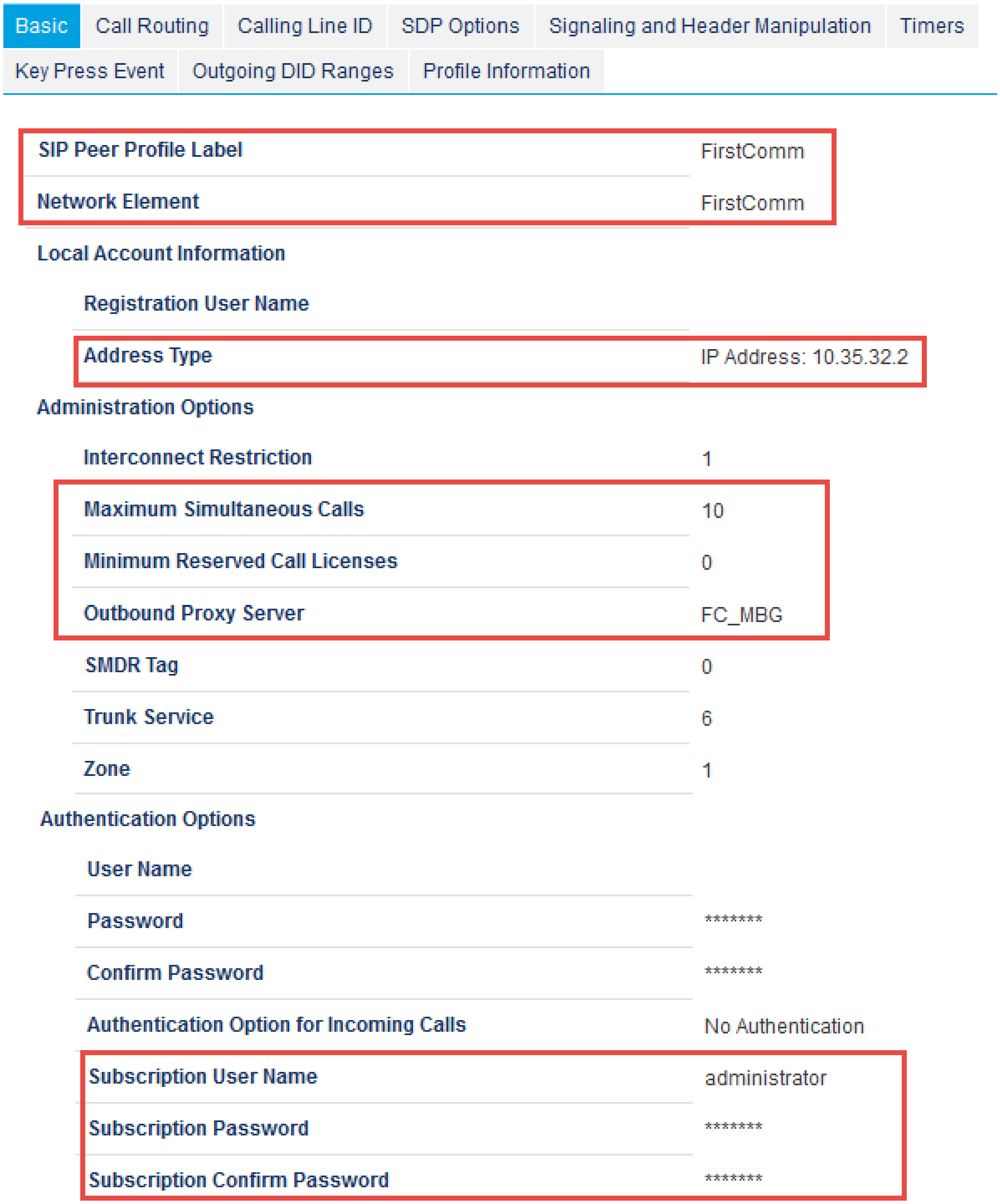


Figure 32 – SIP Peer Profile Assignment- Basic

Basic | **Call Routing** | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Alternate Destination Domain Enabled	No
Alternate Destination Domain FQDN or IP Address	
Enable Special Re-invite Collision Handling	No
Only Allow Outgoing Calls	No
Private SIP Trunk	No
Reject Incoming Anonymous Calls	No
Route Call Using P-Called-Party-ID (if present)	Yes
Route Call Using To Header	No

Figure 33: SIP Peer Profile Assignment- Call Routing

Basic | Call Routing | **Calling Line ID** | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information

Default CPN	
Default CPN Name	
CPN Restriction	No
Override From Header with Default CPN	No
Public Calling Party Number Passthrough	No
Strip PNI	No
Use Diverting Party Number as Calling Party Number	No
Use Original Calling Party Number If Available	No

Figure 34: SIP Peer Profile Assignment- Calling Line ID

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers
Key Press Event | Outgoing DID Ranges | Profile Information

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

Figure 35: SIP Peer Profile Assignment- SDP Options

Figure 36: SIP Peer Profile Assignment- Signaling and Header Manipulation

Use Privacy: none	No
Use P-Asserted Identity Header	Yes
Use P-Asserted Identity for Billing	No
Use P-Call-Leg-ID Header	No
Use P-Early-Media Header	No
Use P-Preferred Identity Header	No
Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No
Use user=phone	No
Use user=phone for Diversion Header	No

Figure 37: SIP Peer Profile Assignment- Signaling and Header Manipulation – Contd.

Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers
Key Press Event | Outgoing DID Ranges | Profile Information

Keep-Alive (OPTIONS) Period	120
Registration Period	3600
Registration Period Refresh (%)	50
Registration Maximum Timeout	90
Session Timer	1800
Session Timer: Local as Refresher	No
Subscription Period	3600
Subscription Period Minimum	300
Subscription Period Refresh (%)	80
Invite Ringing Response Timer	0

Figure 38: SIP Peer Profile Assignment- Timers

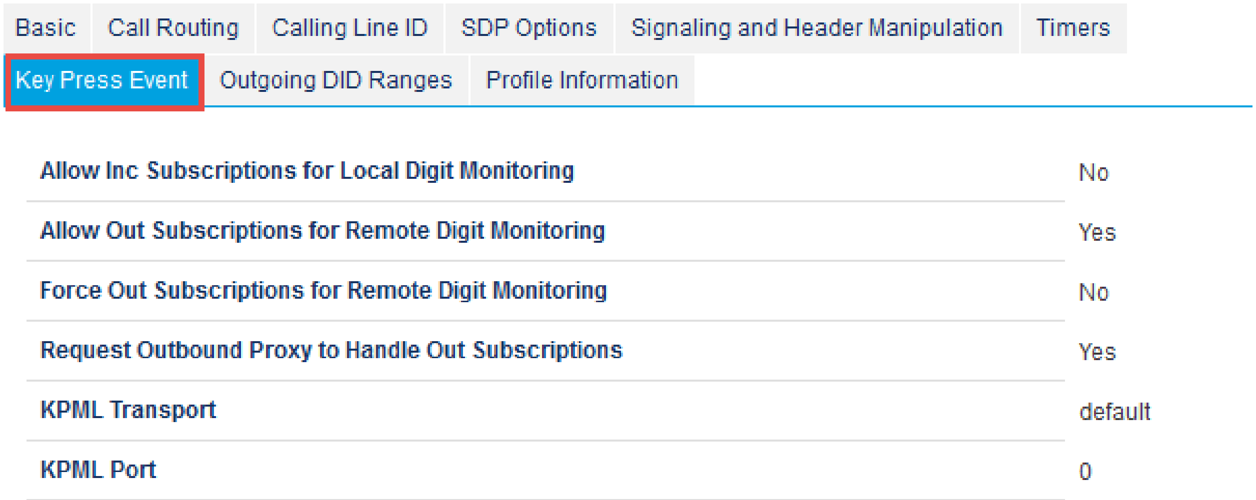


Figure 39: SIP Peer Profile Assignment – Key Press Event

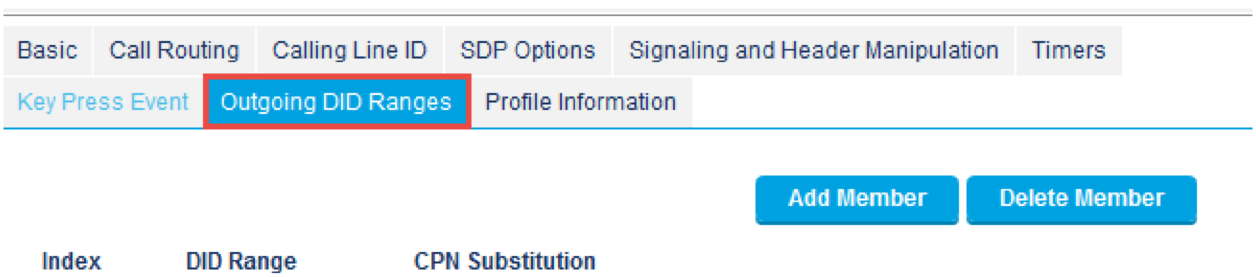


Figure 40: SIP Peer Profile Assignment – Outgoing DID Ranges

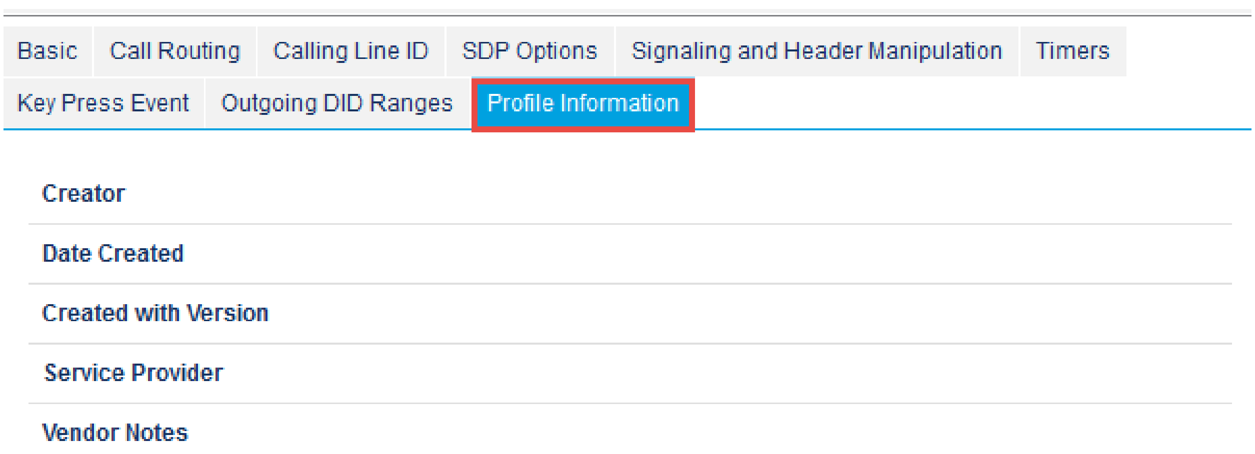


Figure 41: SIP Peer Profile Assignment – Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from First Communications SIP trunk to a particular SIP Peer profile. The configured here settings help matching the incoming DID numbers with the SIP Peer Profile when call is arriving from anonymous caller.
Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, "2247175110 - 2247175114, and 2247175116"). The entire field width is limited to 60 characters.
Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 characters maximum, you can create a new entry for the same profile.
Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier", and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

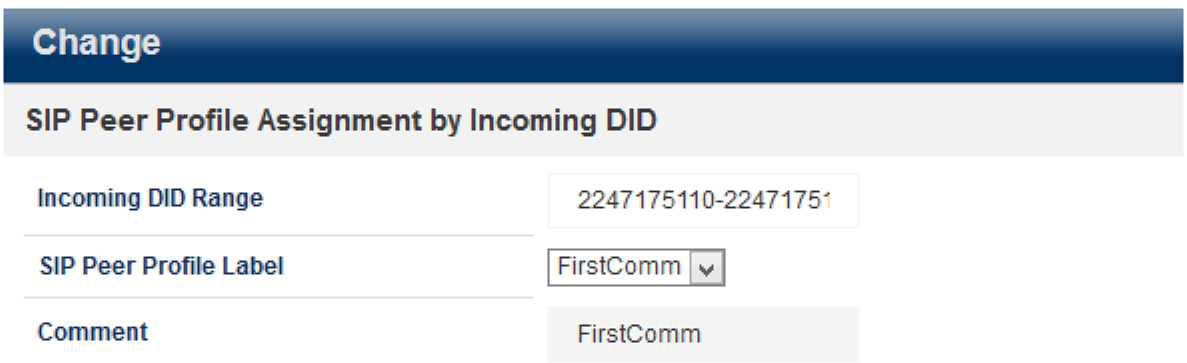


Figure 42: SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to First Communications absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 1 digits (i.e. 8 trunk access code).

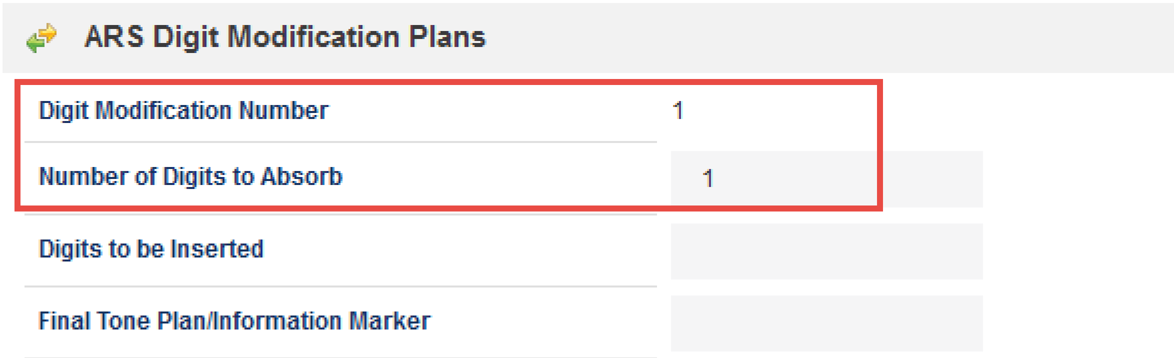


Figure 43: Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to First Communications. In this example, the SIP trunk is assigned to Route Number 5. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

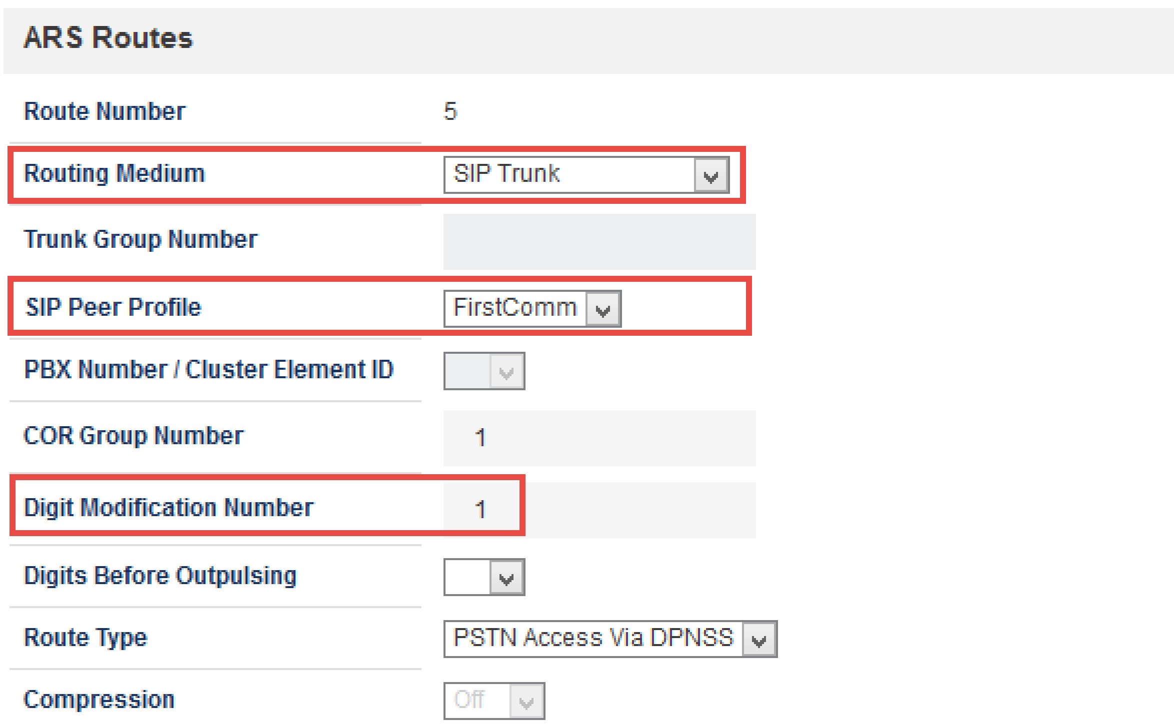


Figure 44: SIP Trunk Route Assignment

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 (8214XXXXXXX), access code 8 will be removed and the call will be routed to First Communications (i.e. Route 5).

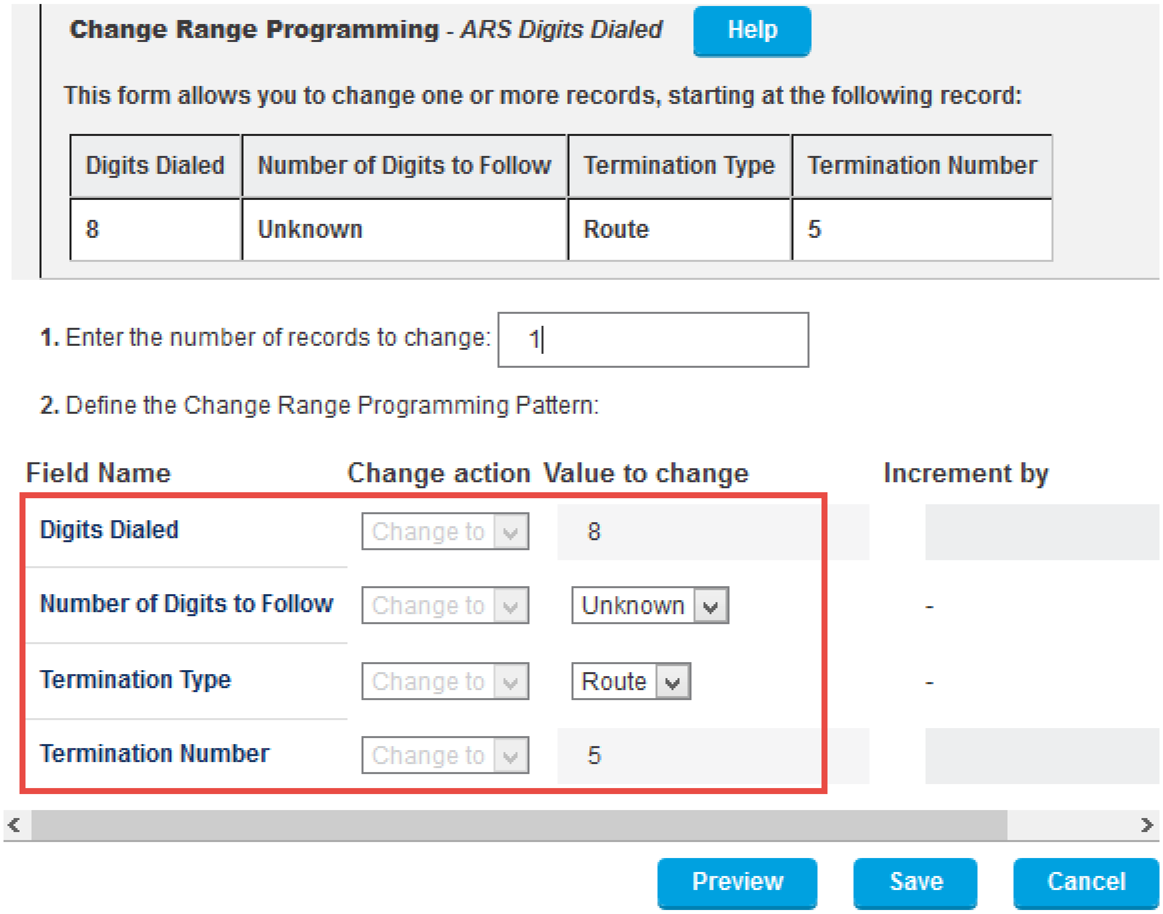


Figure 45: ARS Digit Dialed Assignment

T.38 Fax Configuration

First Communications uses the inter-zone FAX profile. This form allows you to define the settings for FAX communication over the IP network. You can modify the default settings for the:

- **Inter-zone FAX profile**: defines the FAX settings between different zones in the network. There is only one Inter-zone FAX profile; it applies to all inter-zone FAX communication. It defaults to V.29, 7200bps. It defines the settings for FAX Relay (T.38) FAX communication.
- **Intra-zone FAX profile**: defines the FAX settings within each zone in the network.
    - Profile 1 defines the settings for G.711 pass through communication.
    - Profile 2 to 64 define the settings for FAX Relay (T.38) FAX communication.
    - All zones default to G.711 pass through communication (Profile 1).

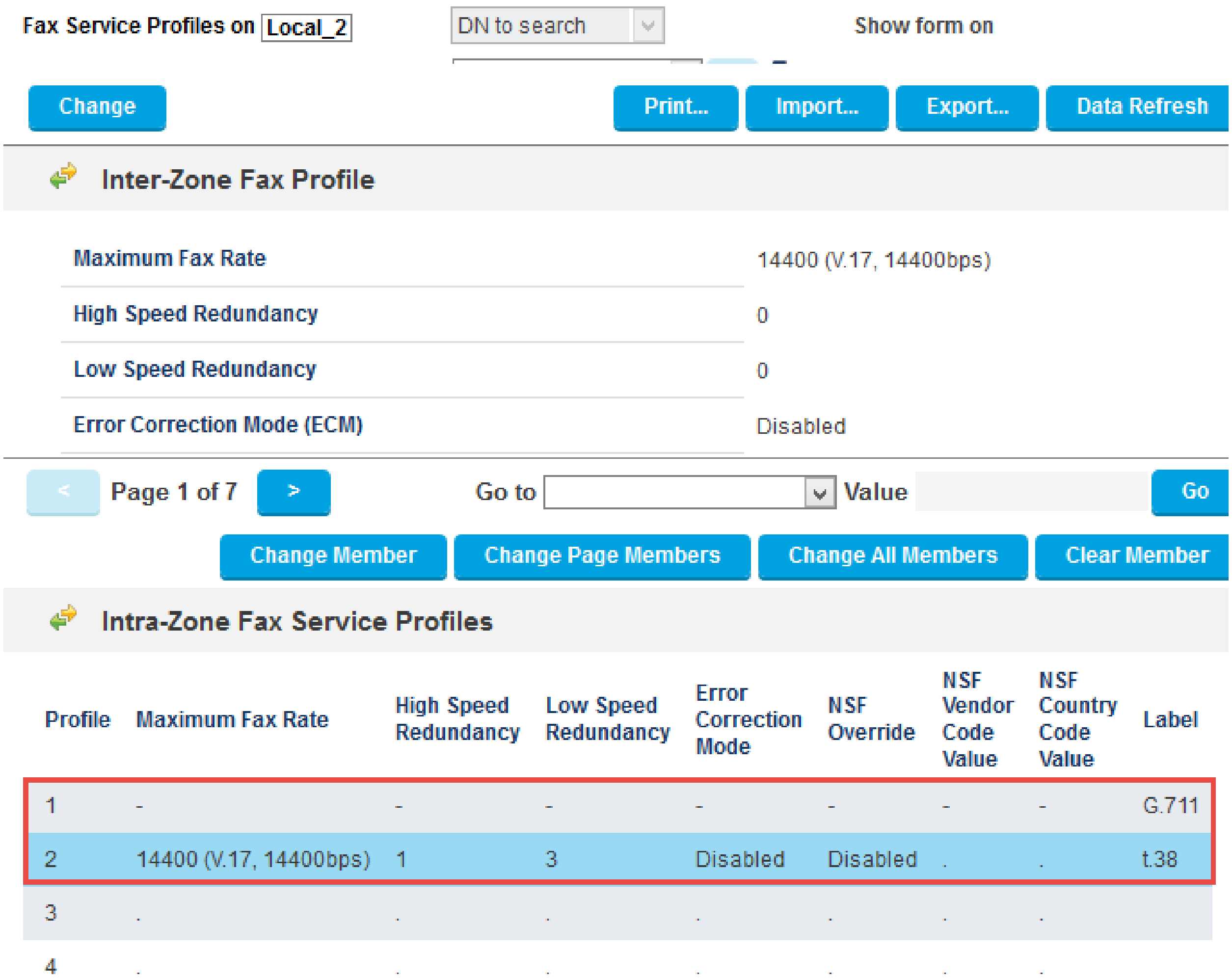


Figure 46: Fax Configuration

Zone Assignment

By default, all zones are set to Intra-zone FAX Profile 1.
Based on your network diagram, assign the Intra-zone FAX Profiles to the Zone IDs of the zones. If audio compression is required within the same zone, set Intra-Zone Compression to "Yes". First Communications uses the Intra-zone FAX Profile 2

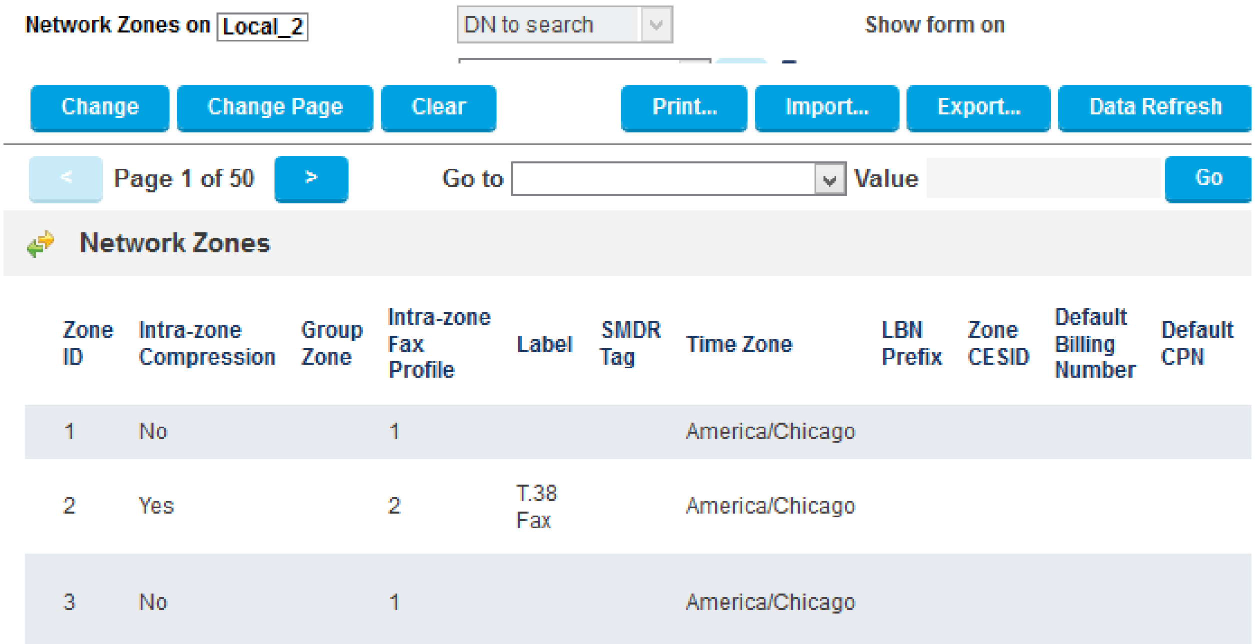


Figure 47: Zone Assignment

NuPoint Configuration

MiVoice Business Setup for Connecting NuPoint

Licensing and Option Selection – SIP Licensing

The first step in setting up the MiVoice Business 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 MiVoice Business are programmed as 5020 IP endpoints on the MiVoice Business. 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 MiVoice Business. Set *** for the Registration Access Code and ### for the Replacement Access Code.

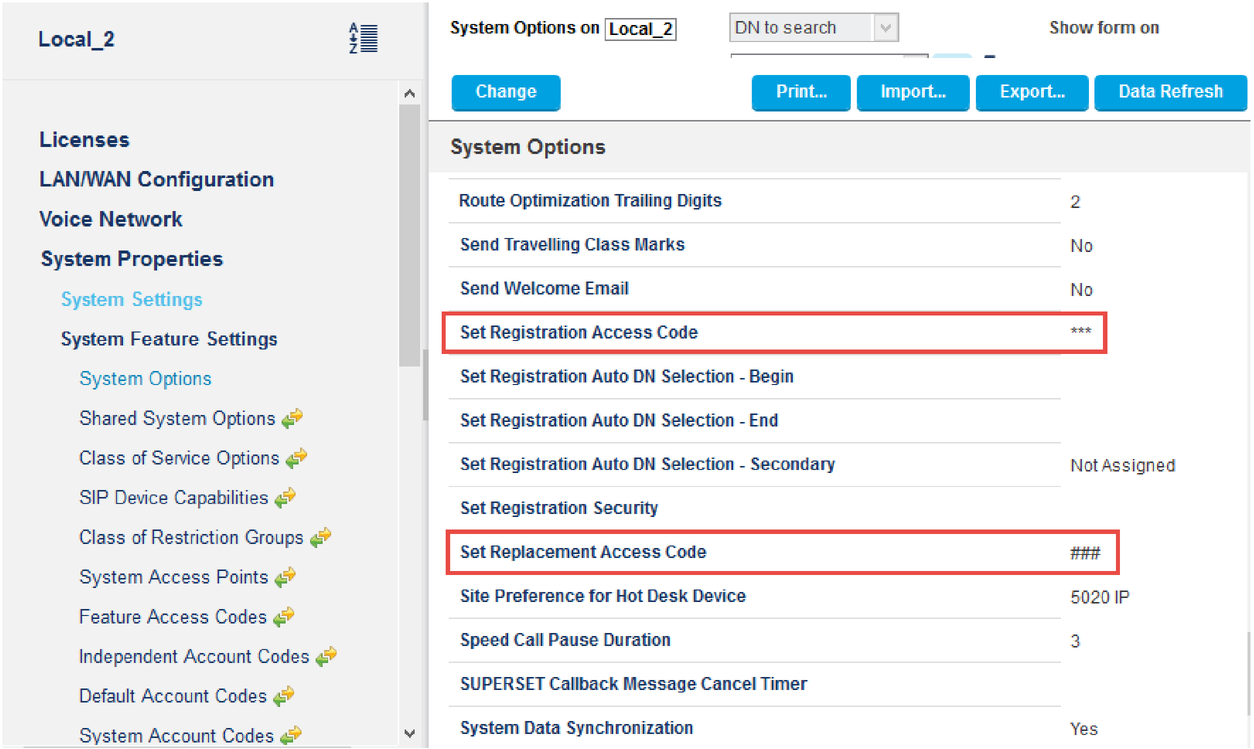


Figure 48: system options

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 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**

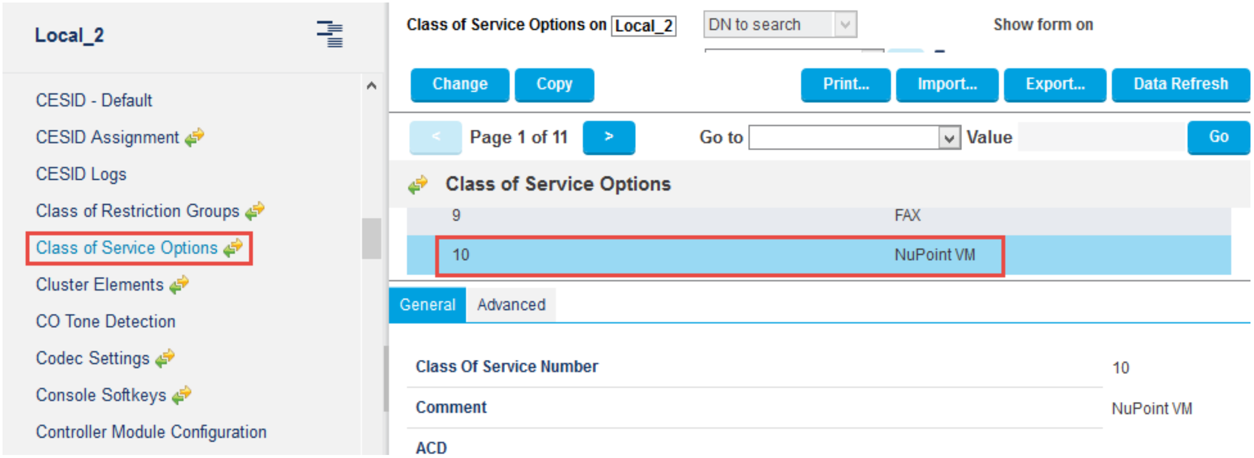


Figure 49: NuPoint Class of Service

IP Endpoints used for NuPoint Ports

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.

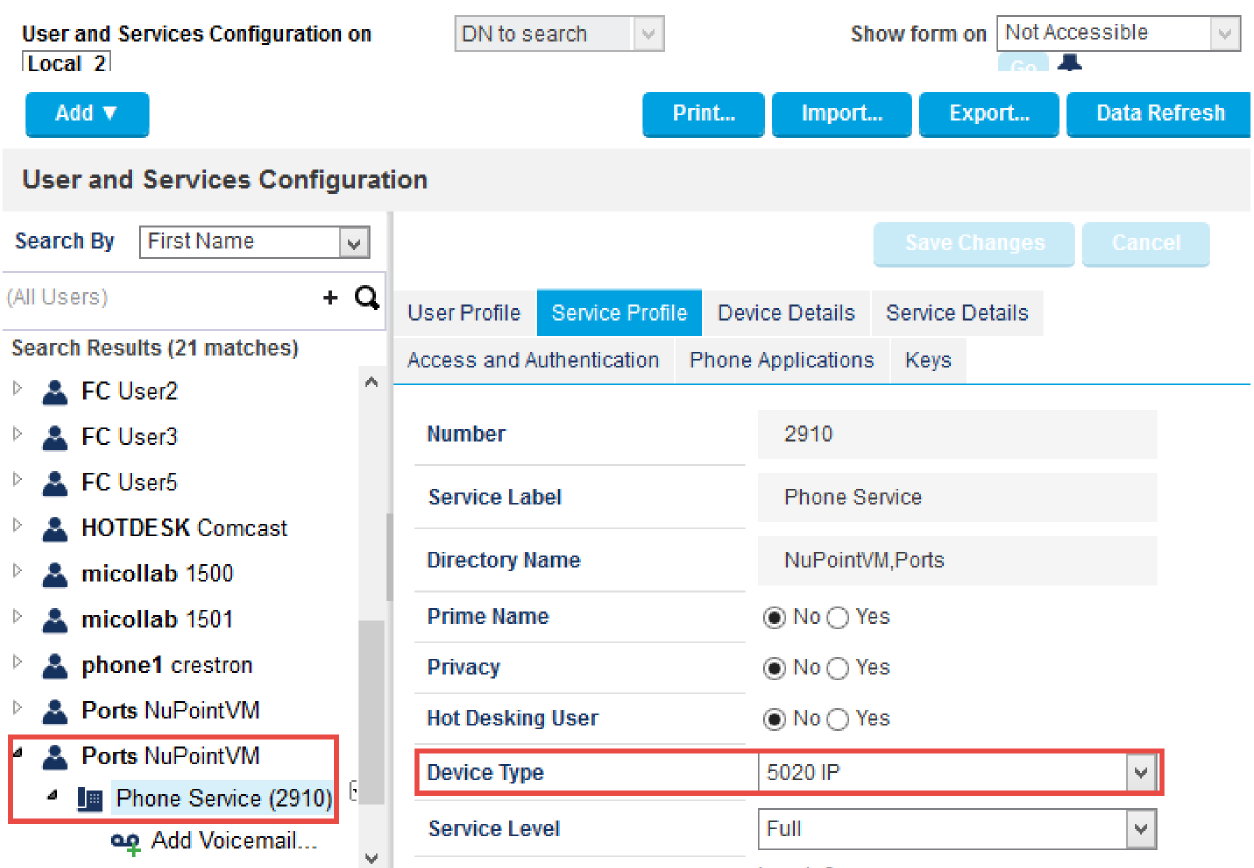


Figure 50: NuPoint Endpoint Configuration

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 10.

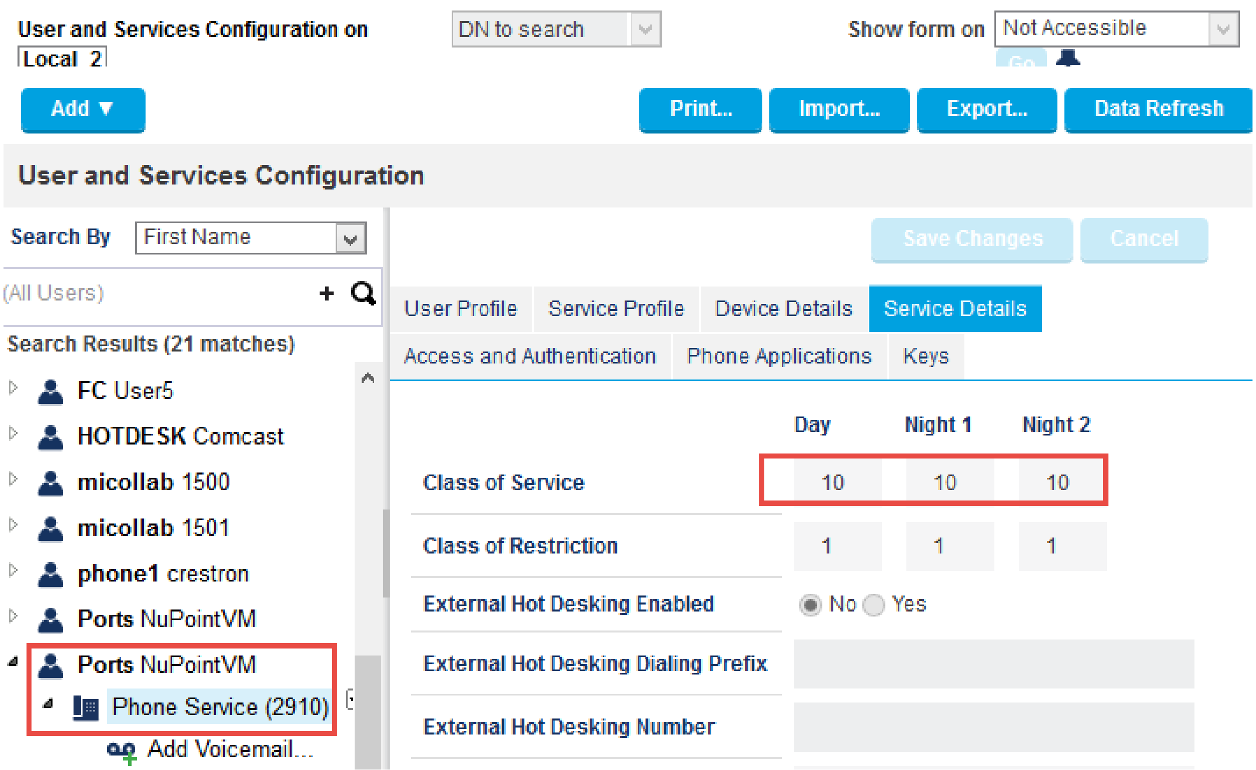


Figure 51: NuPoint Endpoint Configuration – Contd.

Voicemail 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 3000 is created.

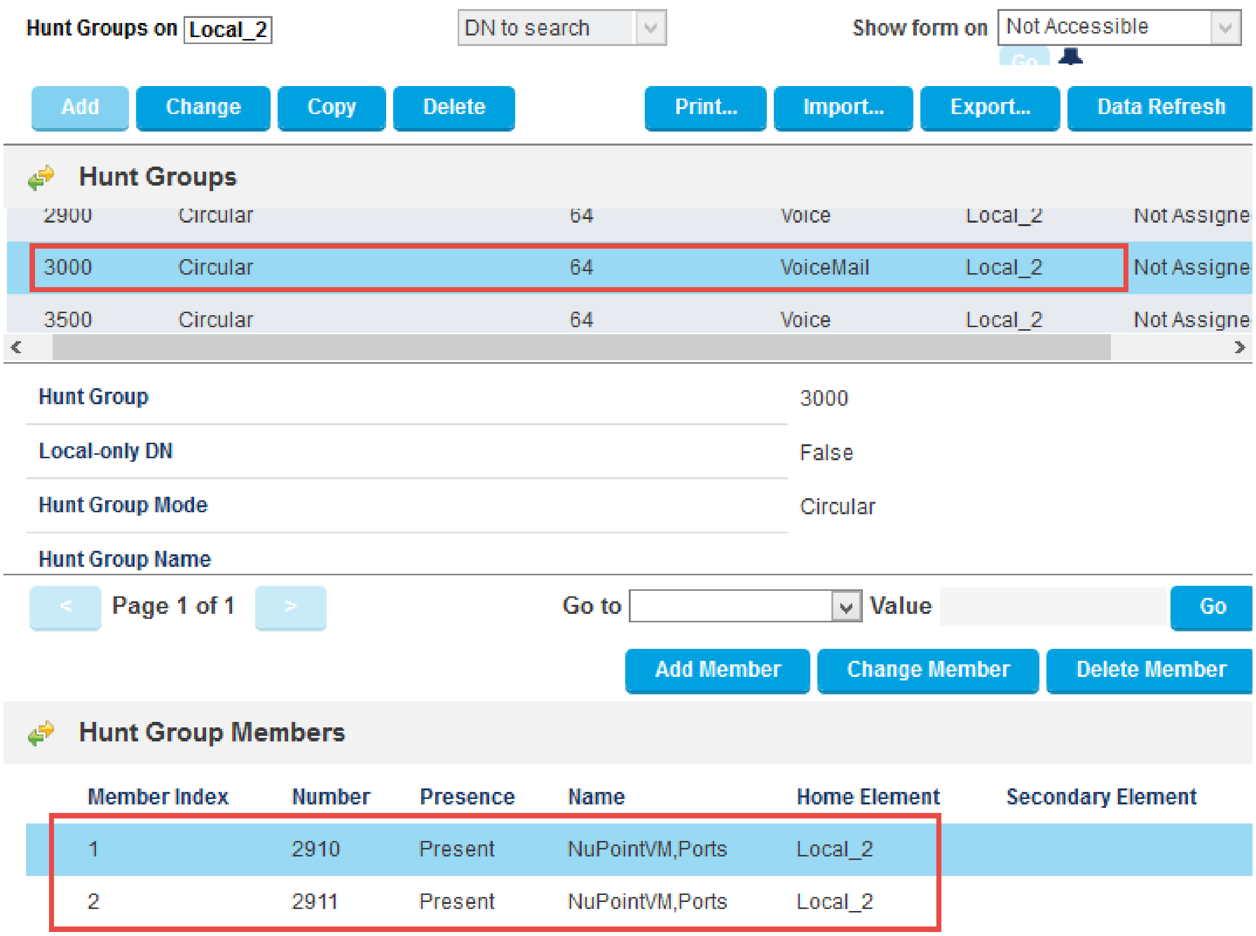


Figure 52: Voicemail Hunt Group Configuration

HCI Reroute 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. 3999 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.

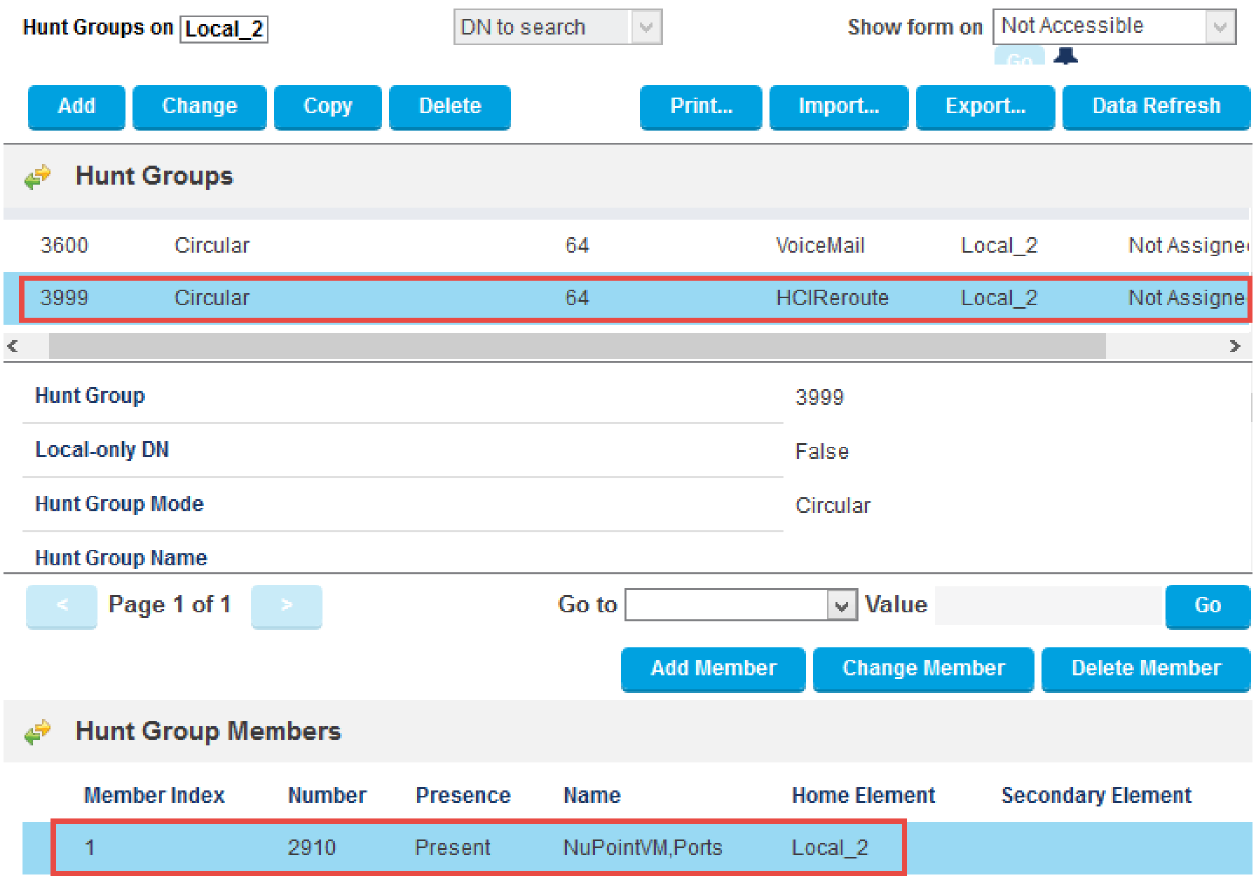


Figure 53: HCIReroute Hunt Group

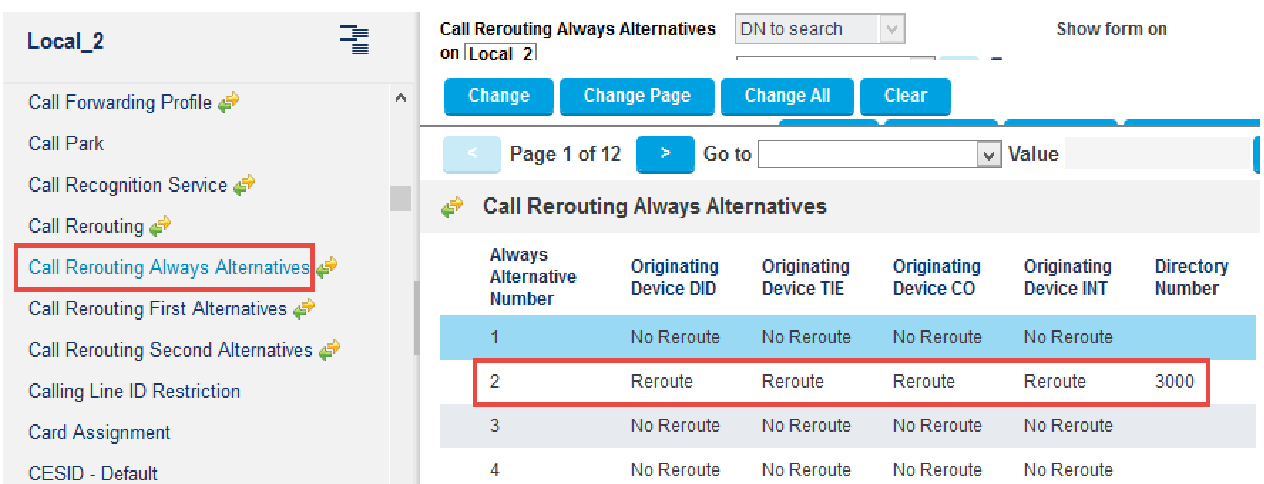


Figure 54: Call Rerouting Always Alternative

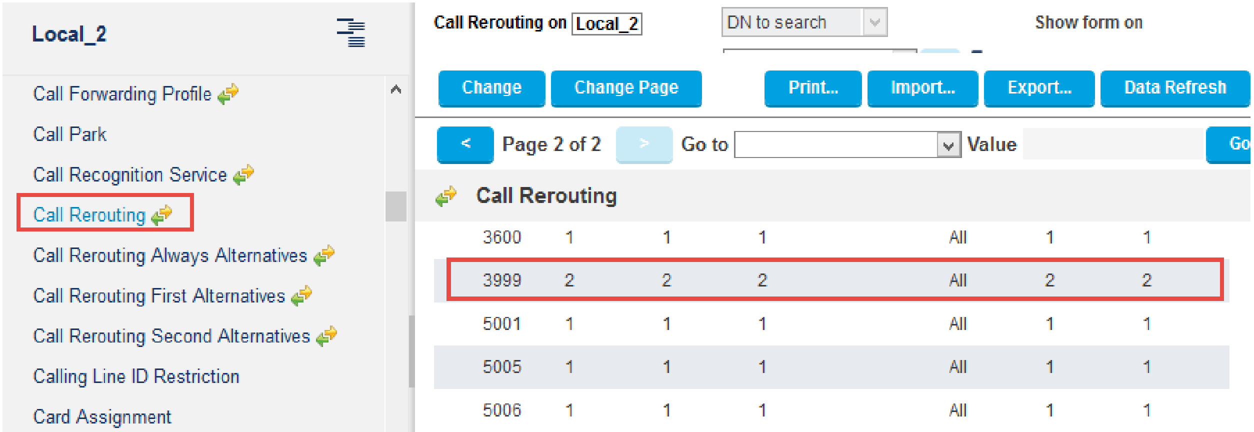


Figure 55: Call Rerouting

MiCollab NuPoint Configuration

- Select **Users and Services**
- Select **the Network Element** tab
- Click **Add**

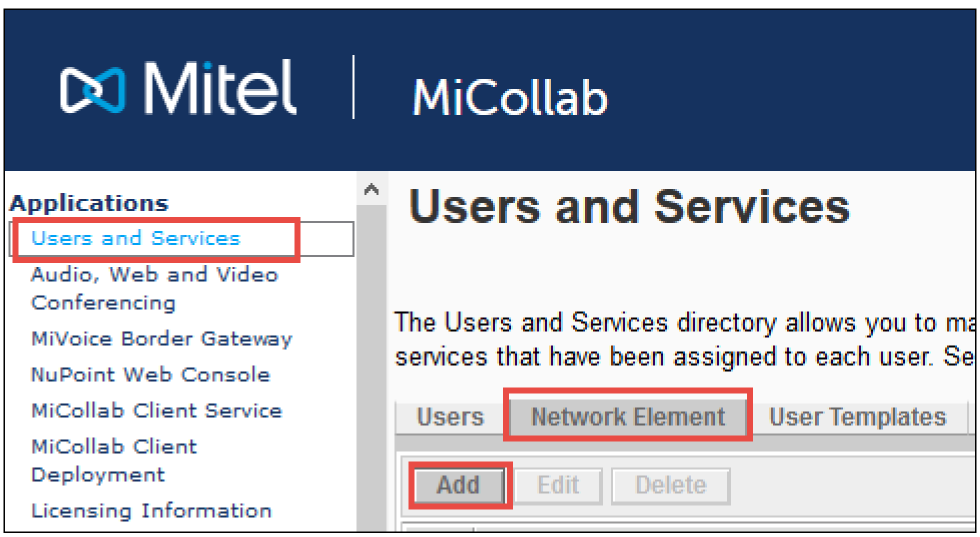


Figure 56: Add Network Element

- Set **System Name: 3300** is given in this test
- Set **Network Address:** Enter the MiVoice Business ICP IP address
- Set **Credentials:** Enter the MiVoice Business ICP administration credentials
- Set **Registration Code: ***** is given which should match the Set Registration Access Code in the System Options section
- Set **Replacement Code: ###** is given which should match the Set Replacement Access Code in the System Options section
- Set **Default COR: 1** is given to all fields in this setup
- Set **Call Forward Destination Directory Number: 3000** is given, which is the Hunt Group Number for NuPoint Voicemail
- Click **Save**

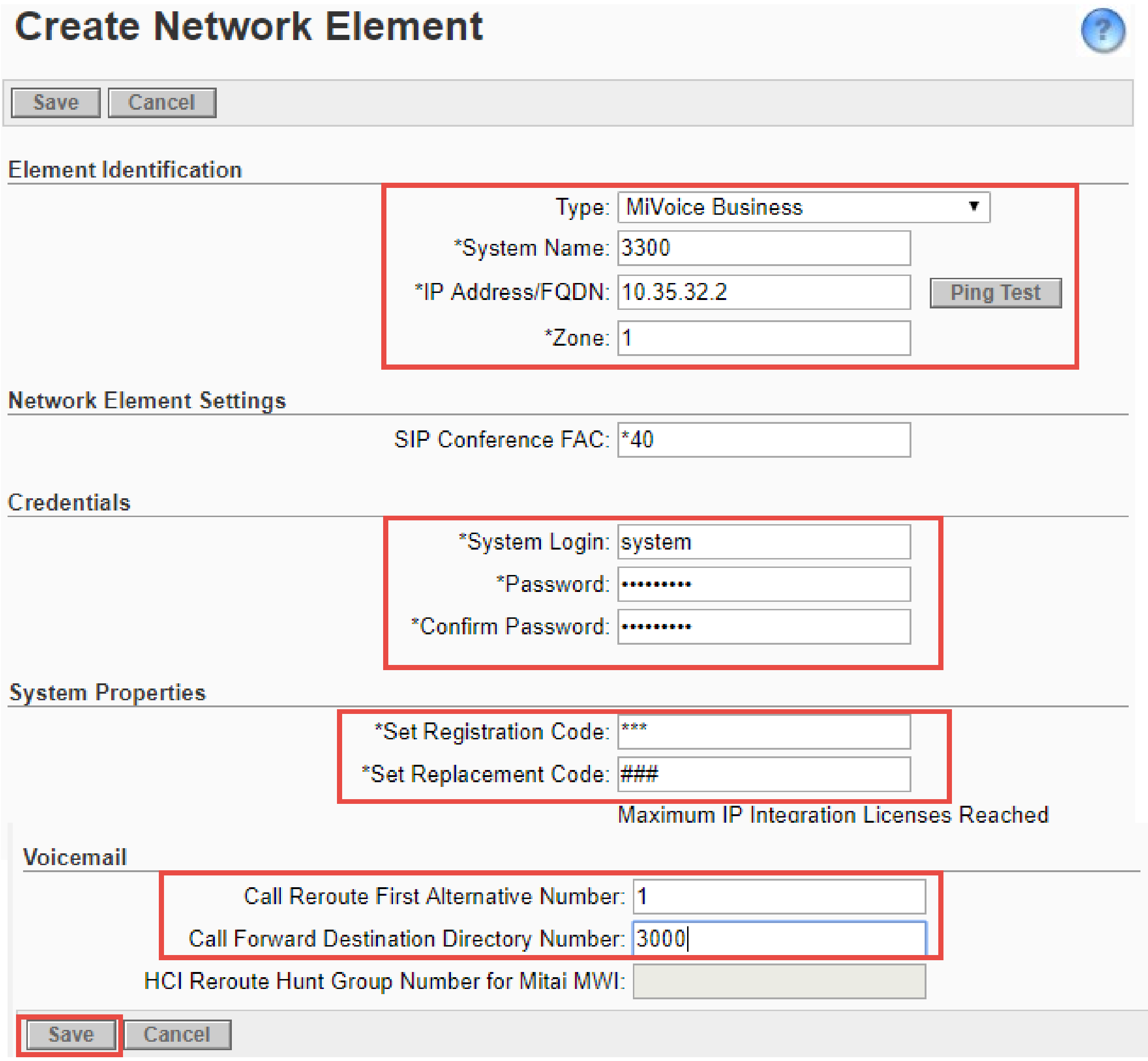


Figure 57: Network Element – Contd.

Voicemail Line Group

- Click On **NuPoint Web Console**

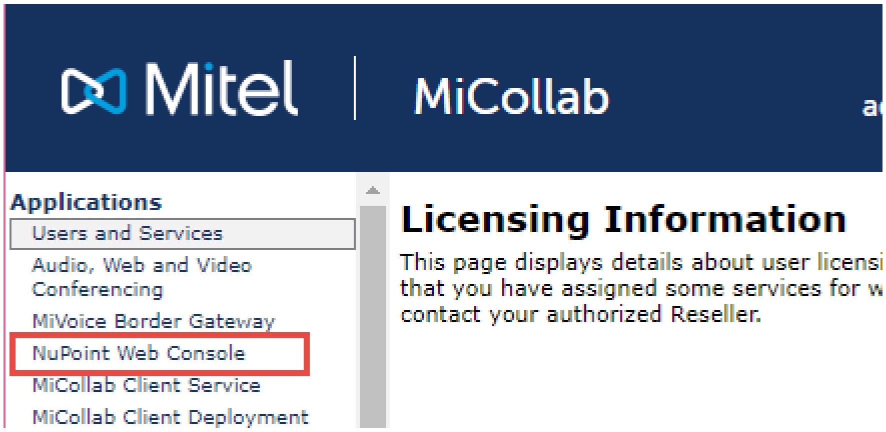


Figure 58: Voicemail Line Group Configuration

- Click **Add**

Figure 59: Voicemail Line Group Configuration – Contd.

- 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).
- Enter a Name such as Voicemail to describe for what the line group will be used.
- Choose NuPoint Voice for the Application and NuPoint Voice for the User Interface.

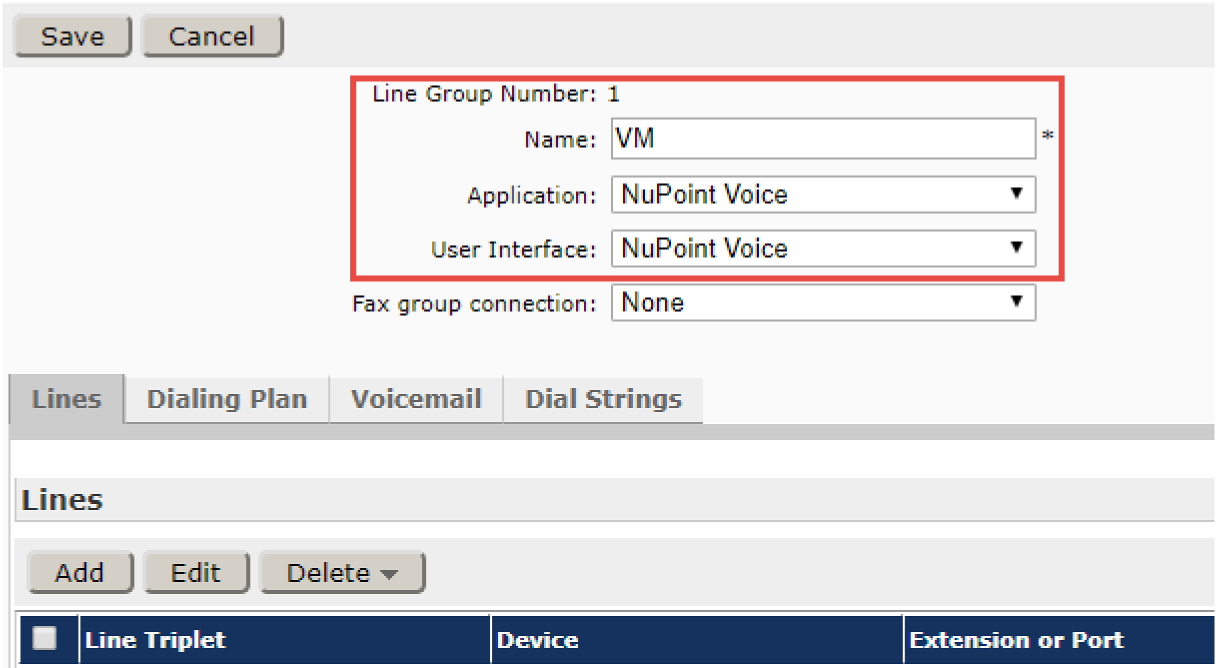


Figure 60: Adding Line Group

- Click **Add** under the Lines heading. This will bring up the Line Triplet dialogue box
- 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).
- Select PBX: MitelPBX. Enter the first extension number that was created in the section IP Endpoints used for NuPoint Ports in the Mapping field.
- Click **Save**

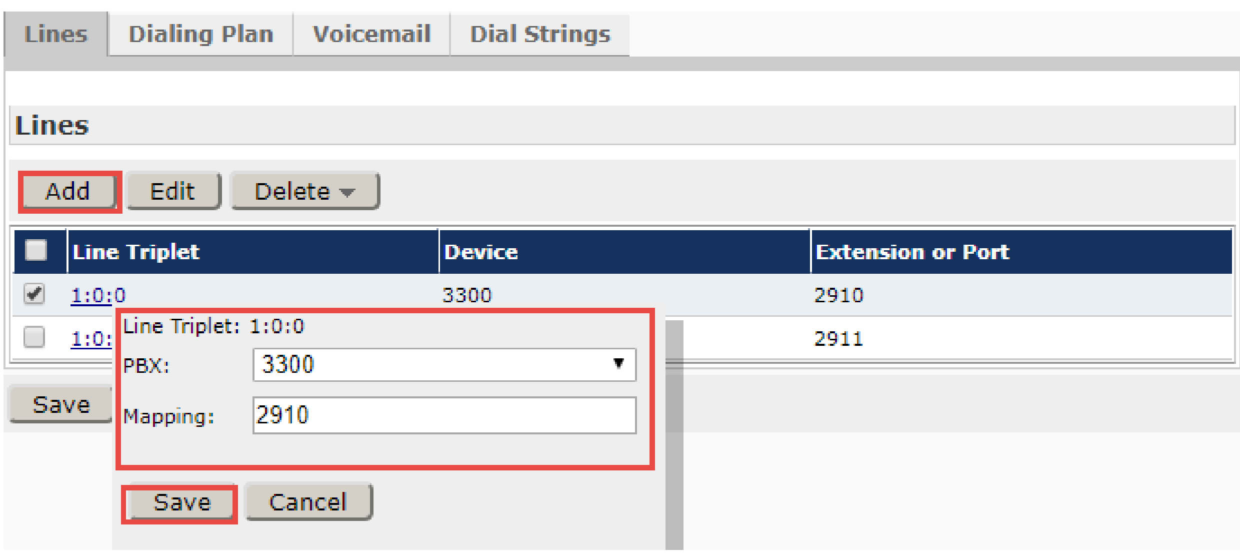


Figure 61: Adding Lines

- 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.

- Click **Save**

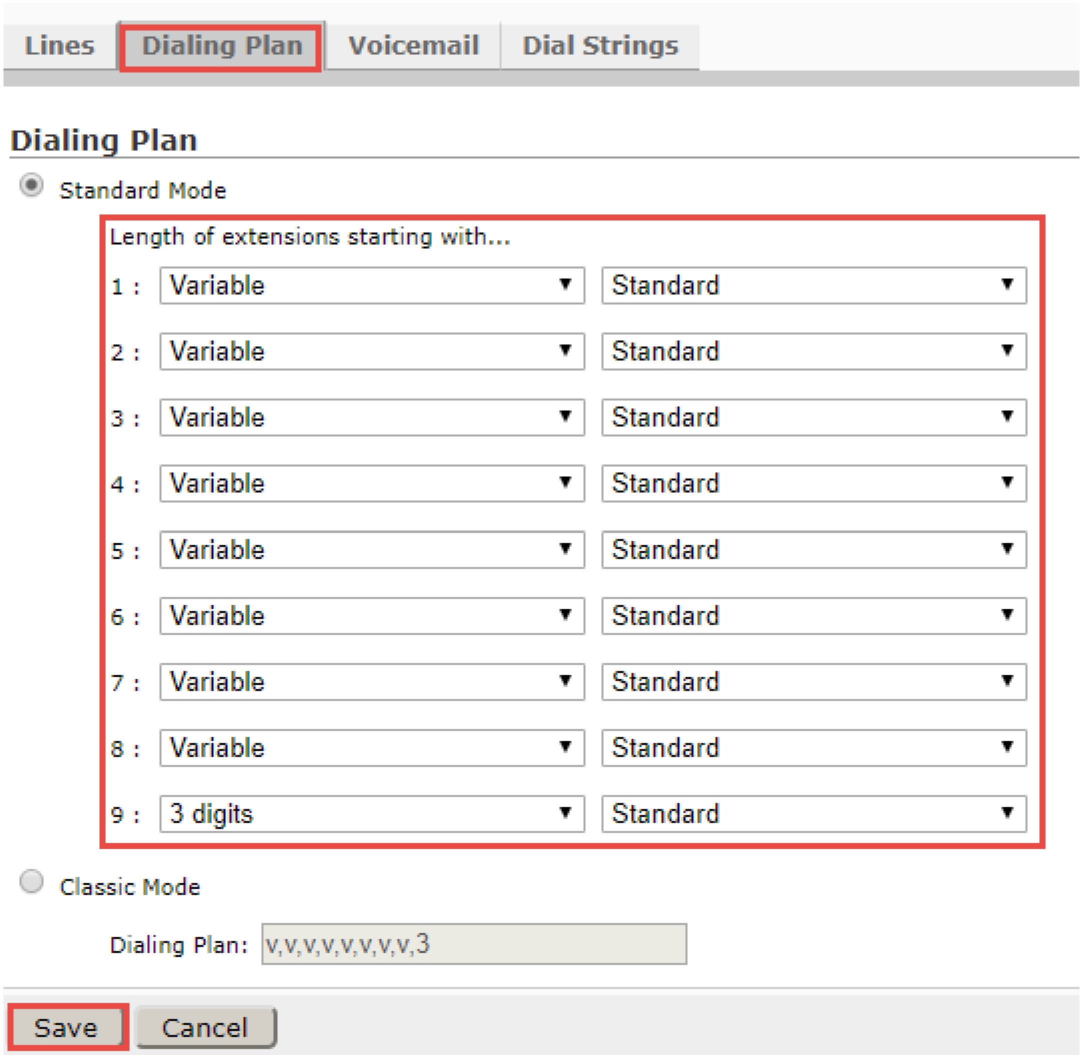


Figure 62: Adding Dial Plan

- The next step is to commit the changes that have been made to the offline configuration.
- Click the **Commit changes & Exit** link under the Offline Configuration heading
- Click **Commit**
- Next click the **Activate** link at the top of the page.
- On the Activate Offline Configuration page, deselect the check boxes for Wait for MWI queue to empty and Wait for Pager queue to empty
- Click **Activate**

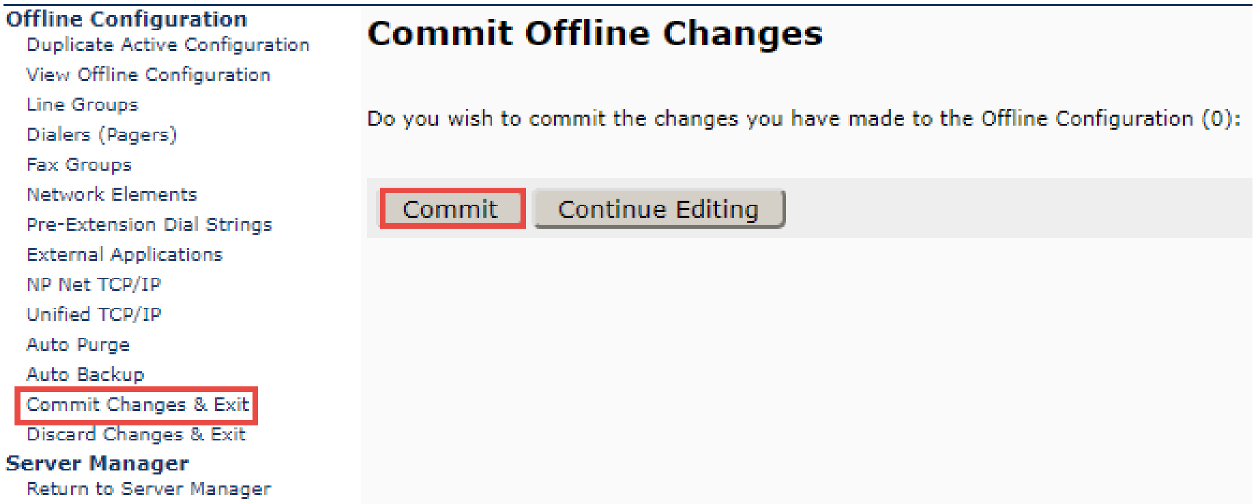


Figure 63: Committing Offline Changes

Adding Mailboxes

- NuPoint with MAS and Single Point Provisioning allows for programming MiVoice Business phones, users and NuPoint Mailboxes from the MAS interface. We assume MiVoice Business phones and users were configured in the MiVoice Business Configuration Notes Section and this chapter only covers adding mailboxes.

- Click **Add**

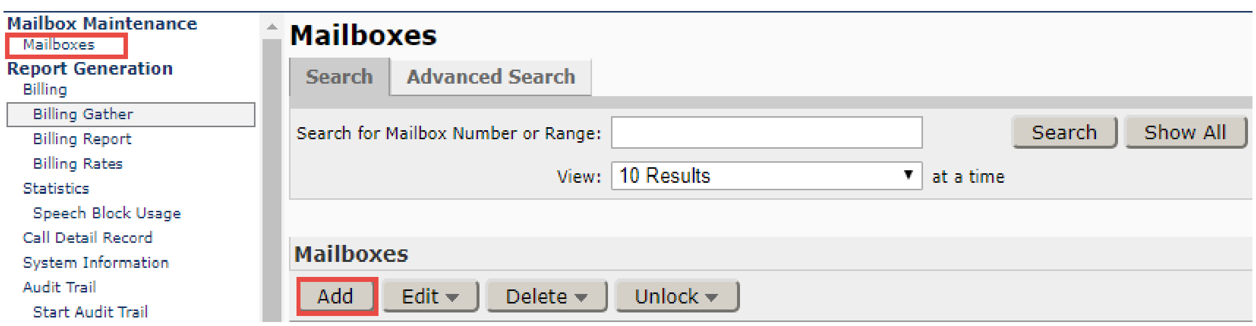


Figure 64: Add Mailbox

- **Mailbox Number** 5112 is created for this test. Under the **General** tab, set the proper **Name, Passcode** and associated MiVoice Business phone/user as **Extension.**

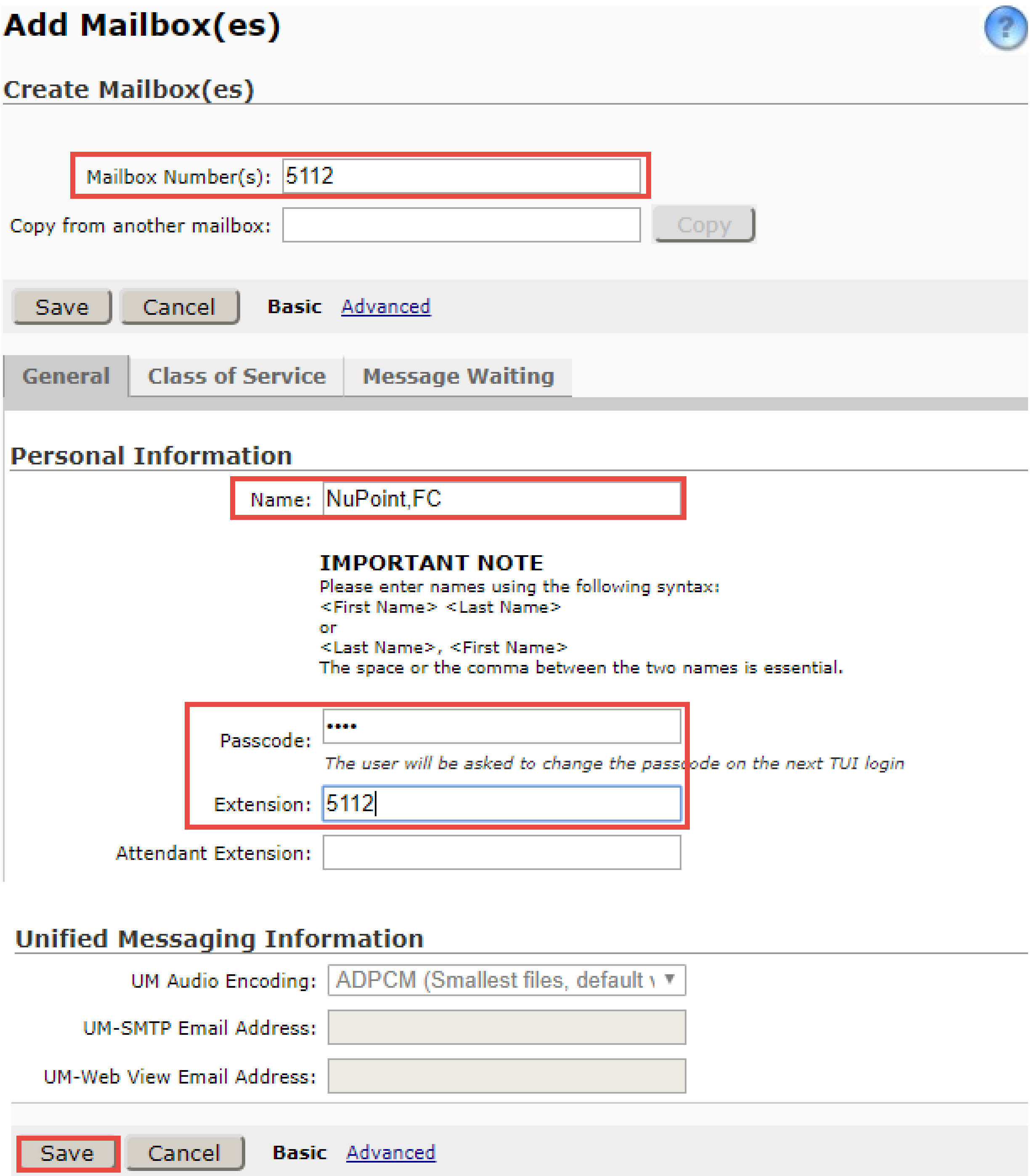


Figure 65: Add Mailbox – Contd.

- Under the **Message Waiting** tab, select **Mitai Messaging** as **Type**
- Click **Save**
- Click **Done** when pop-up window shows the mailbox was added successfully

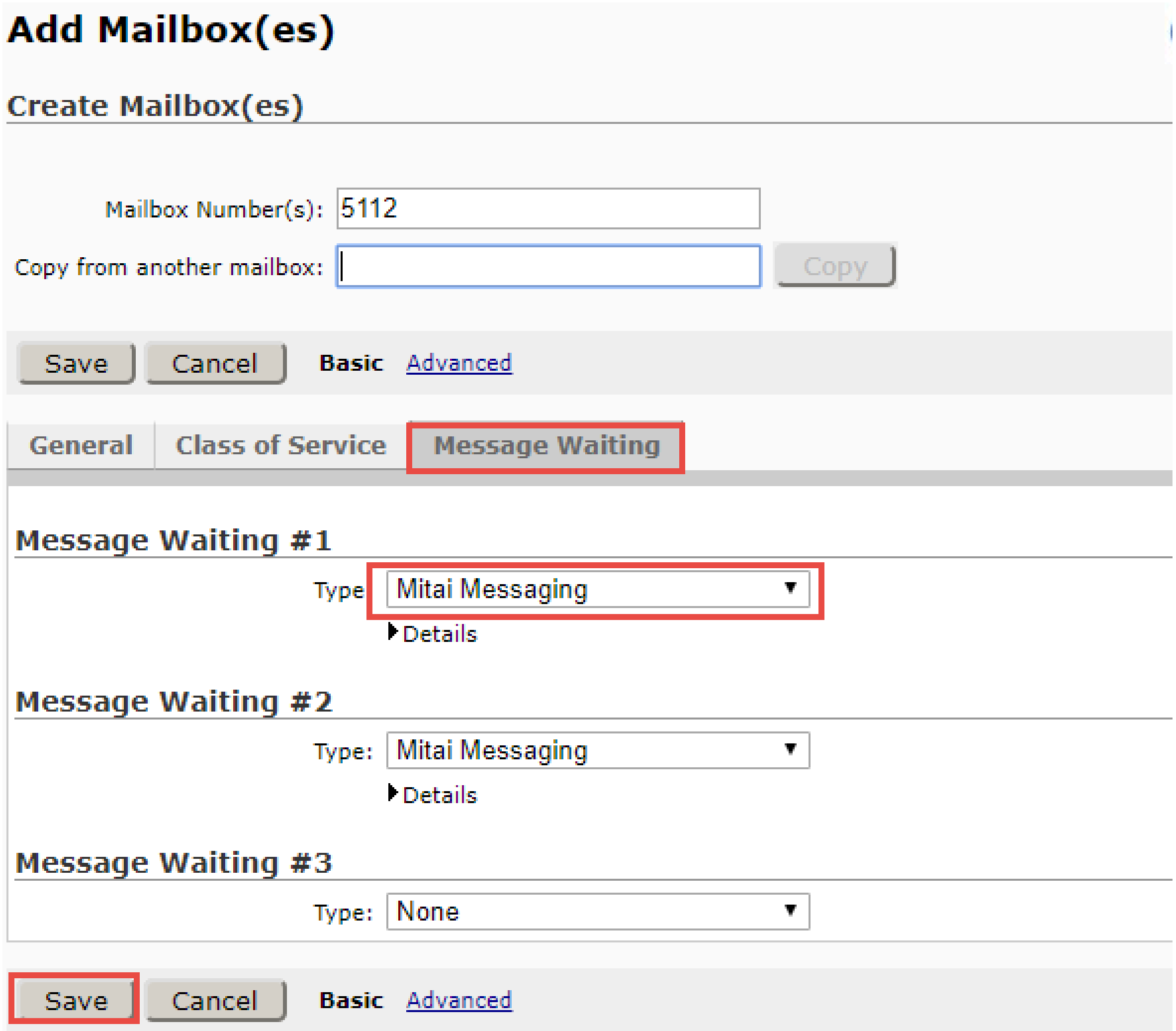


Figure 66: Add Mailbox – Cont.

MiCollab Client Configuration

MiCollab Client Setup on MiVoice Business

Create and modify a **SIP Device Capabilities** for MiCollab

- In the example shown below, the SIP Device Capabilities Number 71 is used.
- The Red boxes denote the changed default values. The tabs not shown have default values.

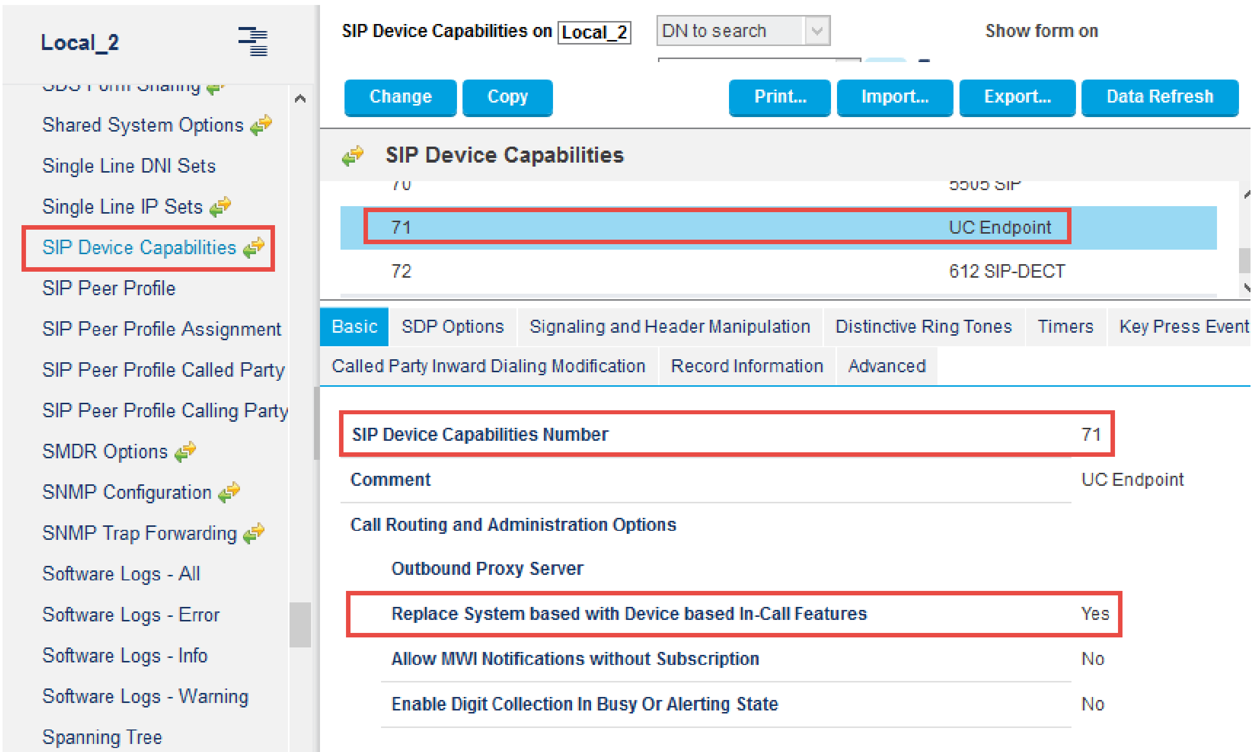


Figure 67: SIP Device Capabilities – Basic

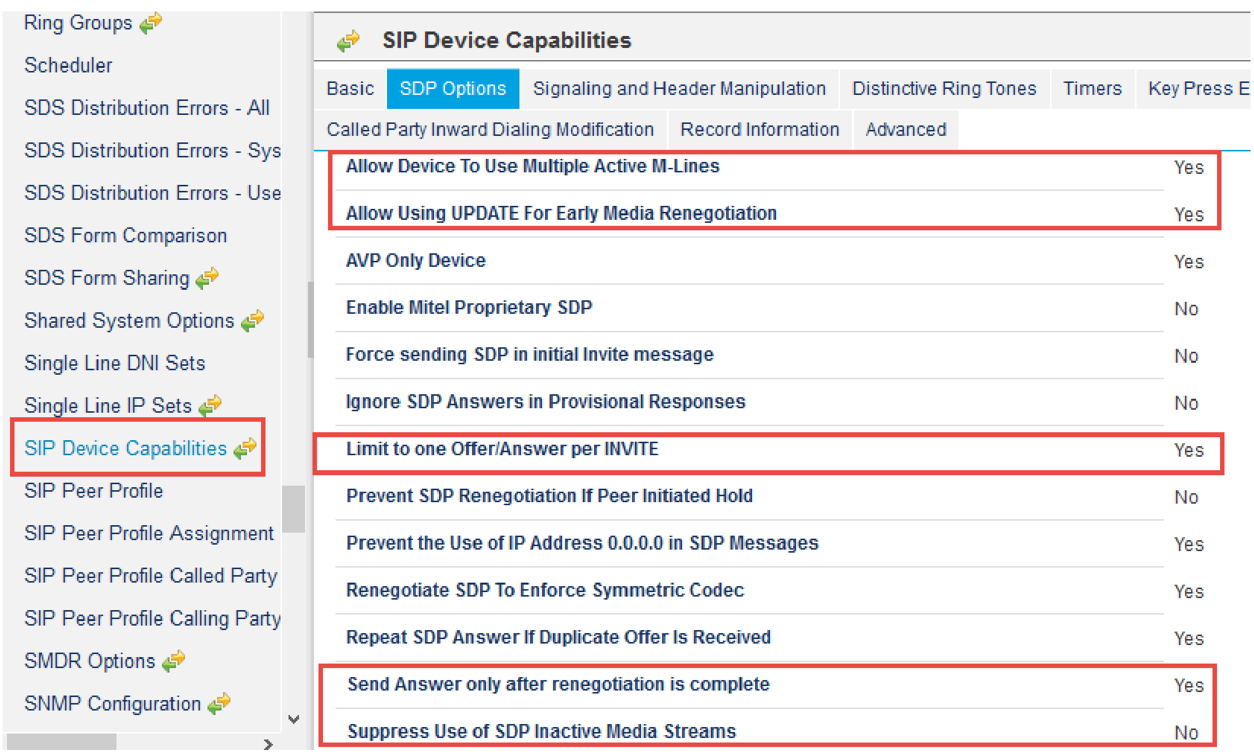


Figure 68: SIP Device Capabilities - SDP Options

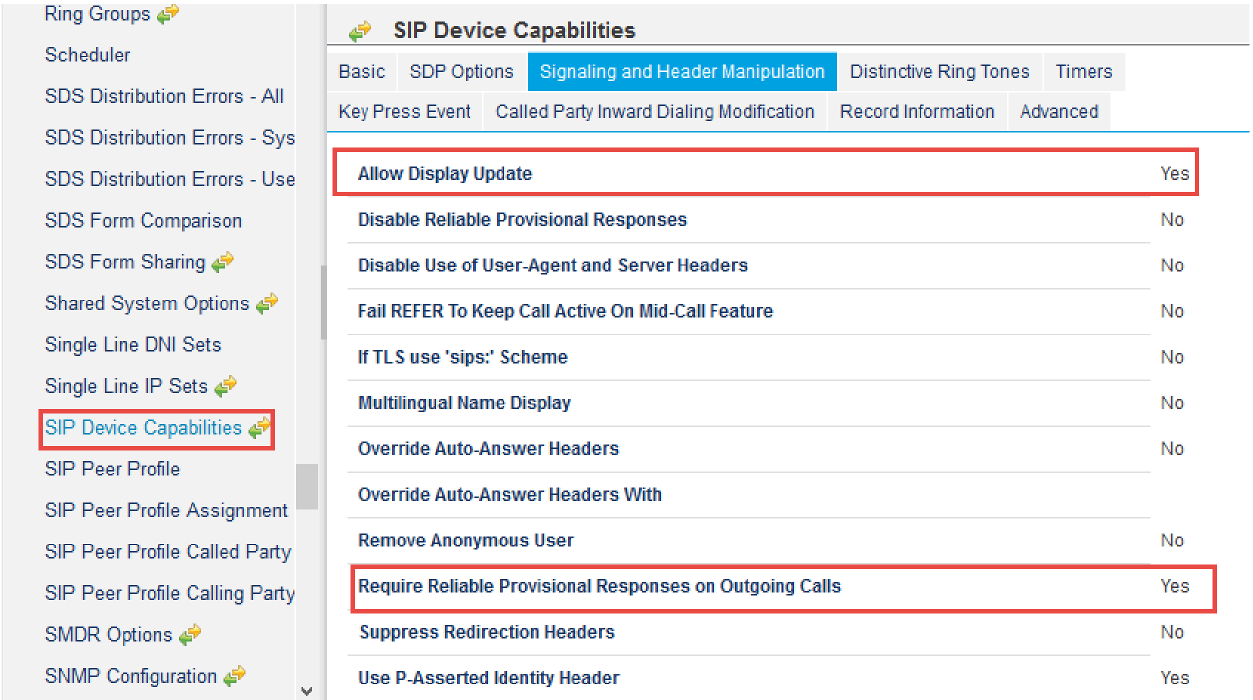


Figure 69: SIP Device Capabilities - Signaling and Header Manipulation

MiCollab Client Setup on MiCollab

Create the MiCollab client user accounts and respective phone numbers. The user login credentials are added to sign into the MiCollab client.

- Select the **Users** tab
- Click **Add**

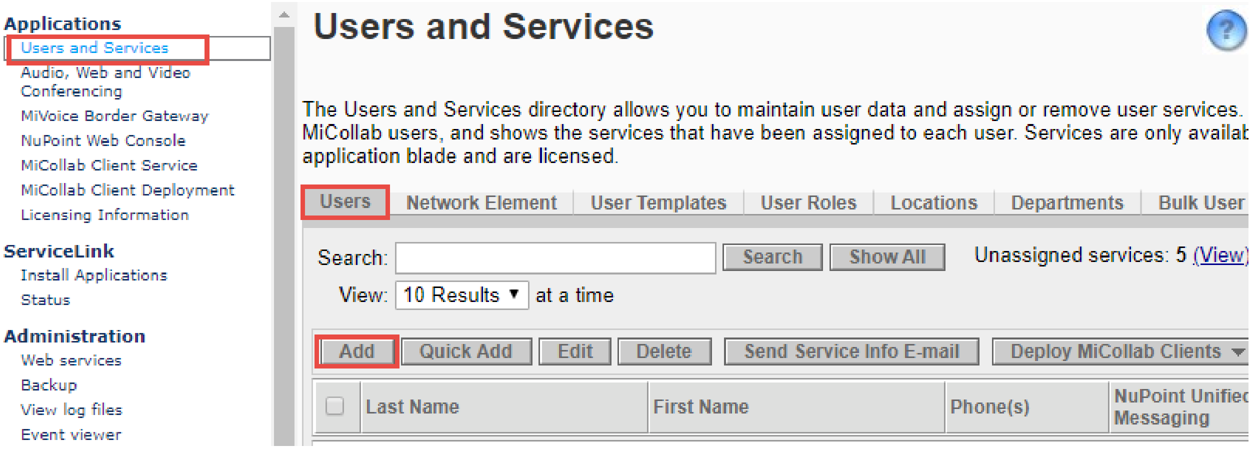


Figure 70: Add MiCollab User Account

Configure the following User fields as follows

- **First, Last Name:** miclient, miclient1 is used in this example
- **Primary Email Address:** Enter the user's email address
- **Login:** MiCollab User login
- **Password:** Enter a temporary password for initial password or Generate Password
- **TUI Passcode:** Enter 4 to 10 digits for the passcode

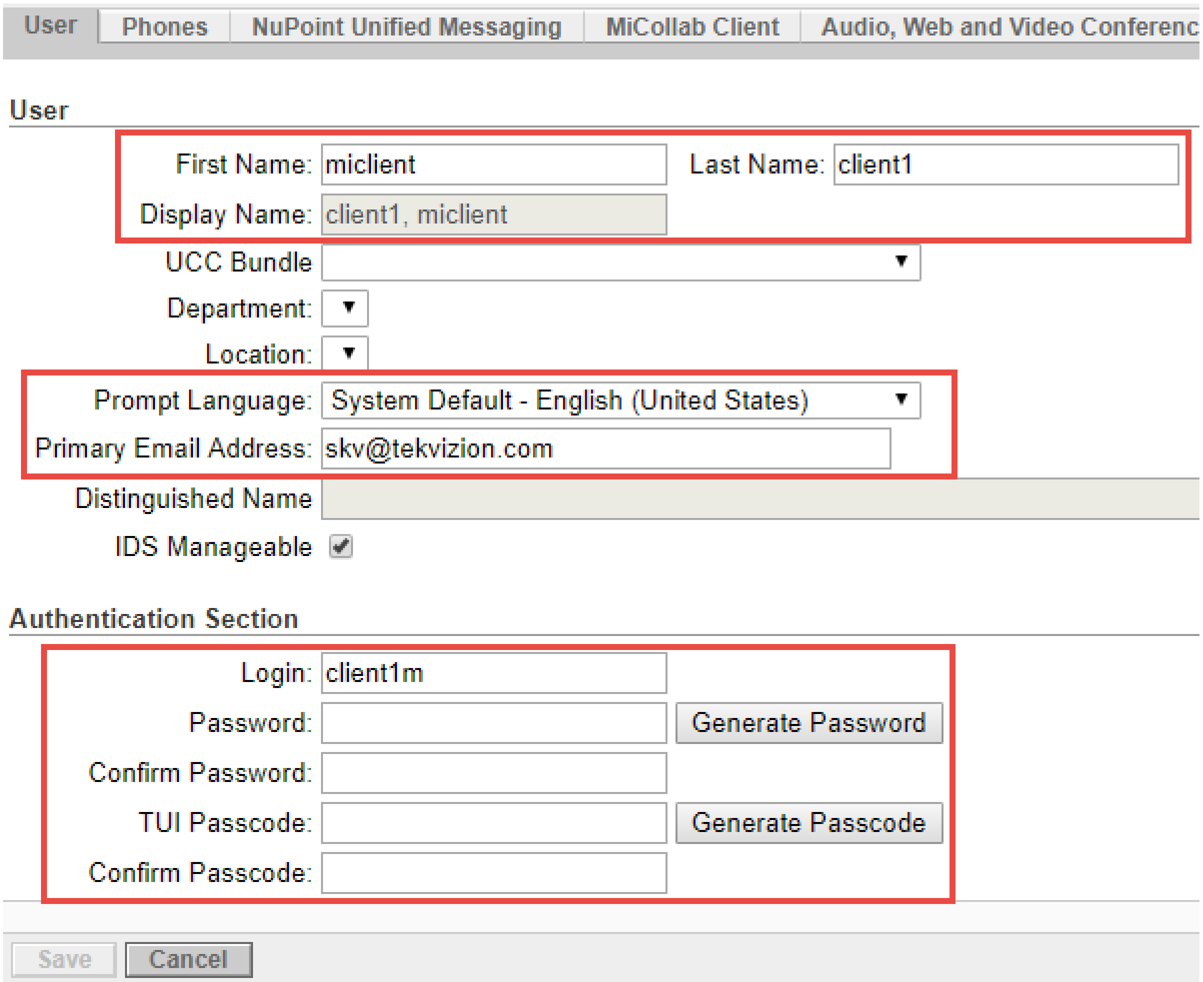


Figure 71: Add MiCollab User Account - Contd.

Select the Phones tab and configure the Desk Phone and Soft Phones as follows

Desk Phone

- **Number:** Phone extension used as Desk phone
- **Service Label**: Desk Phone
- **Device Type:** UC Endpoint type for MiCollab client
- **Deployment Profile:** default

Soft Phone

- **Number:** Phone extension used as Softphone
- **Service Label:** Soft Phone
- **Device Type:** UC Endpoint type for MiCollab client
- **Deployment Profile:** default

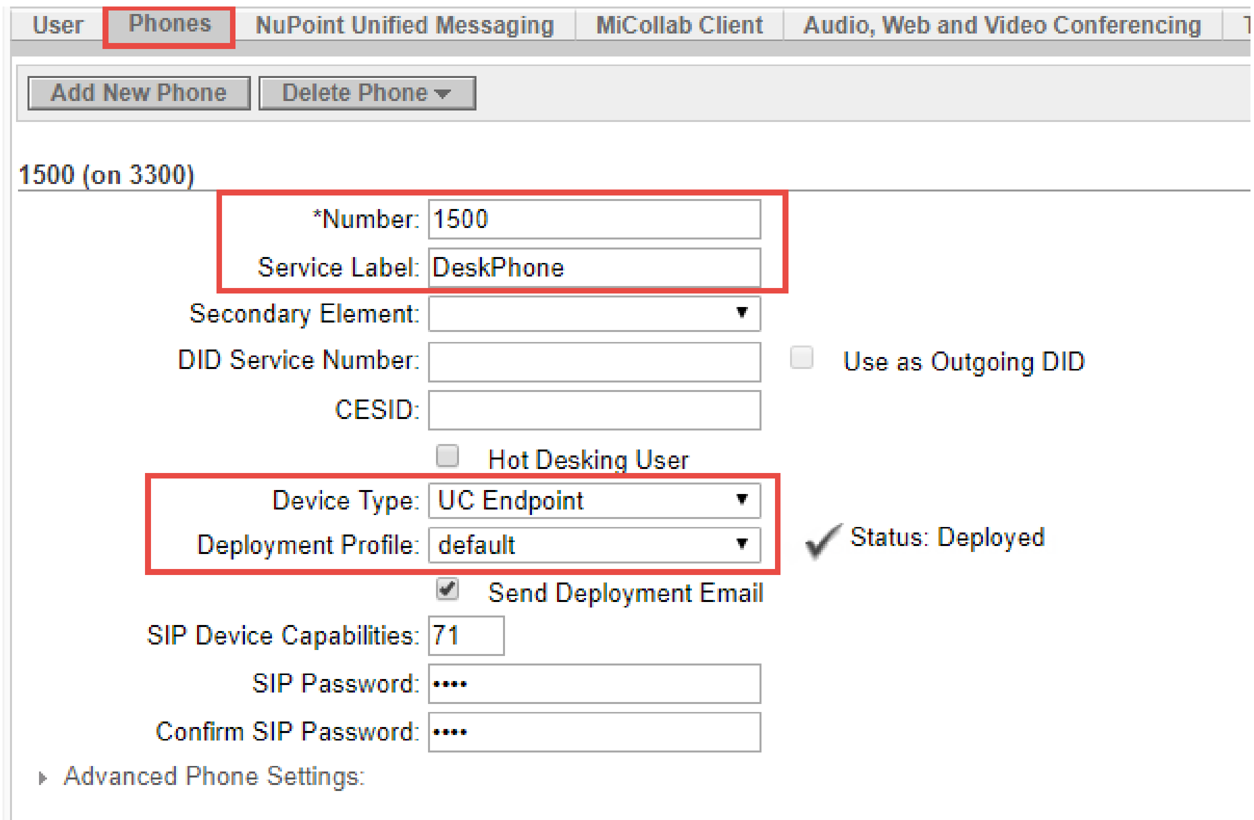


Figure 72: MiCollab Phone Configuration

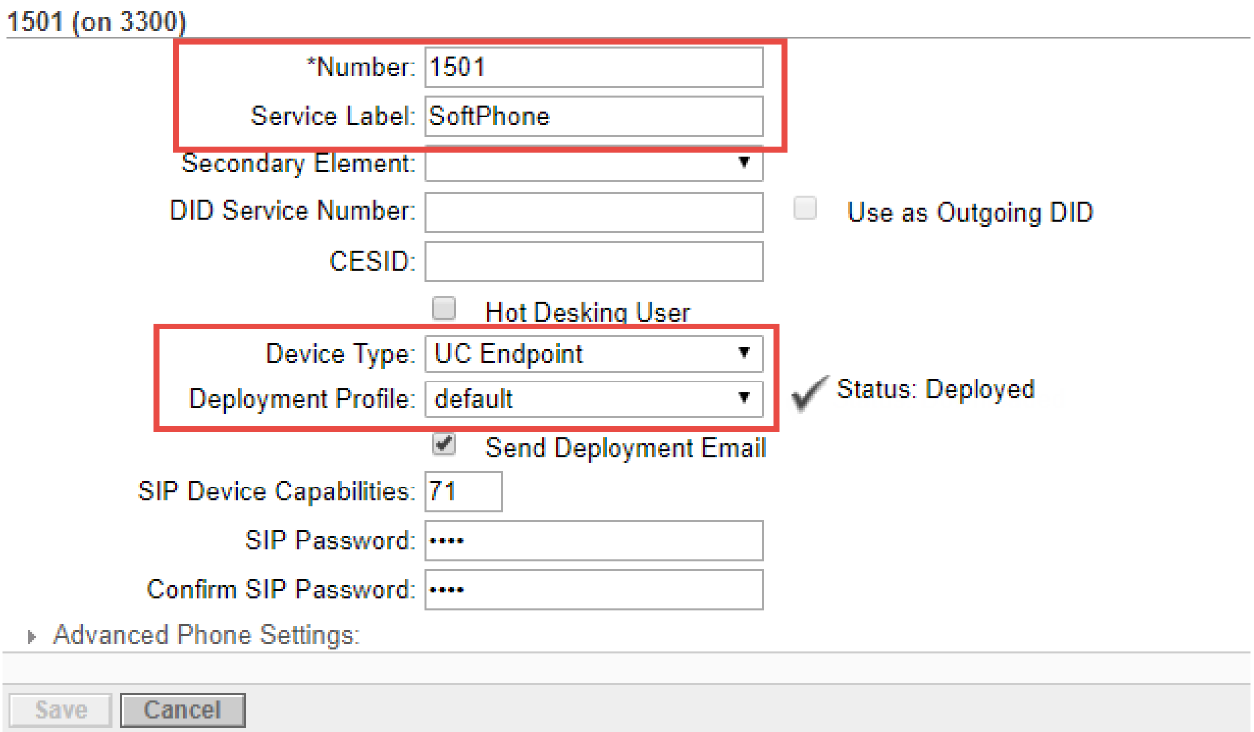


Figure 73: MiCollab Phone Configuration – Contd.

Advanced Phone Settings for Desk and Soft Phones

- **Service Level:** Full
- **Zone ID:** 1
- **Call Coverage Service Number:** 1
- **Class of Service:** 1
- **Class of Restriction:** 1

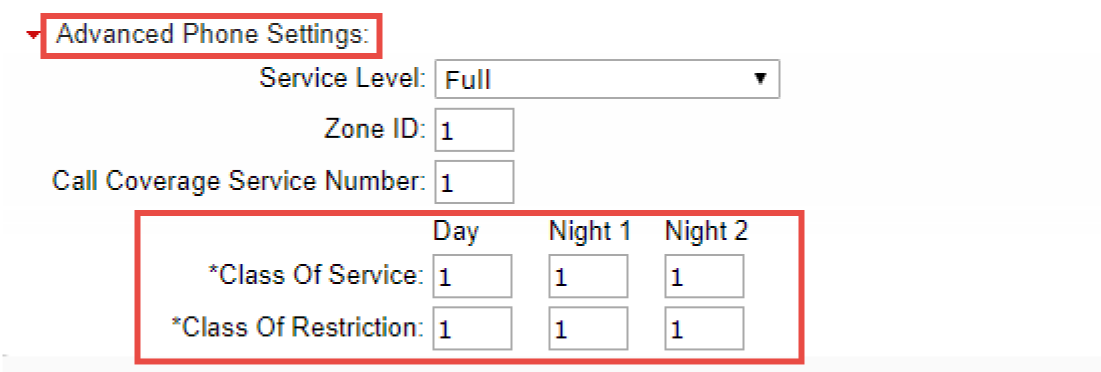


Figure 74: MiCollab Phone Configuration - Contd.

Select the Phones tab and configure the Desk Phone and Soft Phones as follows

- **Feature Profile:** Select the full_Licensed
- **Desk phone extension**: 1500, (Desk phone previously assigned)
- **Soft phone extension:** 1501, (Soft phone previously assigned)
- **Mailbox number:** Other Mailbox was selected
- **Deployment Profile:** default

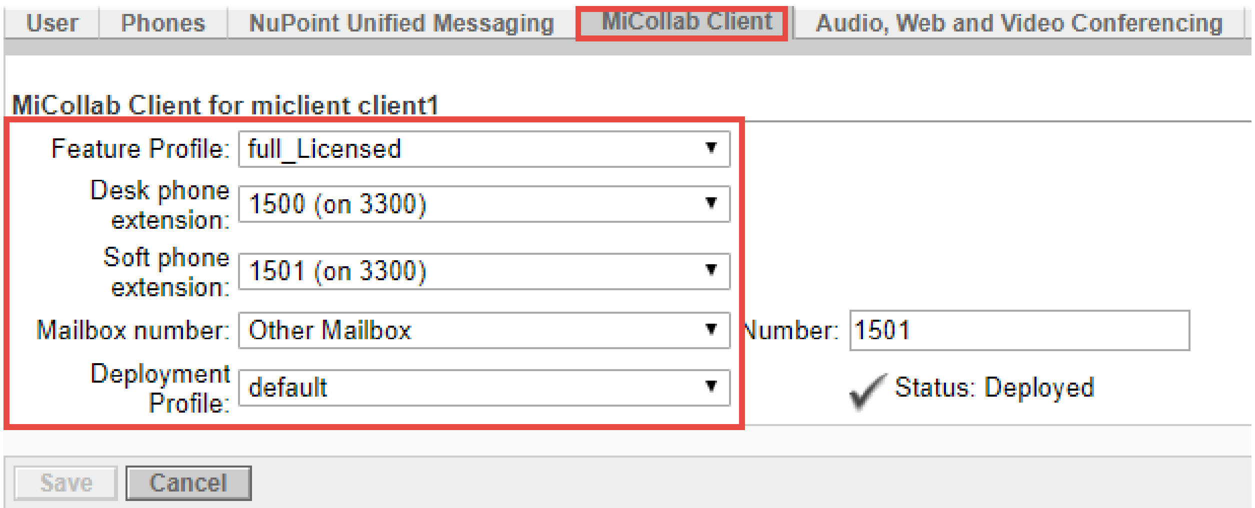


Figure 75: MiCollab Phone Configuration - Contd.

After the MiCollab users are created and performing a list all users will show the users as follows.

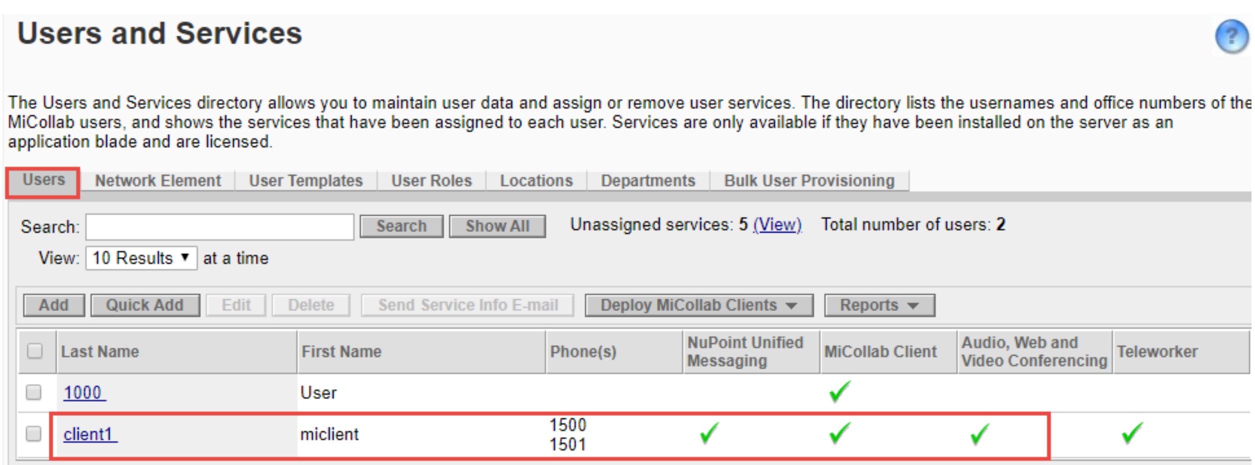


Figure 76: MiCollab Phone Configuration – Contd.

MiVoice Border Gateway Configuration Notes

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

- Login to MBG and click **Mitel Border Gateway**
- In left pane, click **MiVoice Border Gateway** tab and then right pane, click **system Configuration**
- Select **Network profiles**

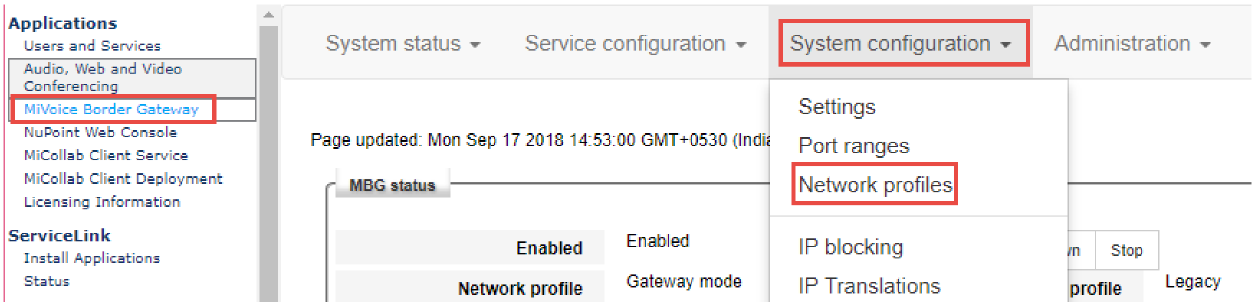


Figure 77: Network Profiles

- Click the "→" beside **Server-gateway configuration on the network edge**
- Click **Apply**

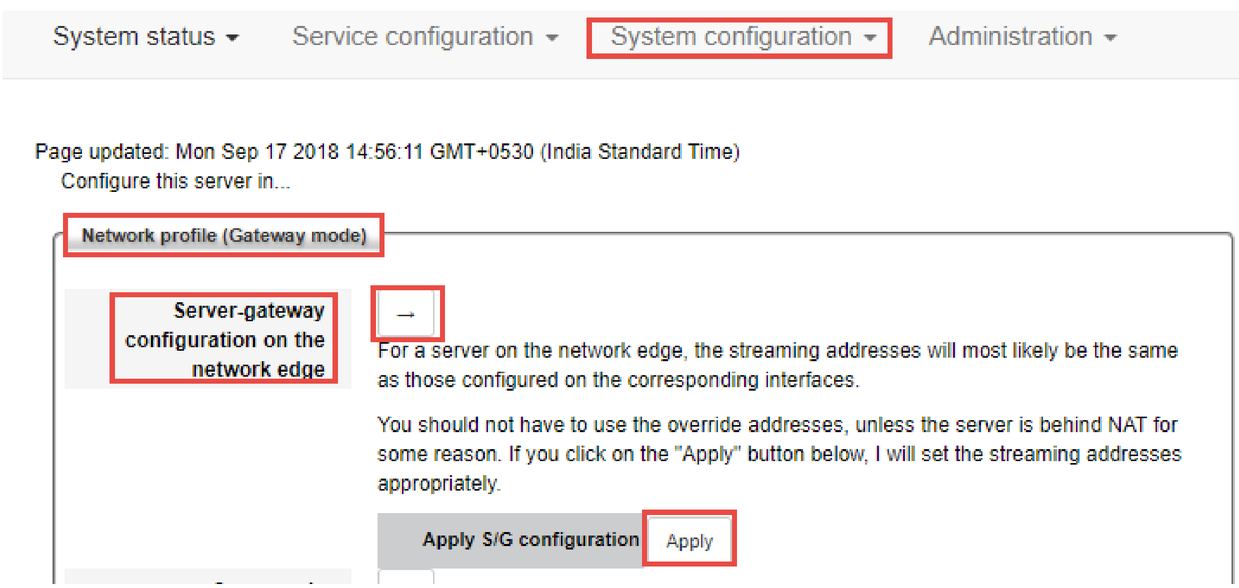


Figure 78: Network Profiles – Contd.

In order to make the mid-call feature works for External Hot Desk User, setup KPML username and password by navigating to **System Configuration** > Settings. Click **Edit.**

- Set **KPML username:** administrator is given which is the same **as Subscription User Name** created in the section SIP Peer Profile
- Set **KPML password:** Enter the same password as **Subscription Password** created in the section SIP Peer Profile

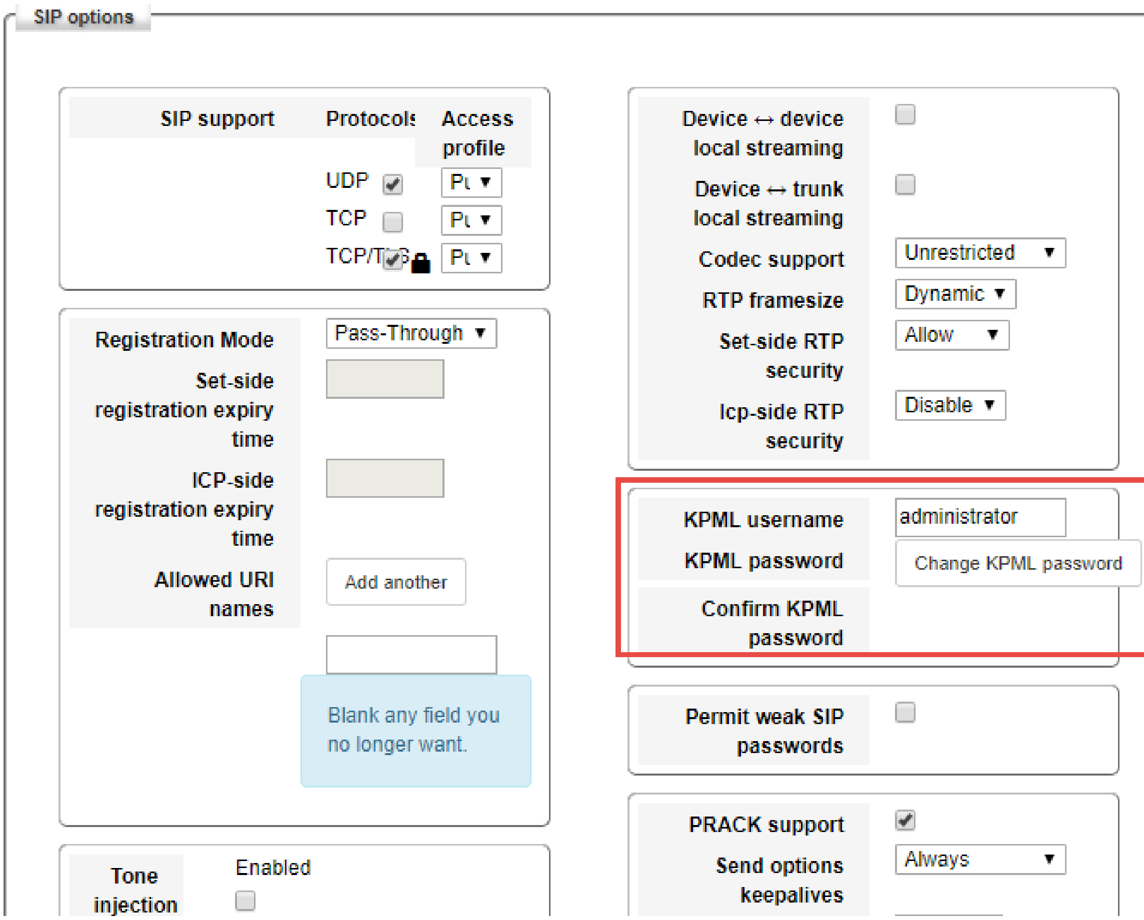


Figure 79: MBG Settings

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

- click **Service Configuration**
- Select **ICPs**

Figure 80: MiVoice Border Gateway Configuration

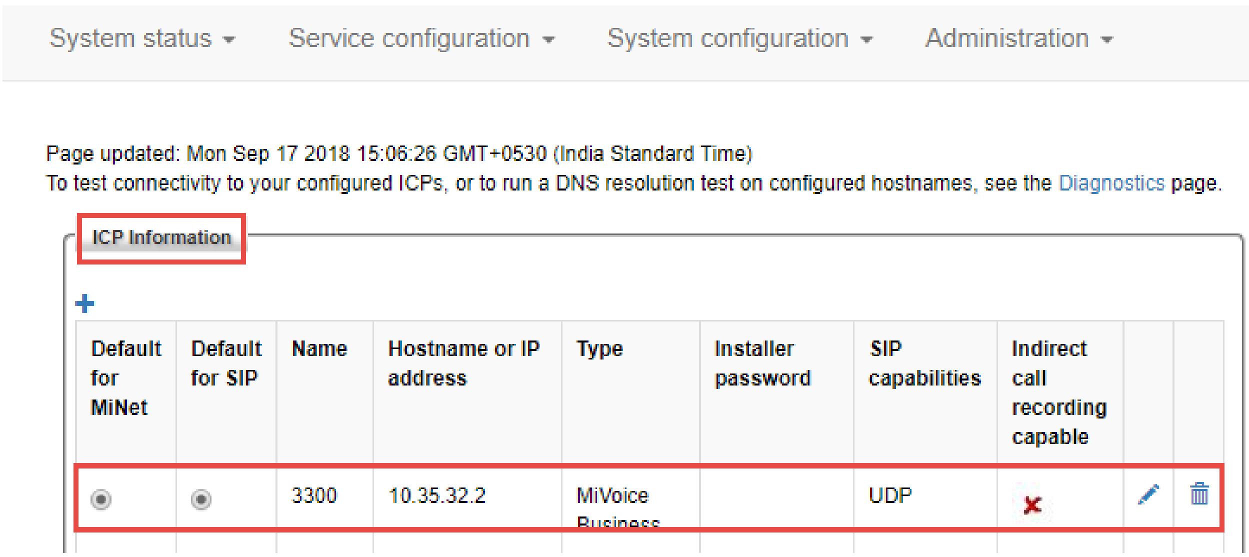


Figure 81: MiVoice Border Gateway Configuration – Contd.

On **ICPs** page, ensure that the "working" MiVoice Business is configured. If needed, click **Add ICP** link and add a new Mitel switch.

Click **Update** Default ICPs

To add a new SIP trunk:

- Click **Service Configuration** tab and then click **SIP trunking**
- Click **Add a SIP trunk** link

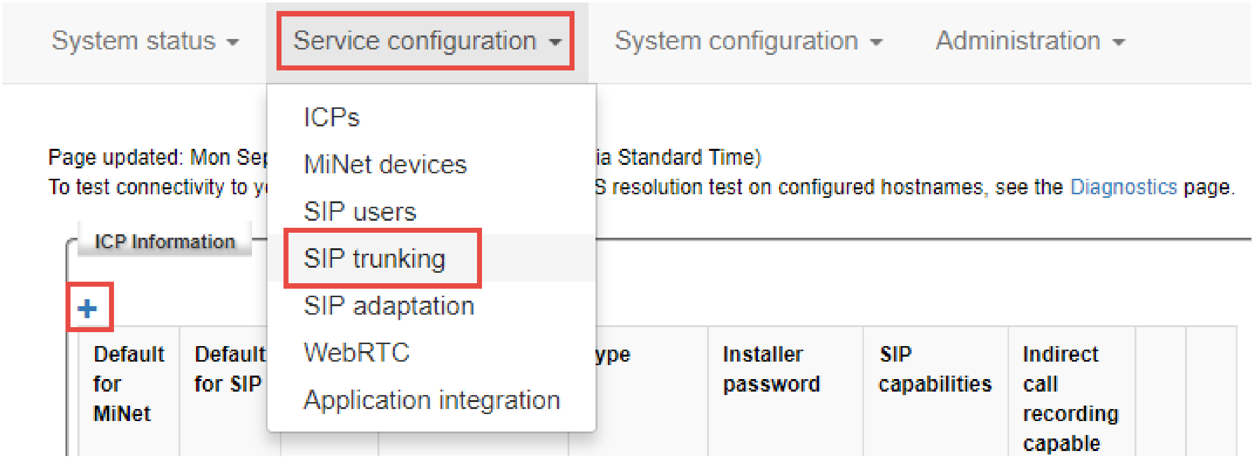


Figure 82: SIP Trunking Configuration

Enter the SIP trunk details as follows:

- Set **Name:** FirstComm is given in this setup
- Set **Remote trunk endpoint address:** Enter the IP address / FQDN for your deployment, crealersouthsipconnect.adpt-tech.com is used here
- Set **Remote Trunk Endpoint Port:** 5060 is used
- Set **DNS SRV Support:** Unchecked
- Set **PRACK Support:** Use master setting
- Set **Authentication username**: None
- Set **Icp-side RTP security**: Disable
- Set **RTP address override**: None
- Set **Match:** Request URI
- Set **Rule**: set the Assigned DID Format 22471XXXXX
- Set **Primary**: MiVoice Business 3300 PBX's -IP
- The remaining settings are optional and could be configured as required
- Click **Save**

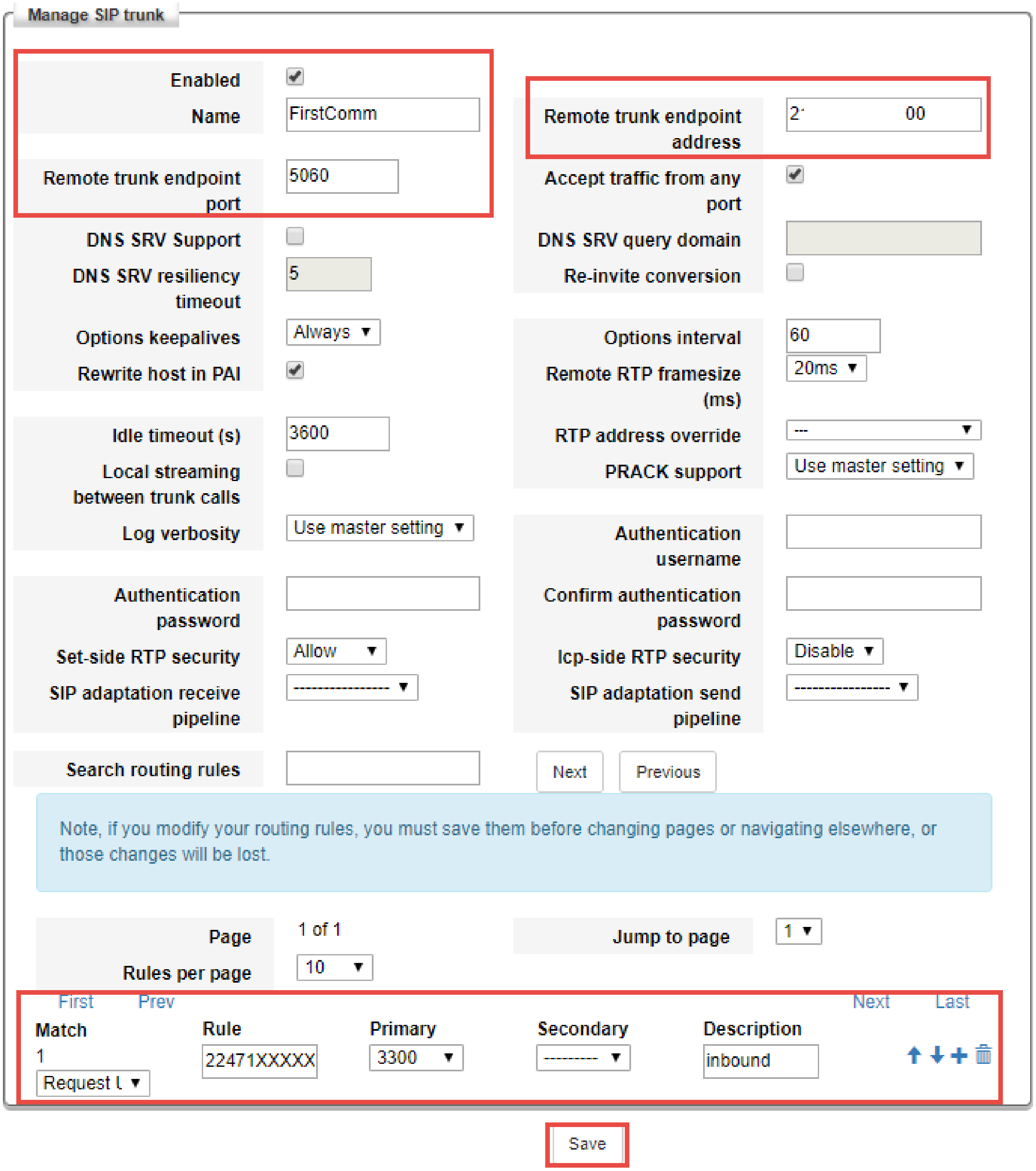


Figure 83: SIP Trunk Configuration Settings