Spectrum Enterprise SIP Trunking Service
Mitel MiVoice Business 7.X
IP PBX Configuration Guide

About Spectrum Enterprise:

Spectrum Enterprise is a division of Charter Communications following a merger with Time Warner Cable and acquisition of Bright House Networks. Spectrum Enterprise is a national provider of scalable, fiber technology solutions. The Spectrum Enterprise portfolio includes networking and managed services solutions, including Internet access, Ethernet and Managed Network Services, Voice, TV and Cloud solutions. Our industry-leading team of experts works closely with clients to achieve greater business success.

About this document:

Spectrum Enterprise assures IP PBX compatibility by conducting interoperability testing to ensure any potential compatibility issues have been resolved prior to installation. Please review the IP PBX configuration instructions in this guide prior to your installation date.

Be advised that this document may contain references to Time Warner Cable Business Class. All references to Time Warner Cable Business Class, TWCBC or TWC should be read as Spectrum Enterprise.

Thank you,

Spectrum Enterprise

Last Update: 2017-06-13

MITEL – SIP CoE

Technical
Configuration Notes

Configure MiVoice Business 7.X for use with Time Warner Cable Business Class SIP Trunking service

MARCH 2015

SIP COE 15-4940-00363

TECHNICAL CONFIGURATION NOTES

NOTICE

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

TRADEMARKS

Mitel is a trademark of Mitel Networks Corporation.

Windows and Microsoft are trademarks of Microsoft Corporation.

Other product names mentioned in this document may be trademarks of their respective companies and are hereby acknowledged.

Mitel Technical Configuration Notes – Configure MiVoice Business for use with Time Warner Cable Business Class SIP Trunking

March 2015 15-4940-00363

Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the MiVoice Business to connect to Time Warner Cable Business Class (TWCBC) 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-Feb-2015	Initial Interop with MiVoice Business Release 7.0 SP1 PR1 Software Load 13.0.1.28 and TWCBC SIP Trunking

Interop Status

The Interop of TWCBC 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 TWCBC SIP Trunking achieved is:

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

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 7.0 SP1 PR1 Active Software Load 13.0.1.28
Mitel	Minet Sets:5320, 5360, 5312,	6.02.00.06
Mitel	MiVoice Border Gateway – Teleworker	8.1.23
Service Provider	TWCBC	NA

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 TWCBC 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 TWCBC 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 TWCBC and their PSTN gateway to and from Teleworker extensions.	☑
Video	Making and receiving a call through TWCBC 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 TWCBC SIP Trunking is connected to the MiVoice Business.

Feature	Problem Description
Packetization Rate	TWCBC supports 20ms packetization rate only **Recommendation:** Set packetization rate to 20ms
Ring back not heard after Mitel unsupervised transfer completion	Mitel phone A completes unsupervised transfer to connect Remote DUT/DUT PSTN to another Mitel phone B. Remote DUT/DUT PSTN continue hears music while Mitel phone B is ringing, this is not Mitel test plan expected result. **Recommendation**: Contact Mitel Support
ACD Interflow/Overflow to external loop detection	When setup ACD interflow/overflow to a Mitel answer point through TWCBC network, TWCBC's loop detection feature will reject the call when Mitel answer point picks up the call. **Recommendation**: configure the proper ACD Interflow/overflow target to avoid using more than 2 SIP trunks for one call.
Video Call	TWCBC does not Support video calls. **Recommendation**: Contact TWCBC for update on this feature
Voice Codec	TWCBC supports only G711Ulaw for Voice codec, **Recommendation:** Configure Intra-zone Compression to No and allow only G711 codecs
FAX T.38	TWCBC does not support T.38 fax, only G711 pass-through fax is supported **Recommendation:** configure Fax using G711 Pass-through only
Split (Mid-Call Feature) via single digit FAC from EHDU in conference call	External Hot Desk User (EHDU) could not invoke call split via single digit FAC while in the conference call, all other Mid-Call Features such as Hold/Retrieve, Transfer, Conference, Swap, Handoff work as expected. **Recommendation**: Contact Mitel Support

Network Topology

This diagram shows how the testing network is configured for reference.

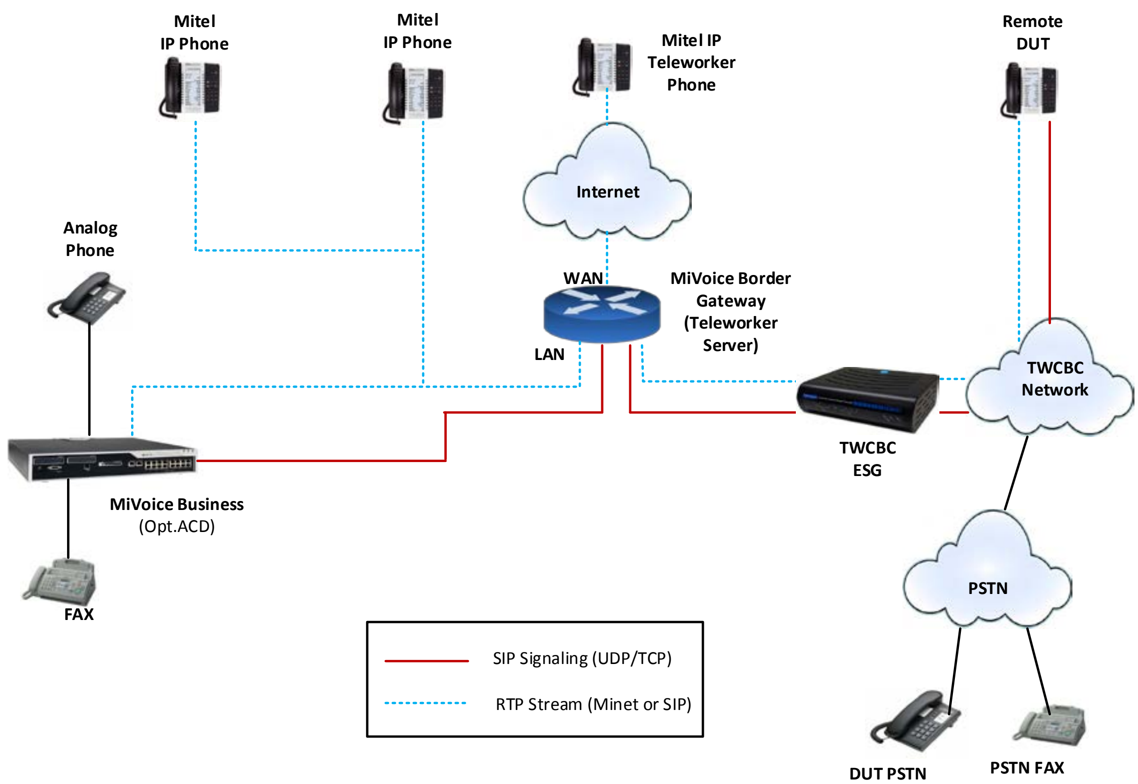


Figure 1 – Network Topology

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 TWCBC 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 TWCBC 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

Navigation: Licenses -> Licenses and Option Selection

Ensure that the MiVoice Business is equipped with enough SIP trunk licenses for the connection to TWCBC 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: this is set to **YES** for NuPoint Voice Mail configuration.

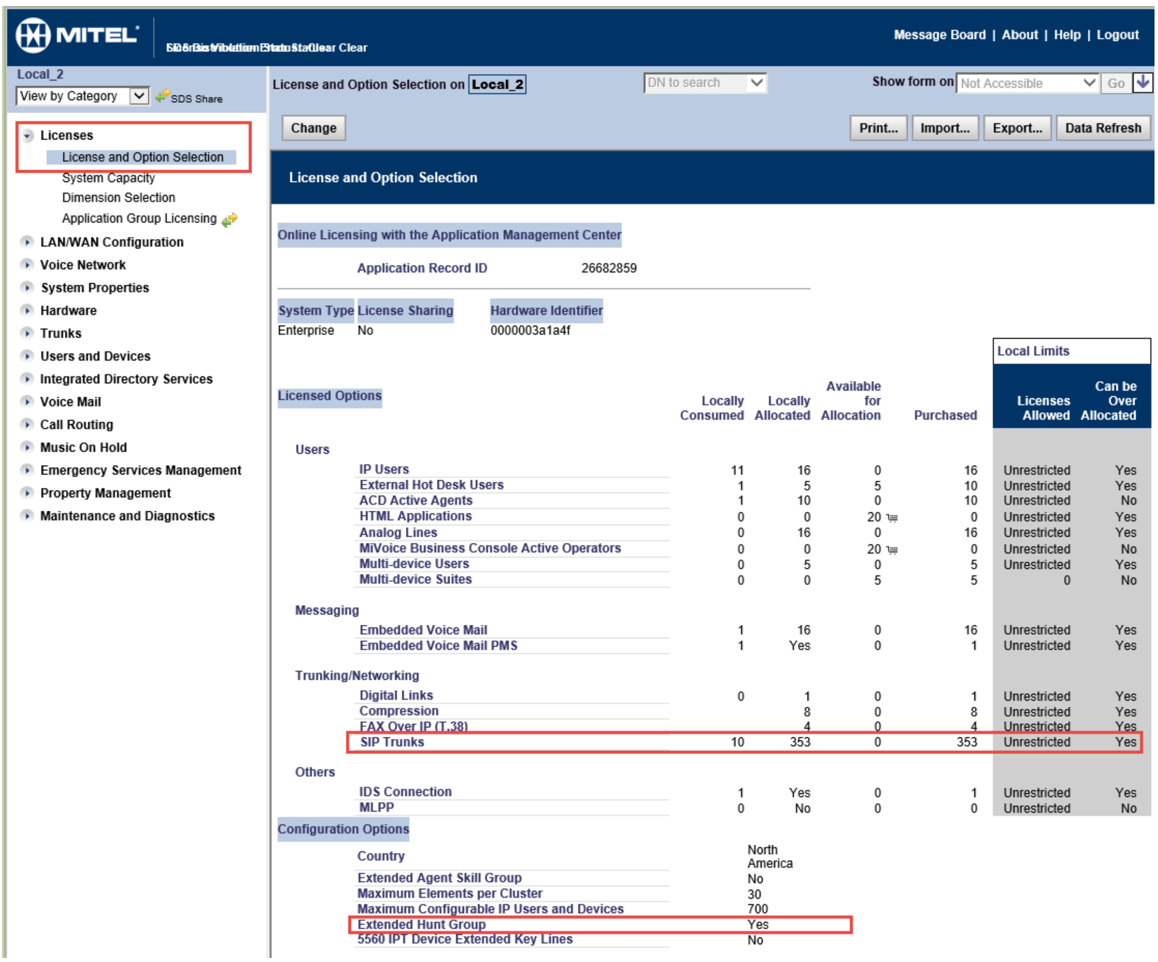


Figure 2 – License and Option Selection

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.

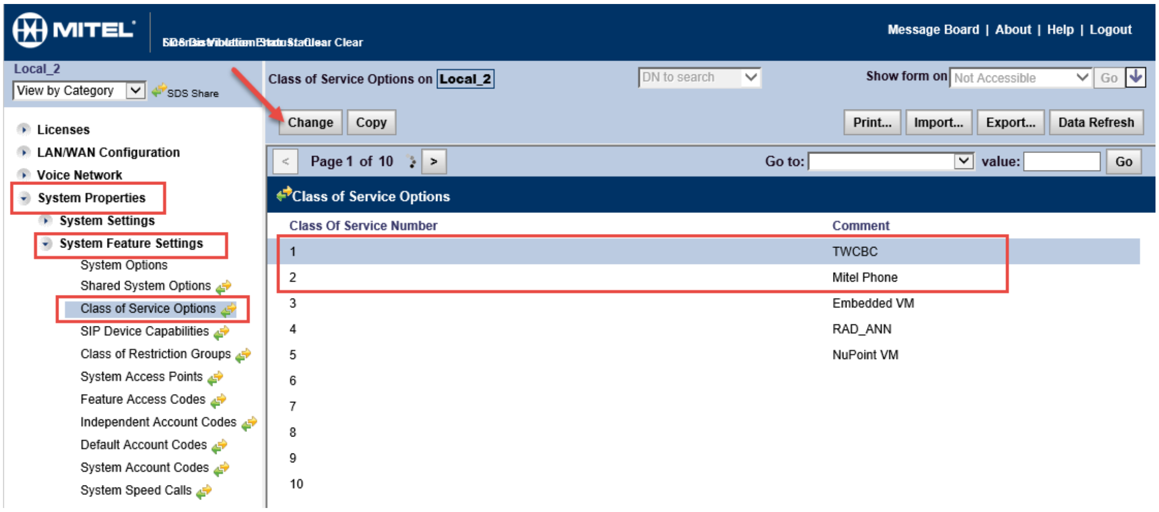


Figure 3 – Class of Service

Class of Service for Trunk

General

General | Advanced

Class Of Service Number	1
Comment	TWCBC
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
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

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

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
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	Yes
Campon	
Auto Campon Timer	
Campon Recall Timer	10

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

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 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	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	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 Adiustment Saved	No

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

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
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	No

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	Mitel Phone
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
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

Figure 10 – Class of Service (Basic) for Phone

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
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	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	10
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No

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

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
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

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

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
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 TWCBC SIP Trunking. In this example, the softswitch is reachable by an IP Address and is defined as “TWCBC” 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.

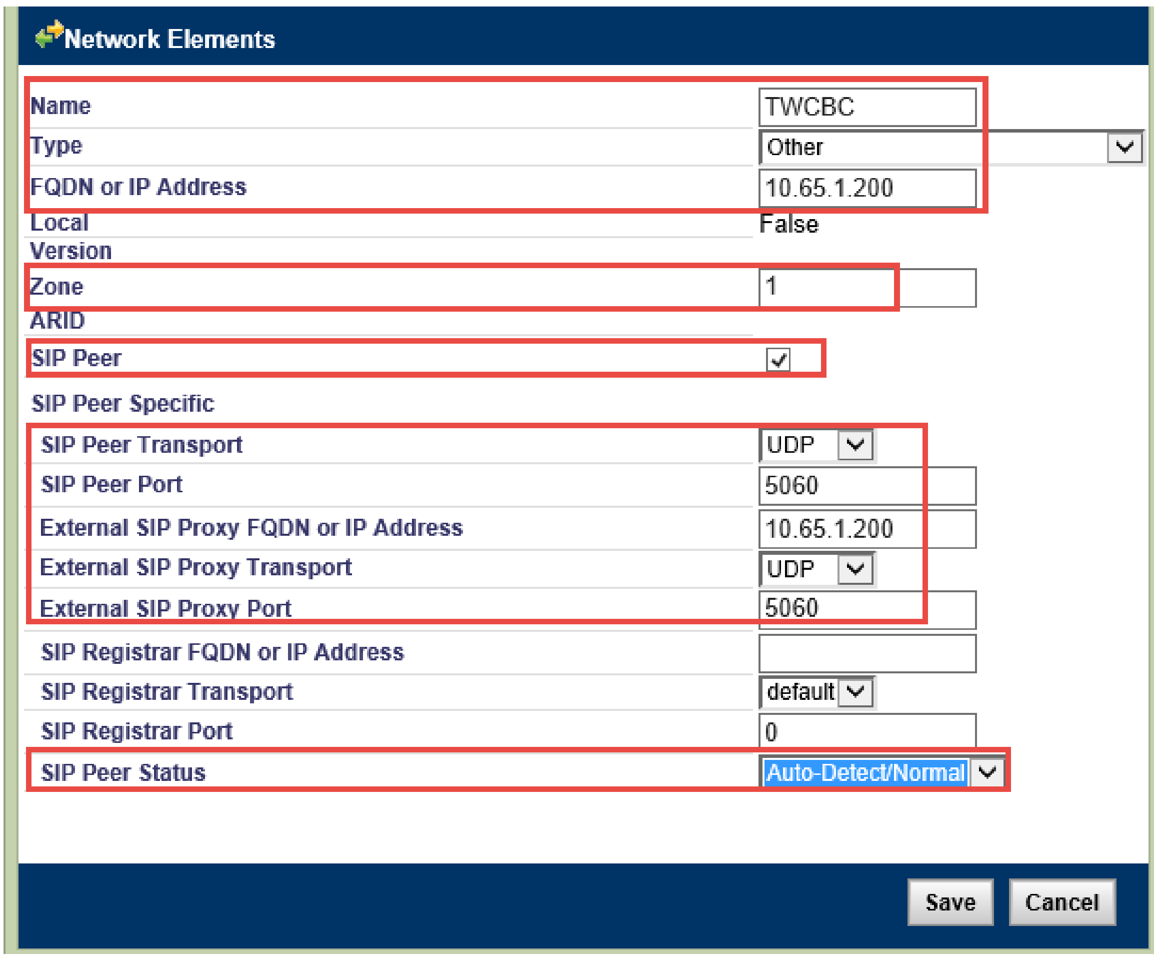


Figure 16 – Network Element Assignment

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).

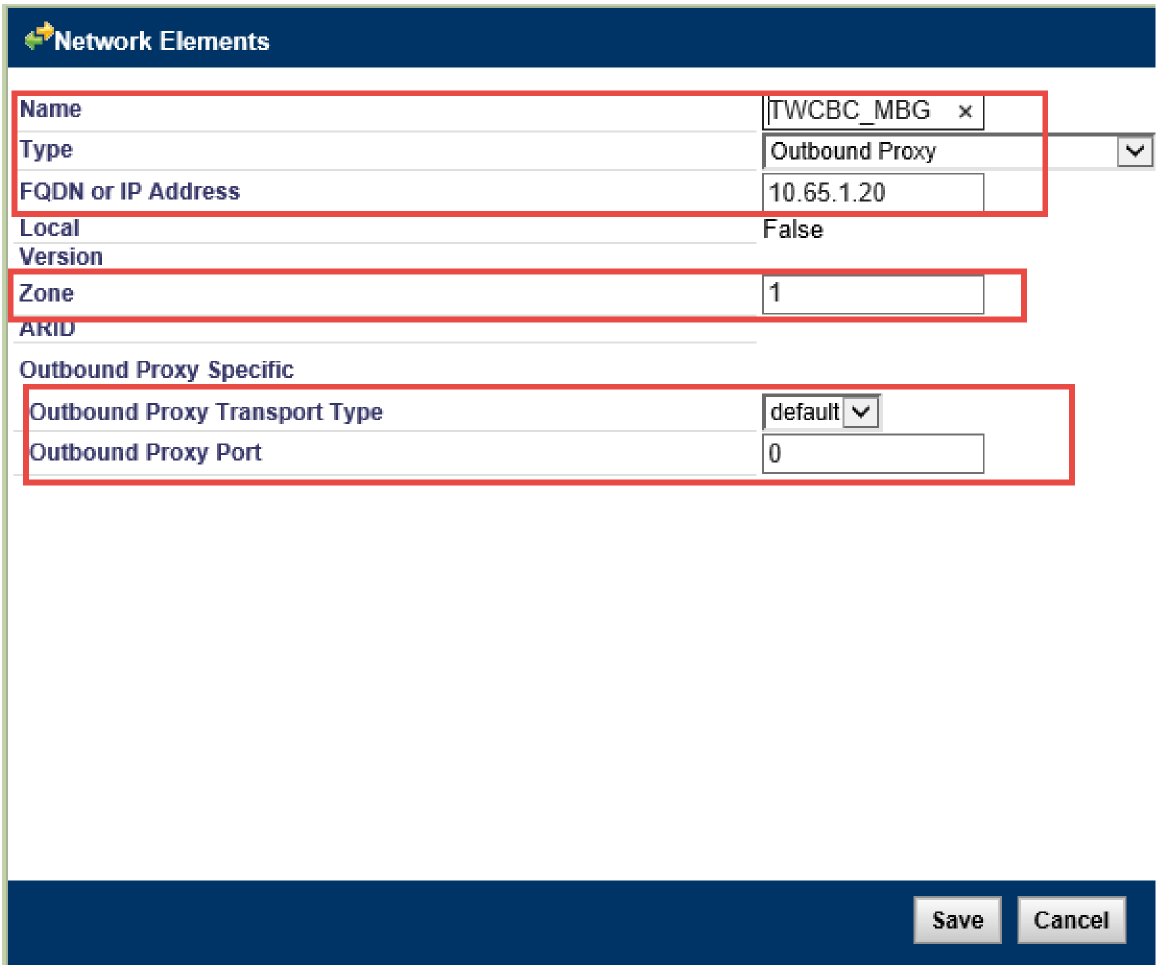


Figure 17 – Network Element Assignment (Proxy)

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.

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. **Dial in Trunks Incoming Digit Modification- Absorb** is set to 0.

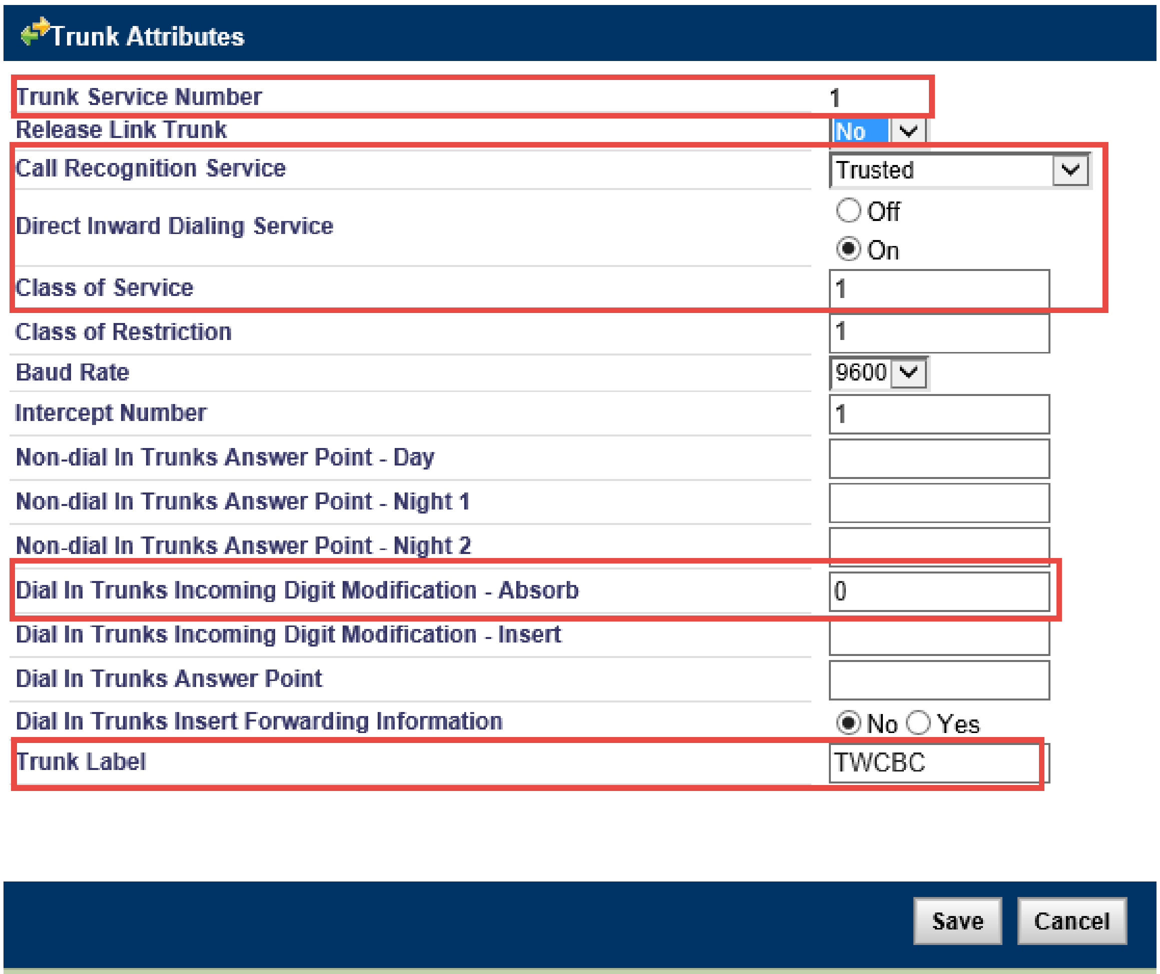


Figure 18 – Trunk Attributes

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 (Figure 19):

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

Registration User Name: Leave this field blank.

Address Type: Select IP address of your Mitel 3300ICP.

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

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

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).

Trunk Service: Enter the trunk service number that was previously configured**- 1** is used in this configuration.

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 screen capture below.

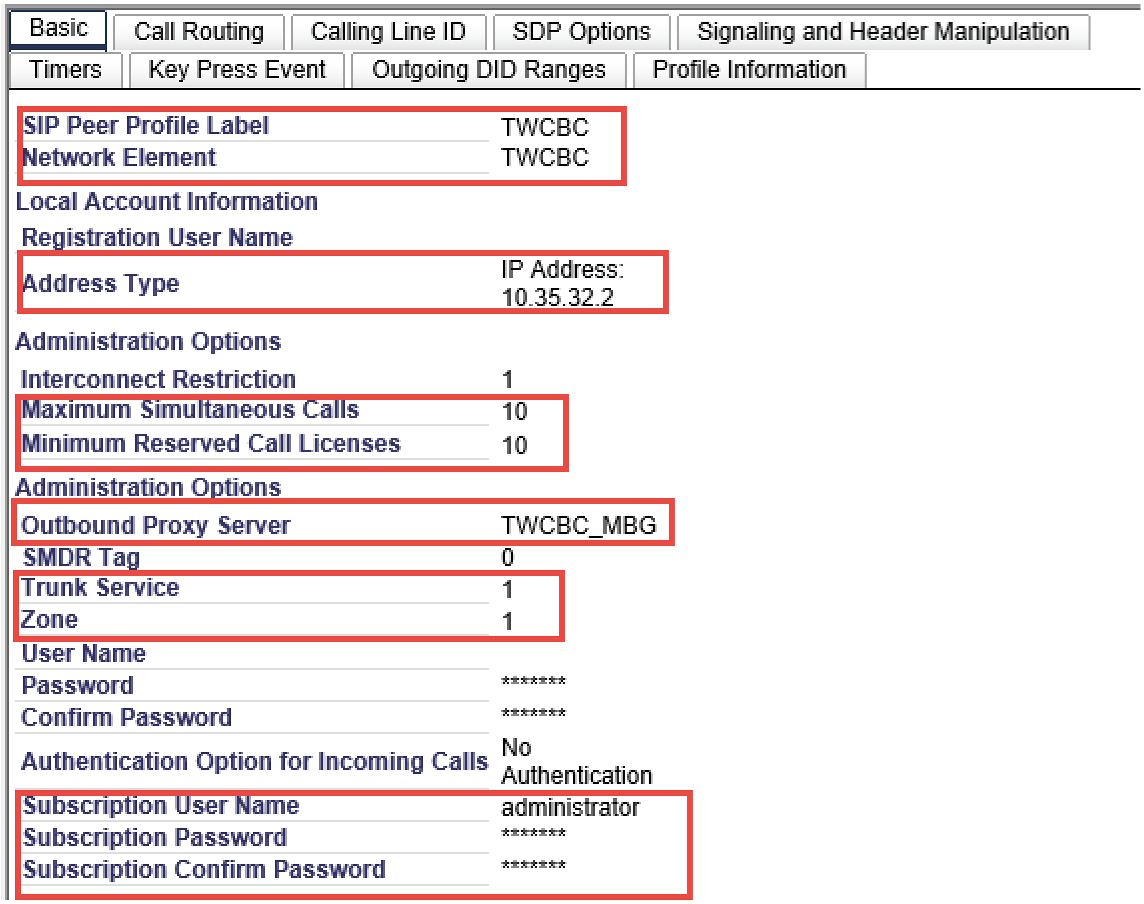


Figure 19 – SIP Peer Profile - Basic

Call Routing (figure 20):

Leave the default settings as shown.

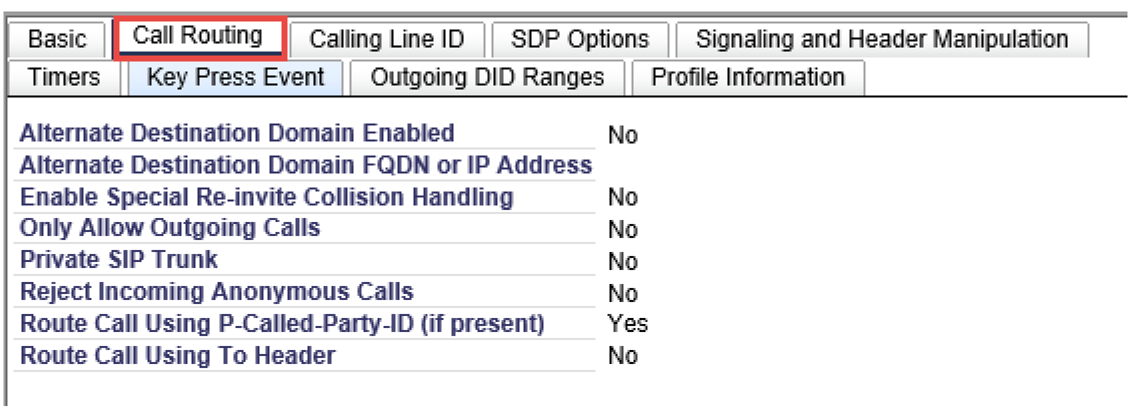


Figure 20 – SIP Peer Profile Assignment- Call Routing

Calling Line ID (figure 21):

Default CPN: Default CPN (Calling Party Number) is applied to all outgoing calls. TWCBC accepts the calls from all assigned DID numbers hence this field is left blank.

CPN Restriction: By default, this parameter is set to “NO” to not block the caller ID.

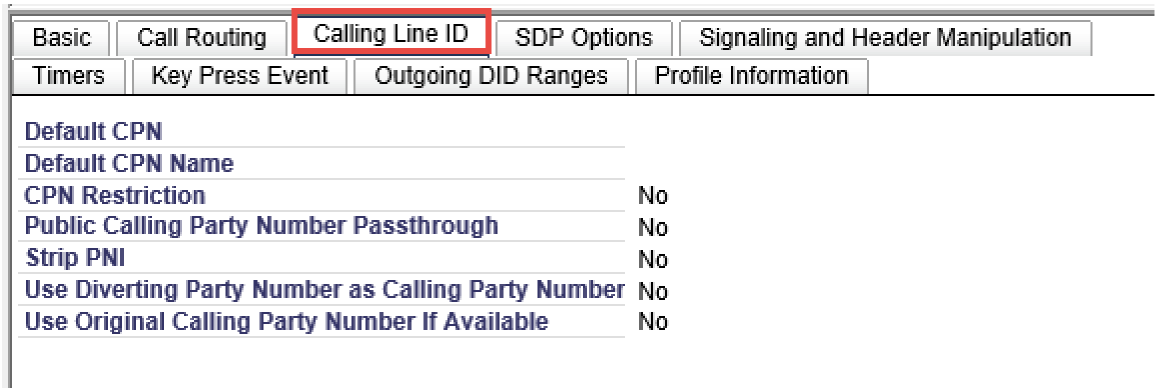


Figure 21 – SIP Peer Profile Assignment- Calling Line ID

SDP Options (figure 22):

Force sending SDP in initial Invite message: **Yes** is selected for this configuration.

Leave all other fields as default.

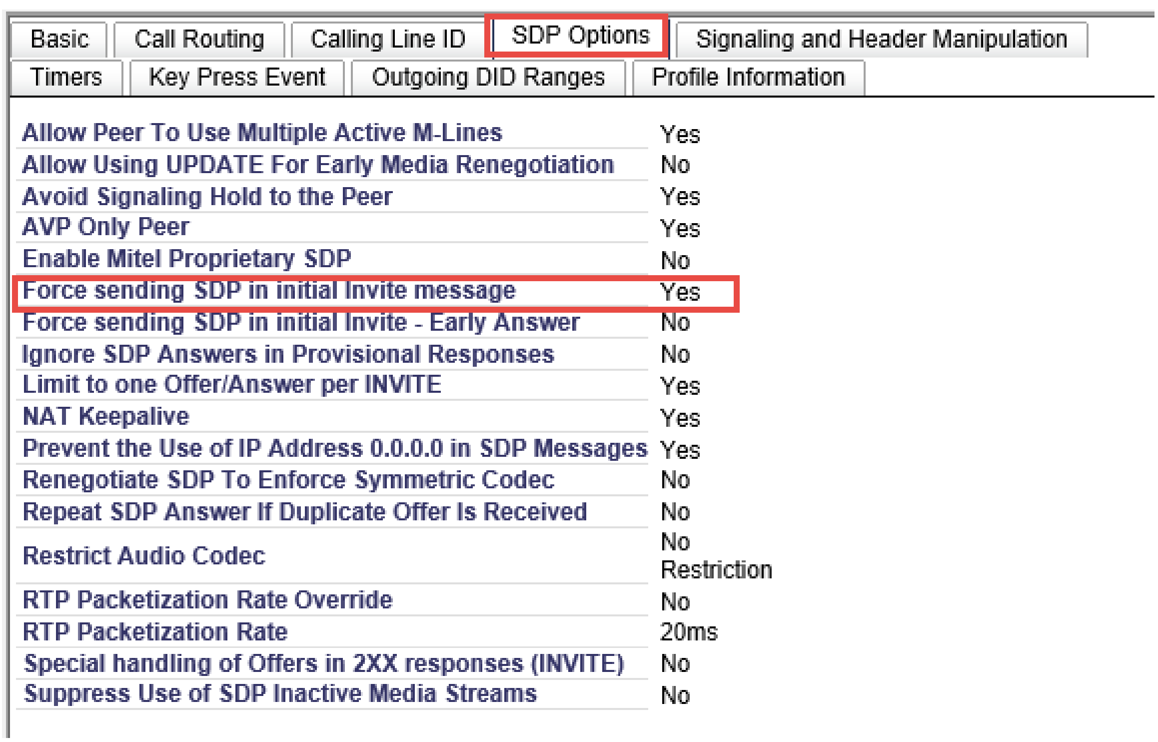


Figure 22 – SIP Peer Profile Assignment- SDP Options

Signaling and Header Manipulation (figure 23):

Disable Reliable Provisional Response: set to **YES** for this setup.

Require Reliable Provisional Response on Outgoing Calls: Set to **No** for this setup.

Leave all other fields as default.

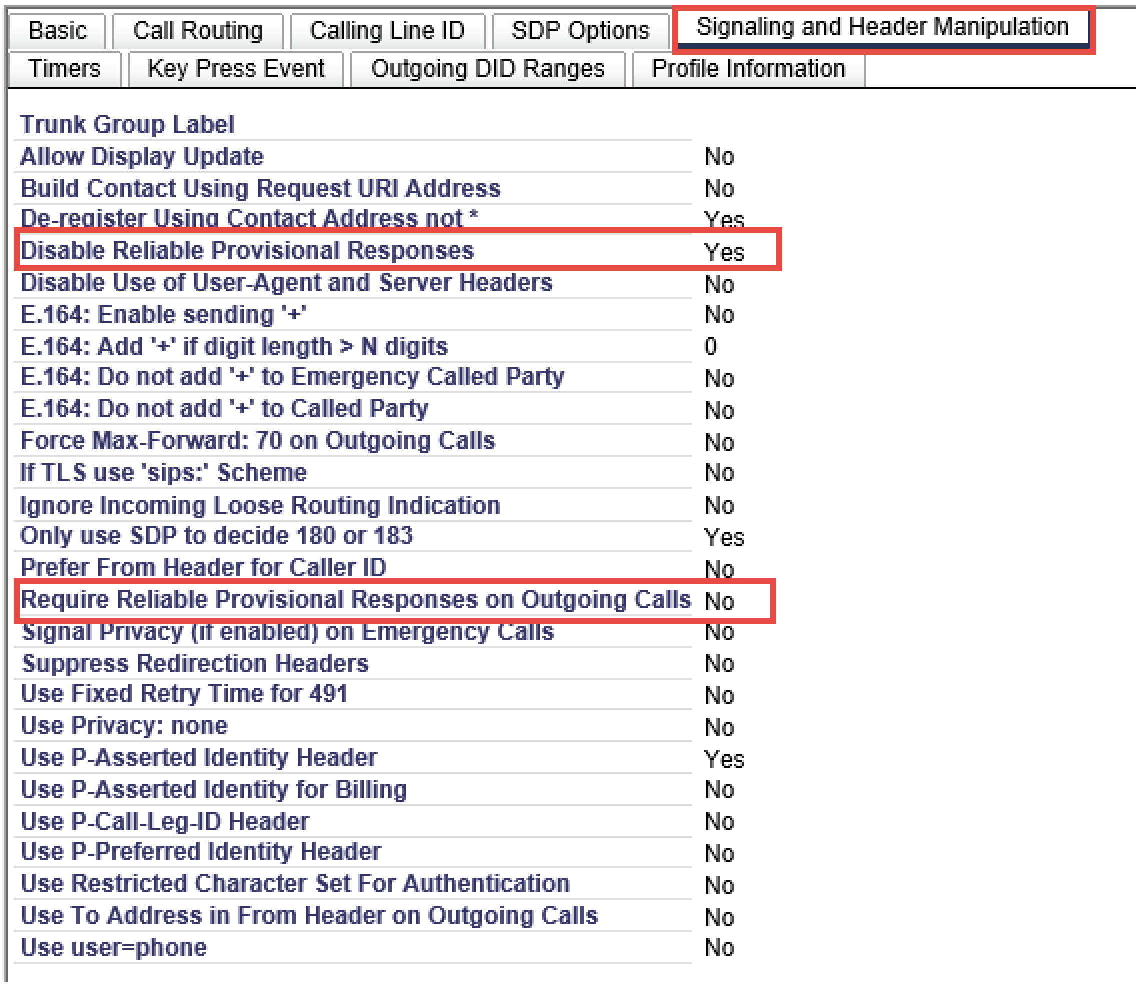


Figure 23 – SIP Peer Profile Assignment- Signaling and Header Manipulation

Timers (figure 24):

Session Timer: **120 seconds** is used in this configuration, set this value to 0 will disable the session audit.

Leave all other fields as default.

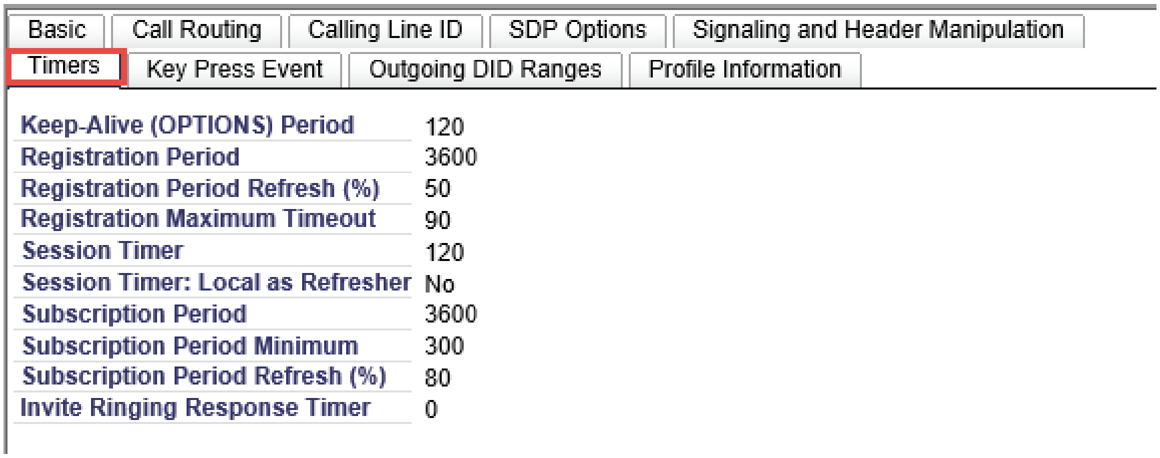


Figure 24 – SIP Peer Profile Assignment- Timers

Key Press Event (figure 25):

Set **Yes** for **Allow Inc Subscription for Local Digit Monitoring** and **Request Outbound Proxy to Handle Out Subscriptions**.

Set **KPML Transport** to **UDP**.

Set **KPML Port** to **5060**

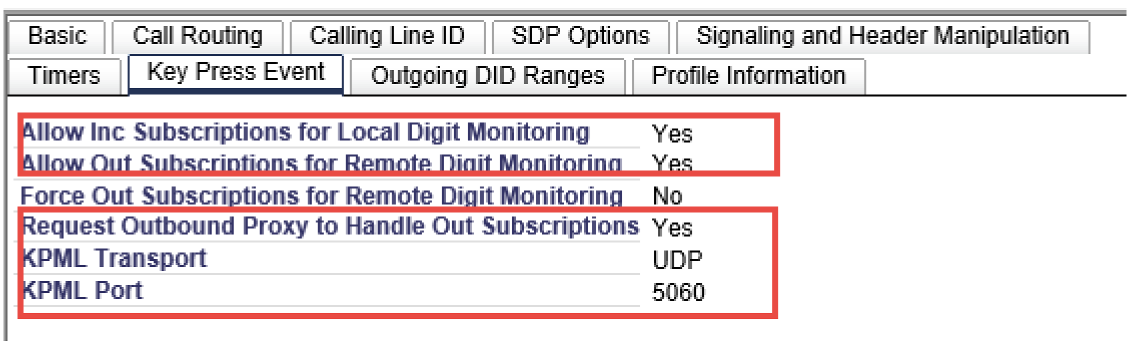


Figure 25 – SIP Peer Profile Assignment- Outgoing DID Ranges

Outgoing DID Range (figure 26) and Profile Information (figure 27):

Leave those two sections as is.

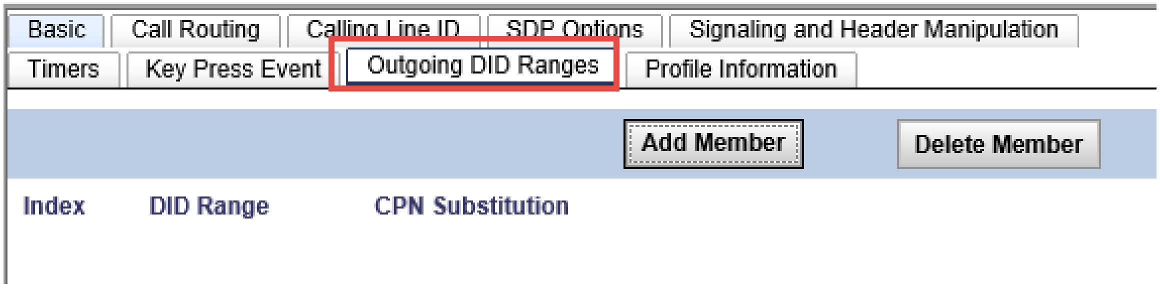


Figure 26 – SIP Peer Profile Assignment- Outgoing DID Ranges

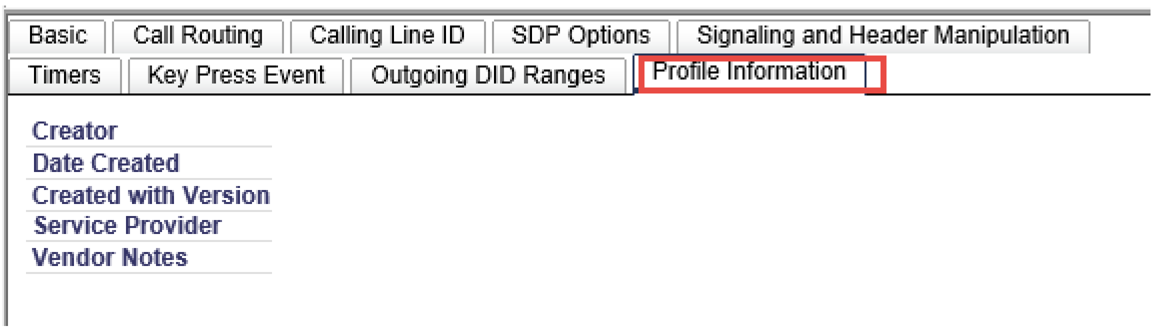


Figure 27 – SIP Peer Profile Assignment – Profile Information

ARS Digit Modification Plans

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

Ensure that Digit Modification for outgoing calls on the SIP trunk to TWCBC 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).

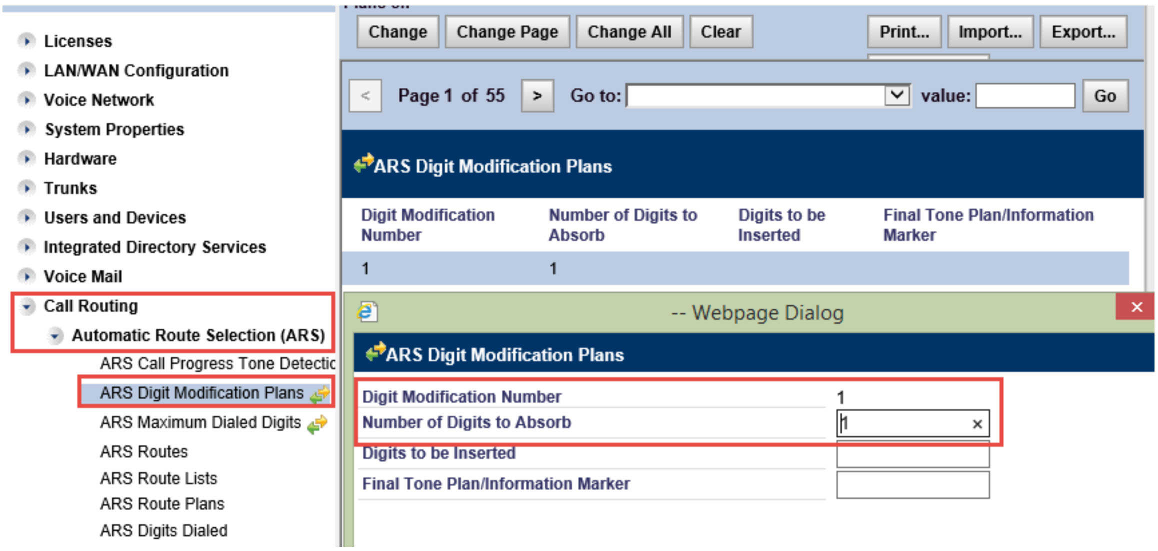


Figure 28 – Digit Modification Assignment

ARS Routes

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

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

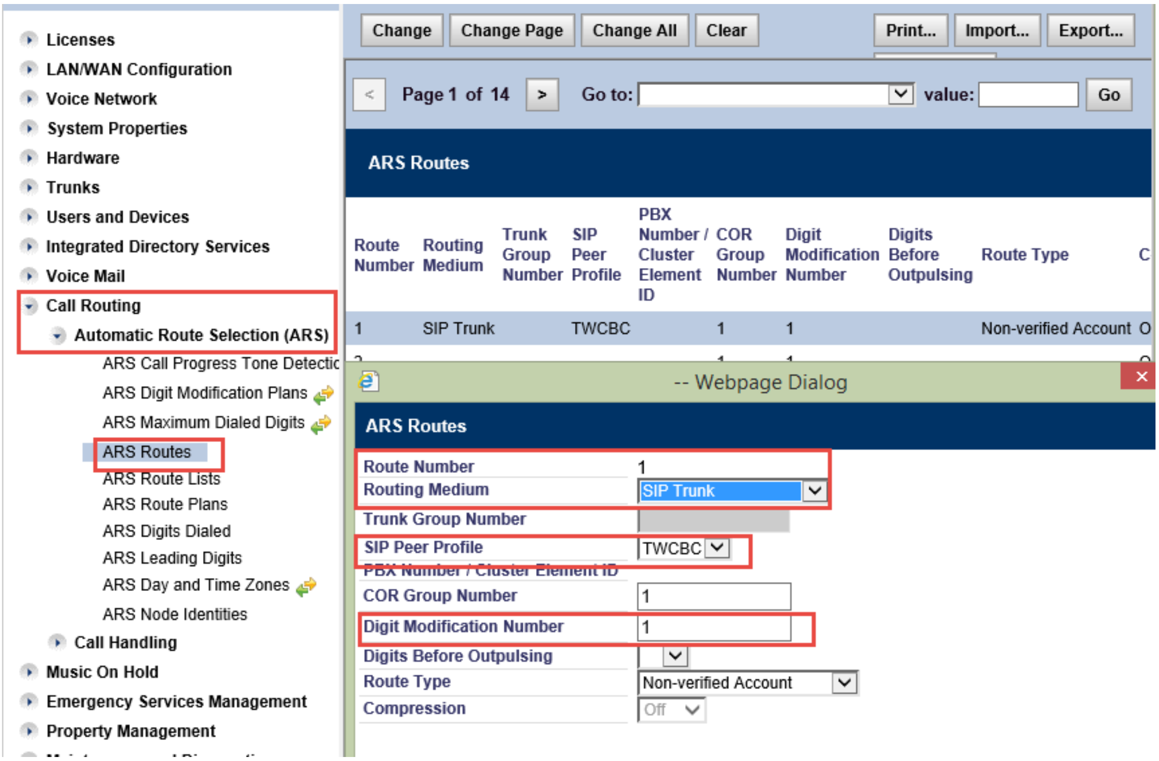


Figure 29 – SIP Trunk Route Assignment

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 TWCBC via route 1 configured in previous step.

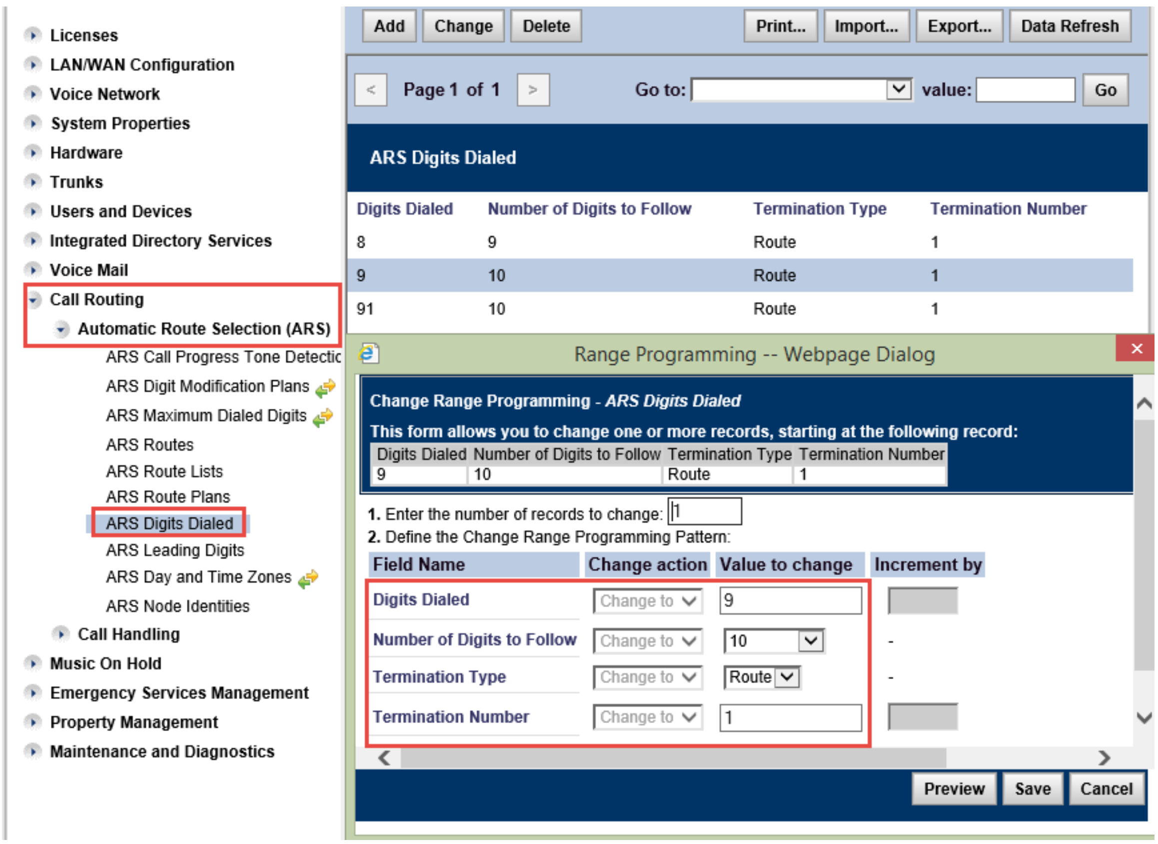


Figure 30 – ARS Digit Dialed Assignment

FAX Configuration

Navigation: **Voice Network** -> **Fax Service Profiles**

TWCBC 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).

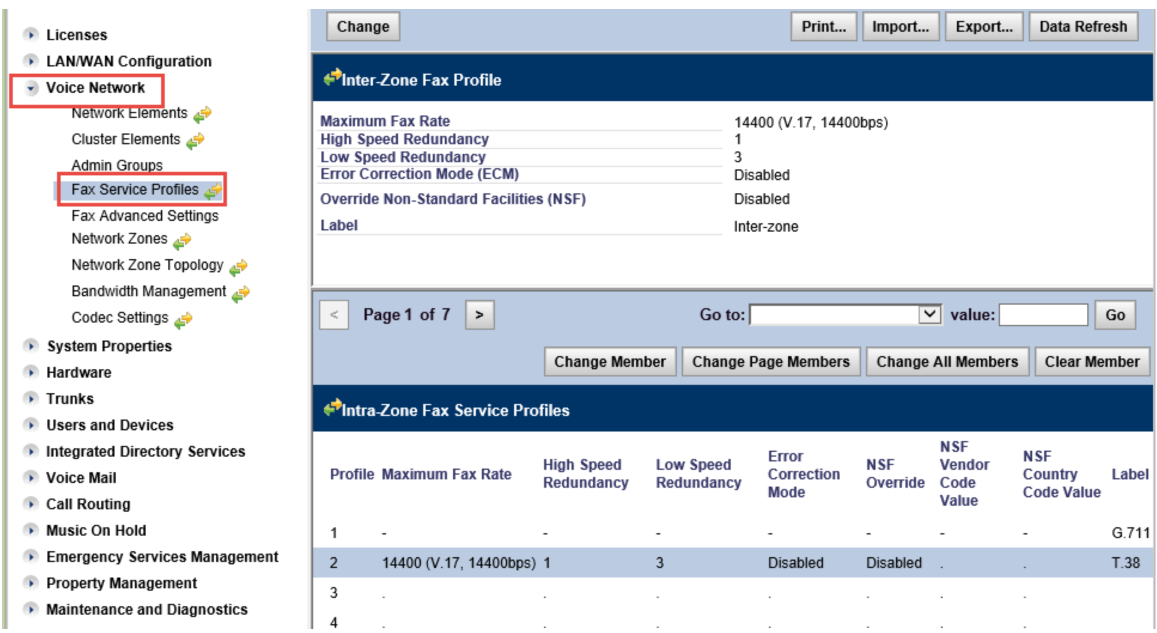


Figure 31 – Fax Configuration

Zone Assignment

Navigation: **Voice Network** -> **Network Zone**

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”. TWCBC only supports G711 codec for voice, hence **Intra-zone Compression** is set to **NO** for this configuration.

TWCBC uses the default **Intra-zone FAX Profile 1** as it only support G711Ulaw pass-through fax.

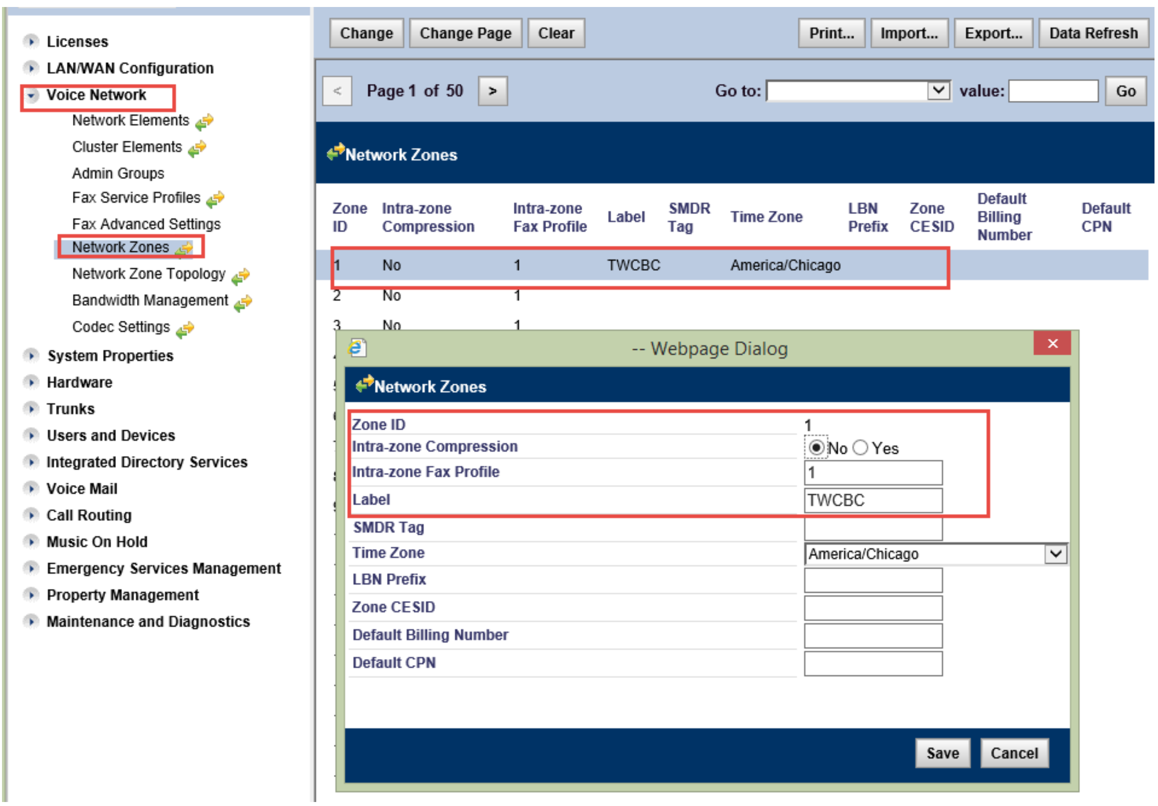


Figure 32 – Zone Assignment

Personal Ring Groups Configuration

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

Mitel phone extension 1029 and an EHDU (External Hot Desk User) 2030 are added as members of Personal Ring Group. EHDU 2030 targets an external PSTN number

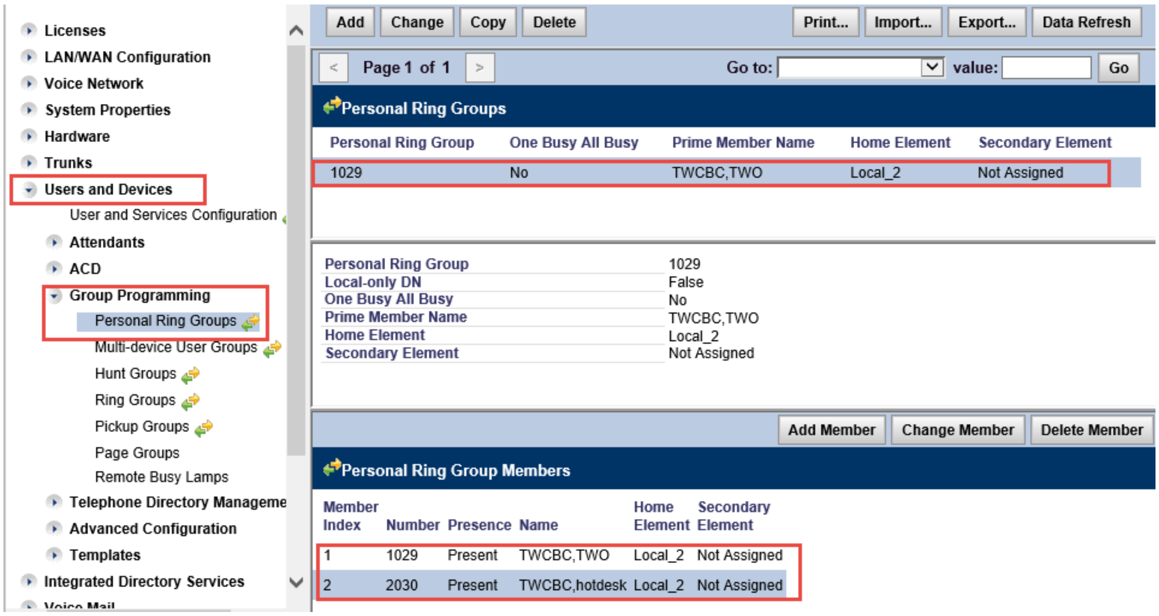


Figure 33 – Personal Ring Groups

Miltiline IP sets 1029 and 2030 are configured as following.

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

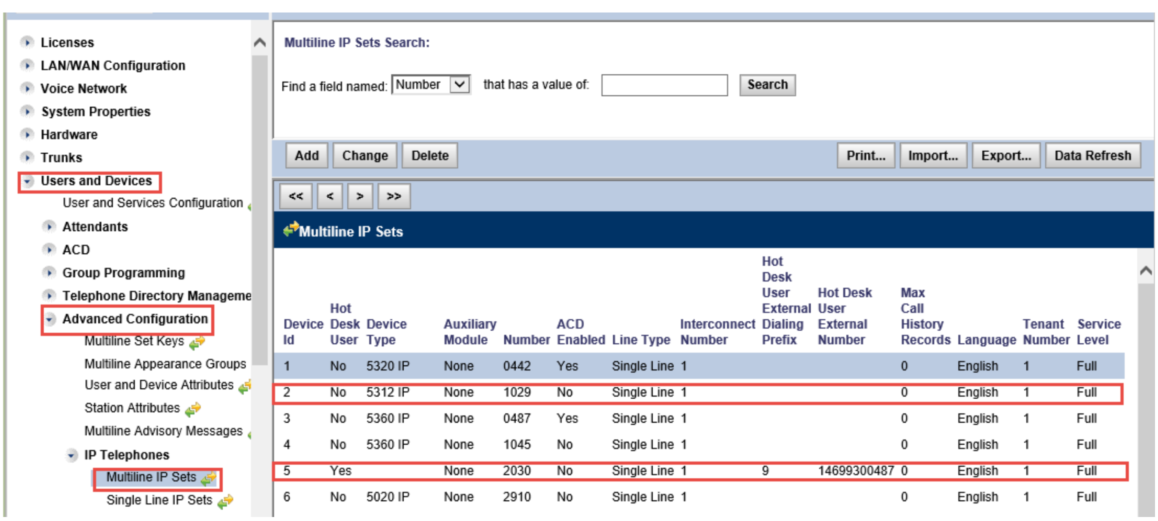
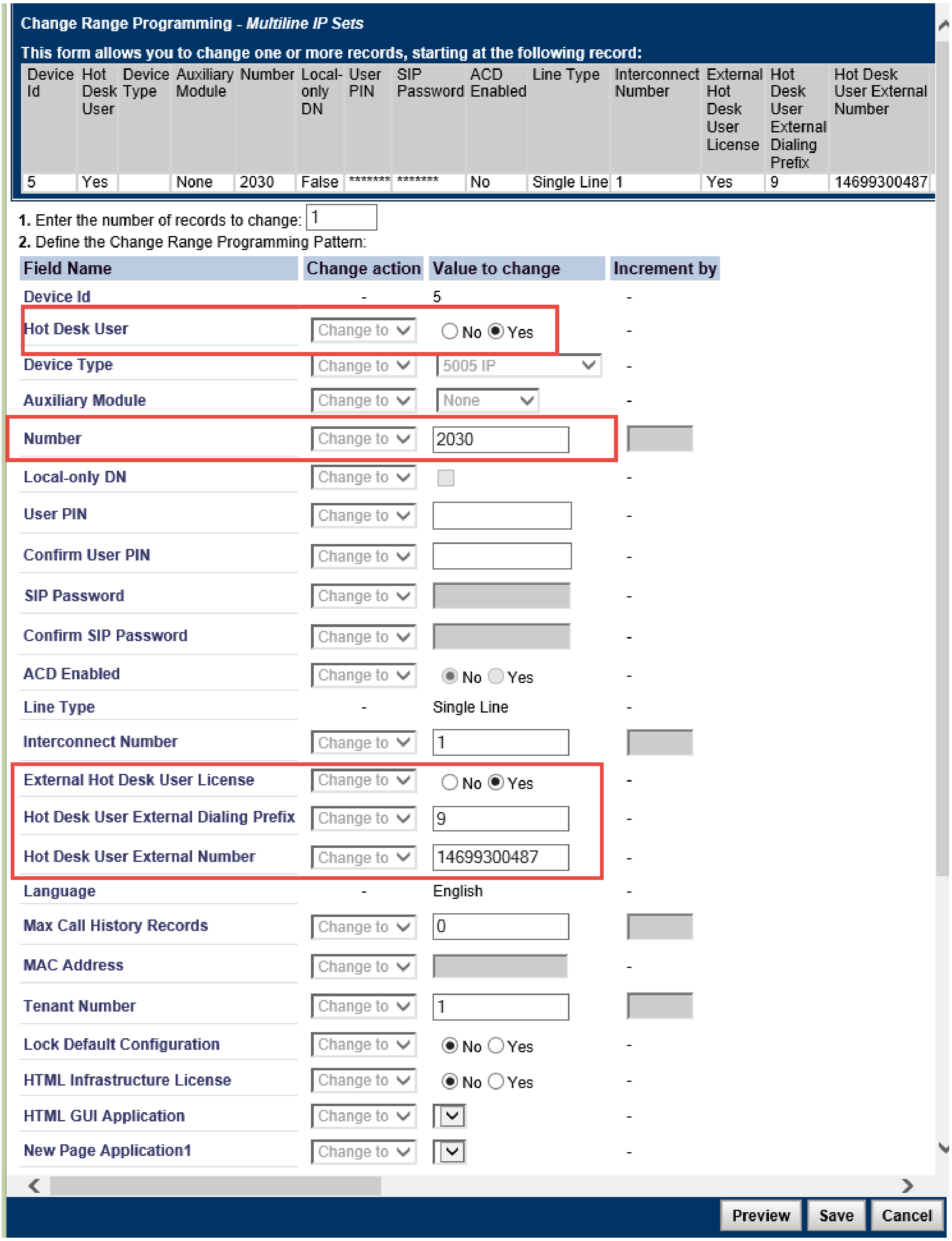


Figure 34 – Multiline IP Sets



Change Range Programming - *Multiline IP Sets*

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

Device Id	Hot Desk User	Device Type	Auxiliary Module	Number	Local-only DN	User PIN	SIP Password	ACD Enabled	Line Type	Interconnect Number	External Hot Desk User License	Hot Desk User External Dialing Prefix	Hot Desk User External Number
5	Yes		None	2030	False	*******	*******	No	Single Line	1	Yes	9	14699300487

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Device Id	-	5	-
Hot Desk User	Change to	No / Yes (selected)	-
Device Type	Change to	5005 IP	-
Auxiliary Module	Change to	None	-
Number	Change to	2030	
Local-only DN	Change to		-
User PIN	Change to		-
Confirm User PIN	Change to		-
SIP Password	Change to		-
Confirm SIP Password	Change to		-
ACD Enabled	Change to	No (selected) / Yes	-
Line Type	-	Single Line	-
Interconnect Number	Change to	1	
External Hot Desk User License	Change to	No / Yes (selected)	-
Hot Desk User External Dialing Prefix	Change to	9	-
Hot Desk User External Number	Change to	14699300487	-
Language	-	English	-
Max Call History Records	Change to	0	
MAC Address	Change to		-
Tenant Number	Change to	1	
Lock Default Configuration	Change to	No (selected) / Yes	-
HTML Infrastructure License	Change to	No (selected) / Yes	-
HTML GUI Application	Change to		-
New Page Application1	Change to		-

Preview Save Cancel

Figure 35 – Programming Multiline IP Sets

NuPoint Configuration

3300 Setup for Connecting NuPoint

Licensing and Option Selection – SIP Licensing

The first step in setting up the 3300 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 3300 are programmed as 5020 IP endpoints on the 3300. 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 3300. Set ******* for the **Registration Access Code** and ### for the **Replacement Access Code**.

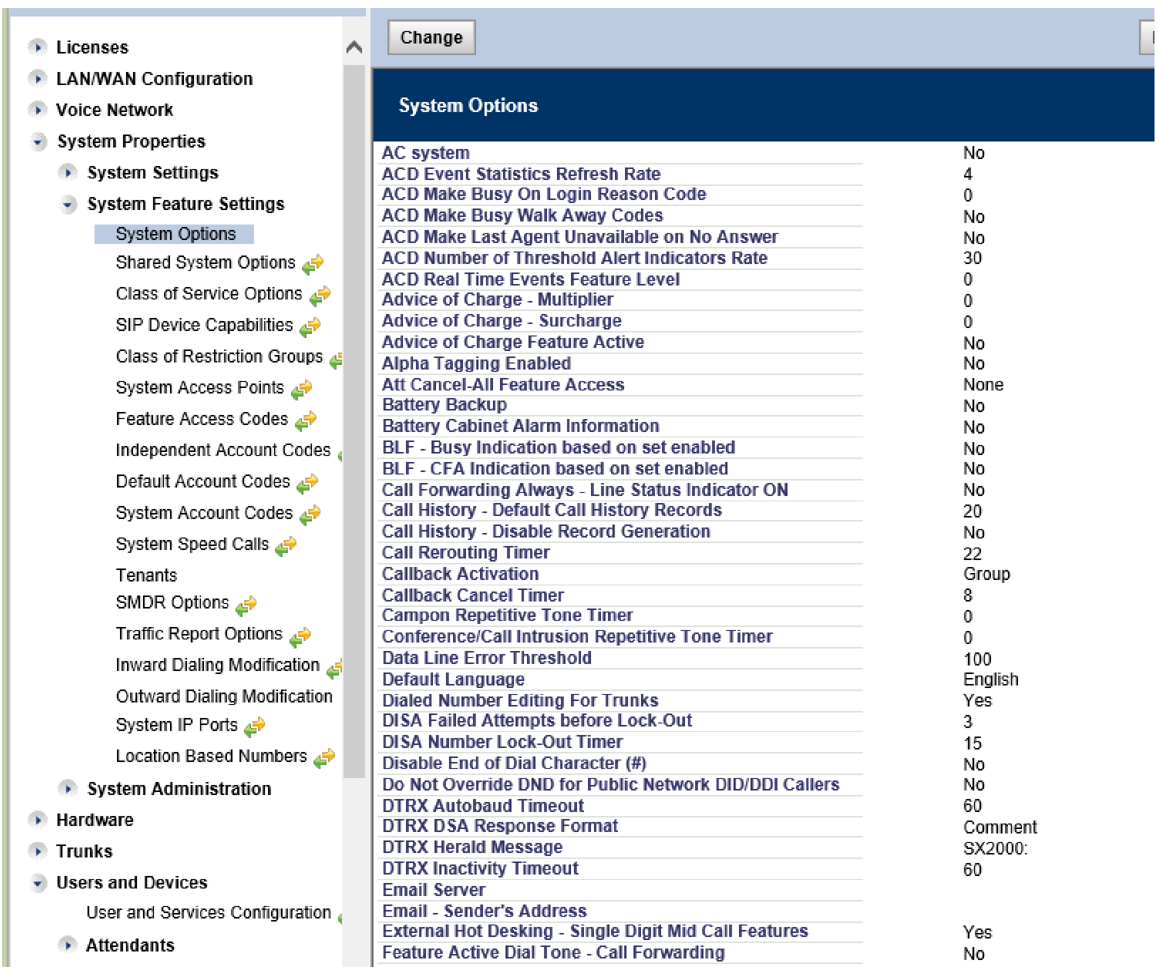


Figure 36 – System Option

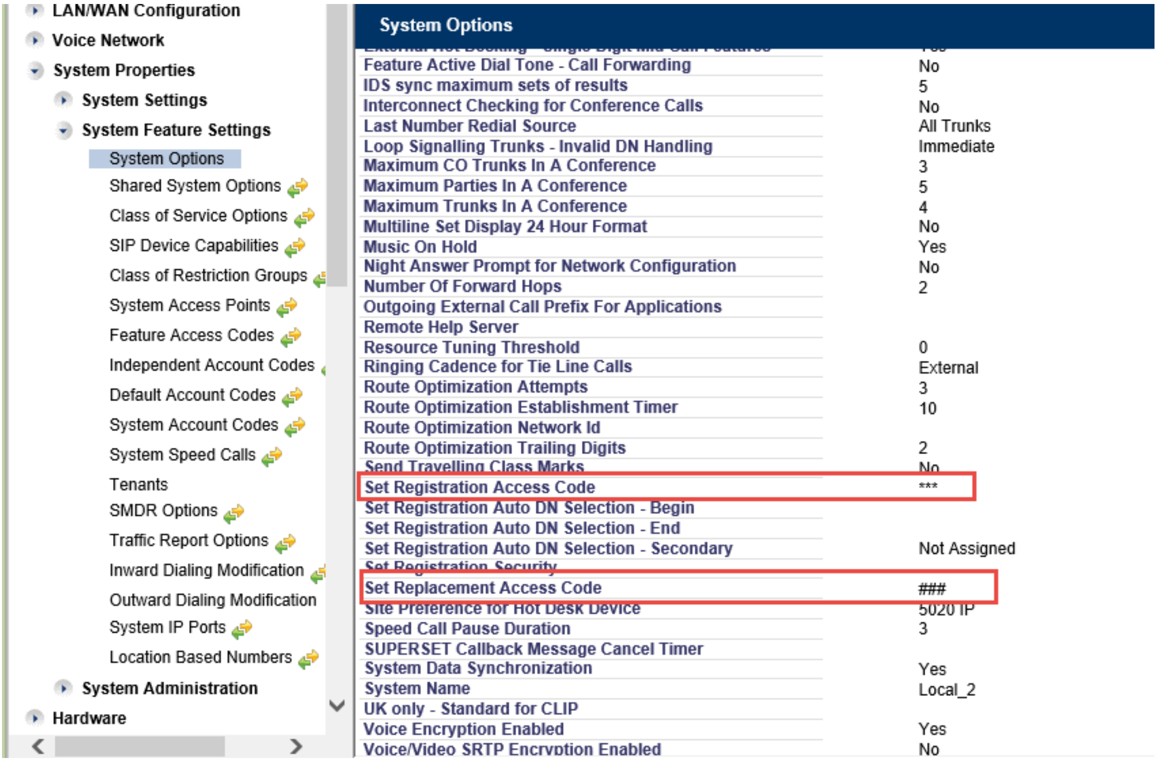


Figure 37 – System Options cont.

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 Voice Mail.

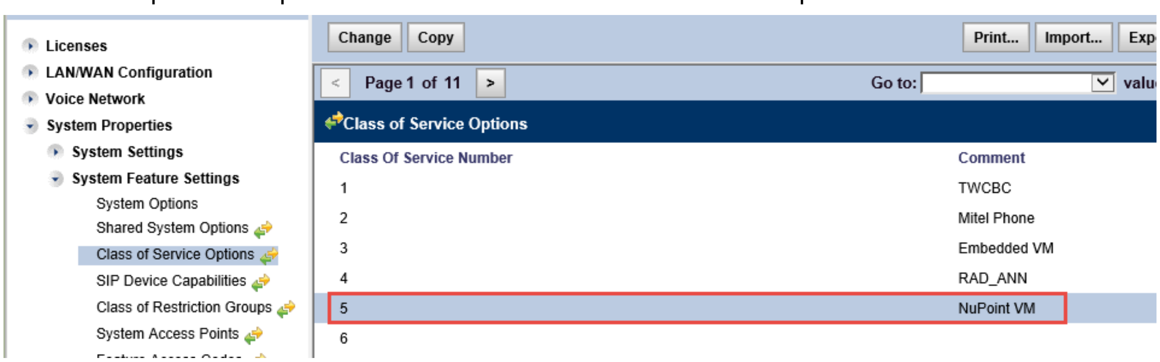


Figure 38 – Class of Service Option for NuPoint Voice Ports

In Class of Service for NuPoint Voice Mail enable the following:

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

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 Number 2910~2913 are configured as NuPoint Voice Ports for this test.

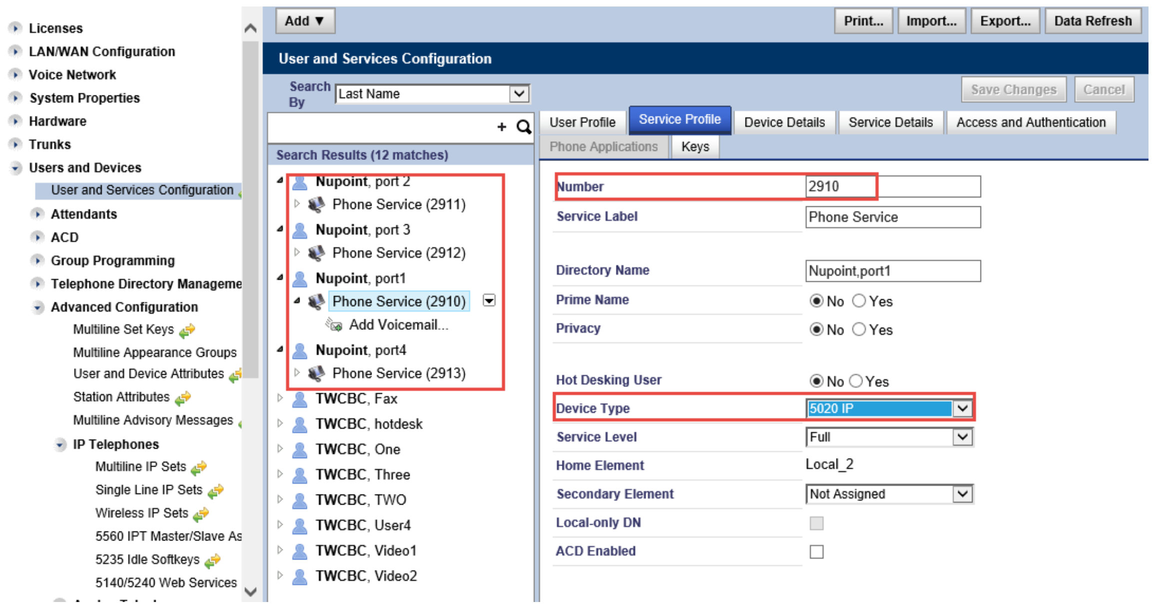


Figure 39 – IP Endpoints configuration

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.

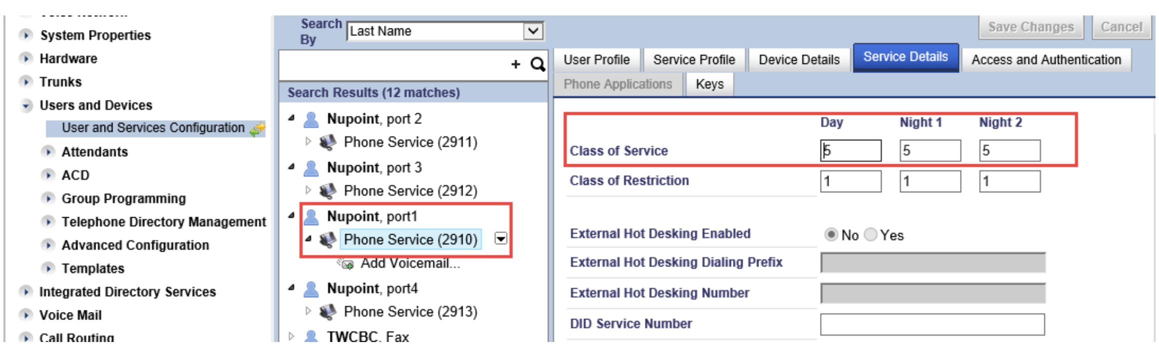


Figure 40 – IP Endpoints Class of Service

Voice Mail Hunt Group

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

Create Voice Mail hunt group that will be used to call voice mail. 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 Voice Mail and change the Hunt Group type to Voice Mail. Here hunt group 2900 is created.

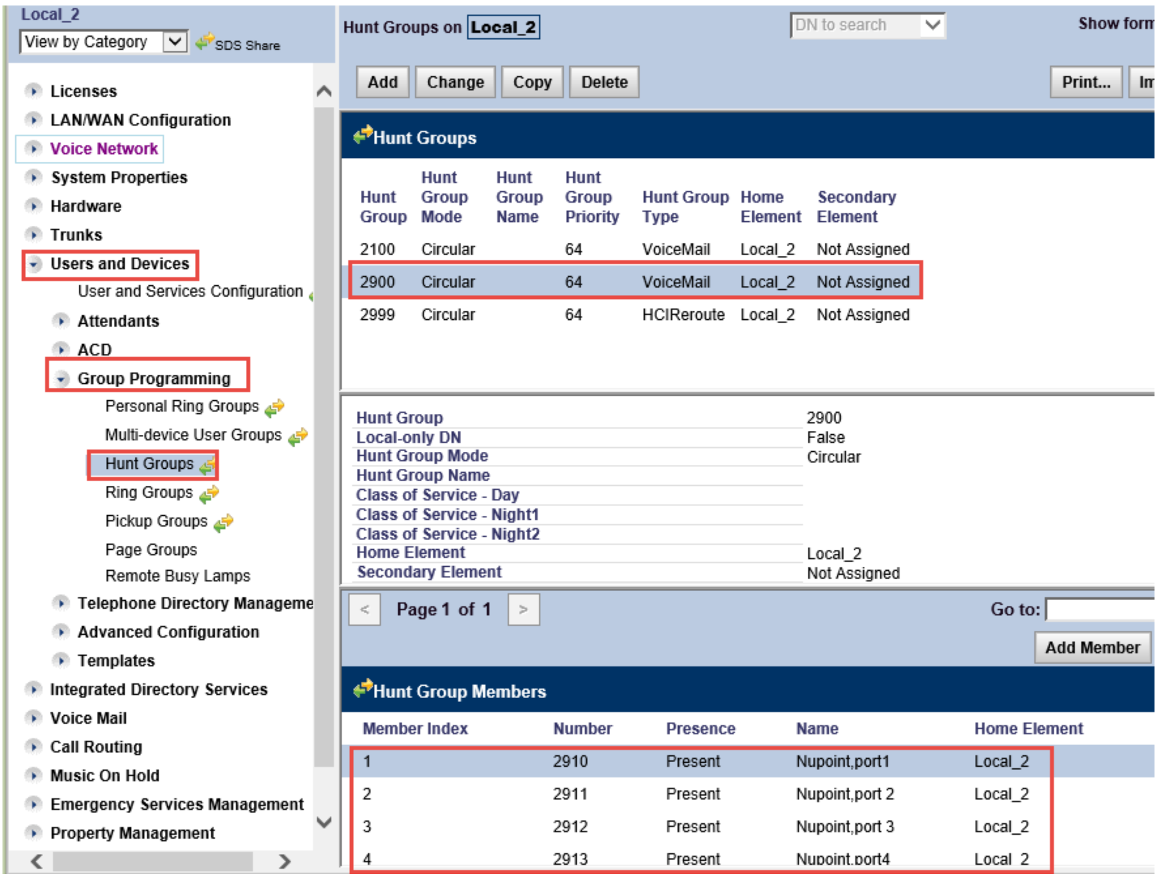


Figure 41 – Voicemail Hunt Group Configuration

HCIReroute Hunt Group

Program the HCIReroute Hunt Group and set it to always route to the NuPoint Voice Mail Hunt Group. The primary reason for setting up a 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 Voice Mail Hunt Group.

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

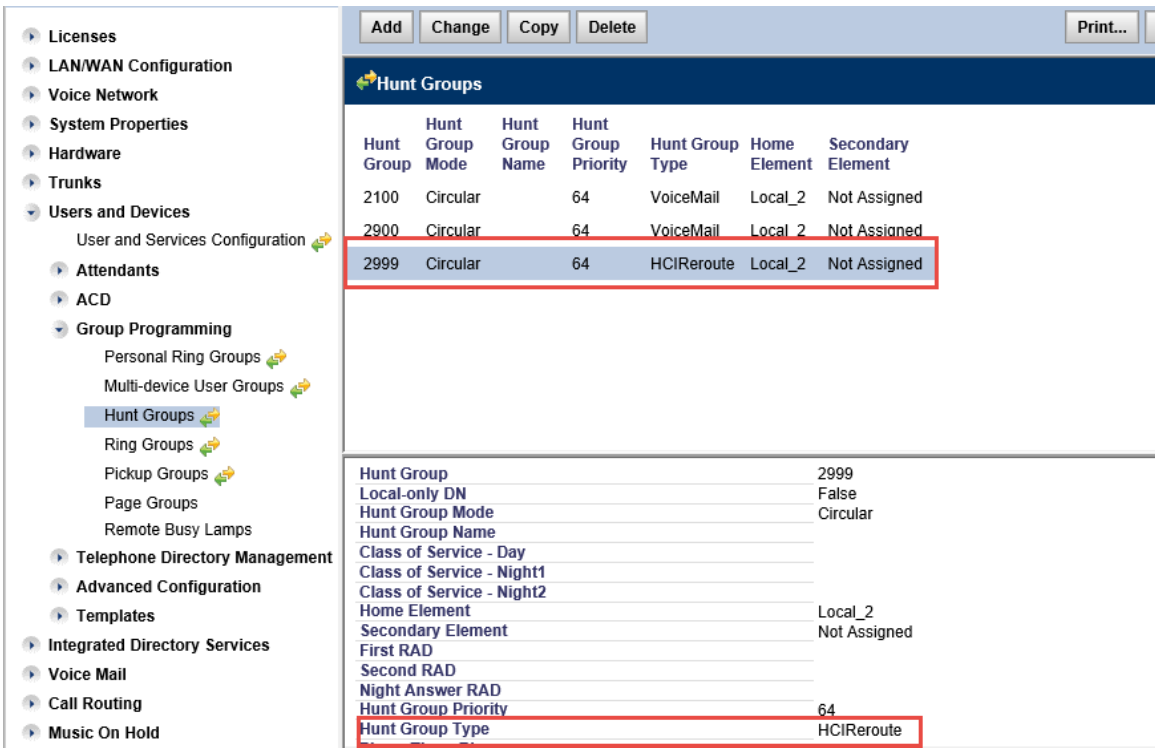


Figure 42 – HCIReroute Hunt Group

Navigation: **Call Routing** -> **Call Handling** -> **Call Rerouting Always Alternative**

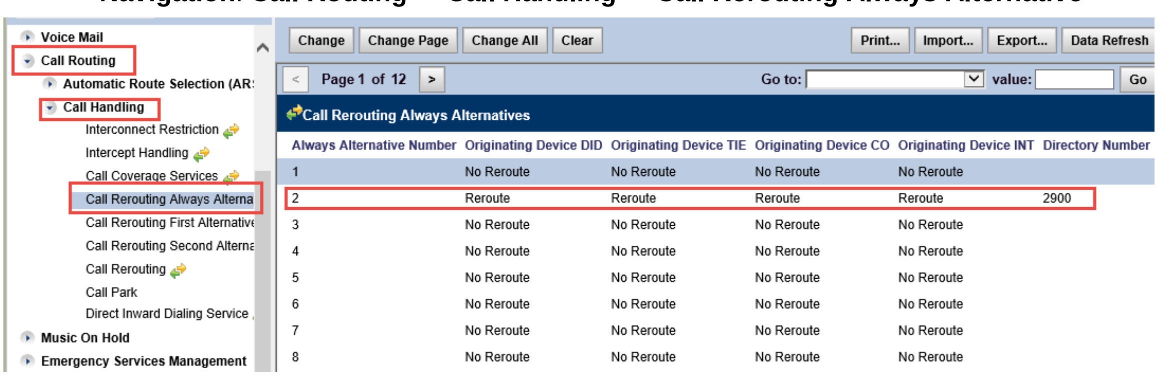


Figure 43 – Call Rerouting Always Alternative

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

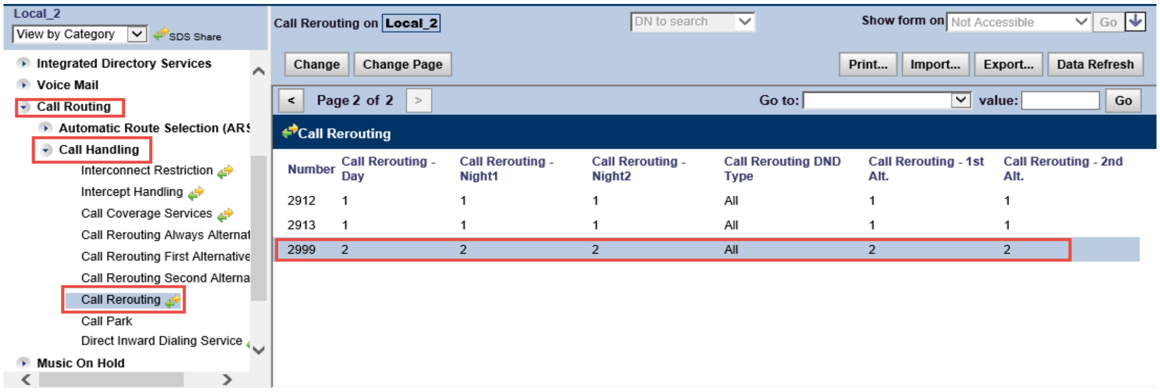


Figure 44 – Call Rerouting

MiCollab NuPoint Configuration

Network Elements

From Server Manager, Navigate to: Applications -> Users and Services -> Network Element, Click Add

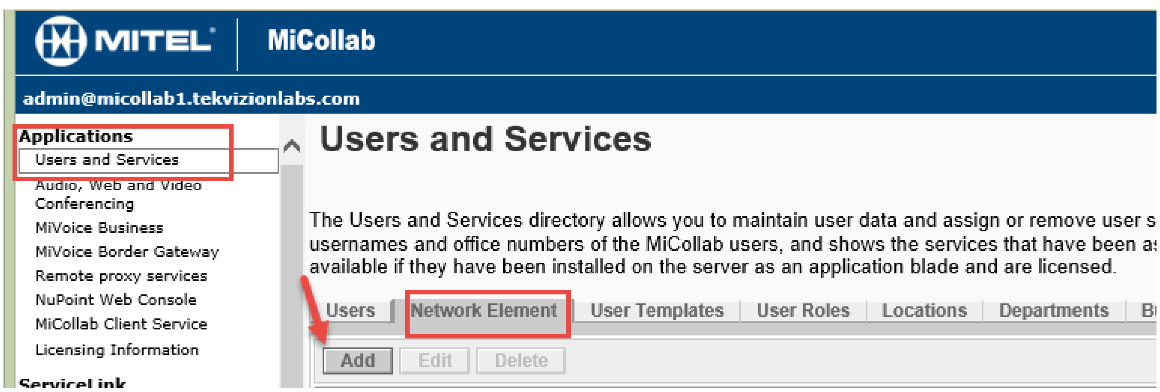


Figure 45 Add Network Element

- Set **System Name**: **3300** is given in this test.
- Set **Network Address**: This is your MiVoice Business 3300 ICP IP address.
- Set **Credentials**: This is your Mivoice Business 3300 ICP administration credential.
- **Set Registration Code**: ******* is given which is match **Set Registration Access Code** in System Options in section.
- **Set Replacement Code**: ### is given to match **Set Replacement Access Code** in System Options section.
- Set **Standard Phone COS**: **5** is given for all fields to match the Class of Service for Nupoint Voice Mail port created in Class of Service Option section.
- Set **Default COR**: **1** is given to all fields in this setup.
- Set **Call Forward Destination Directory Number**: **2900** is given which is the Hunt Group Number for NuPoint Voice Mail.

- Set **HCI Reroute Hunt Group Number for Mitai MWI**: **2999** is given to match previous configuration
- Click **Save**.

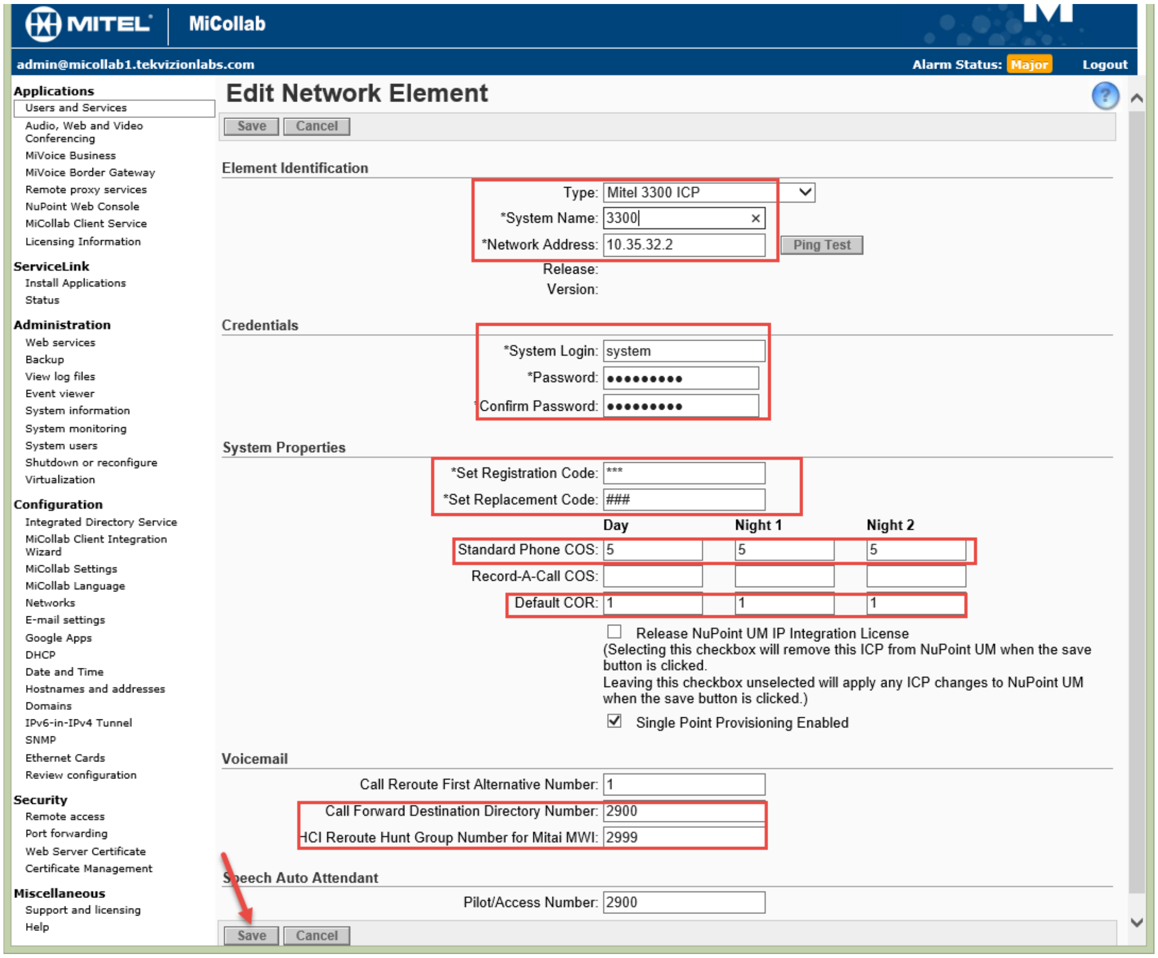


Figure 46 – Network Element cont.

Voice Mail Line Group

From Offline Configuration window select the Line Groups link under the Offline Configuration heading. On the Line Groups web page, click the Add button

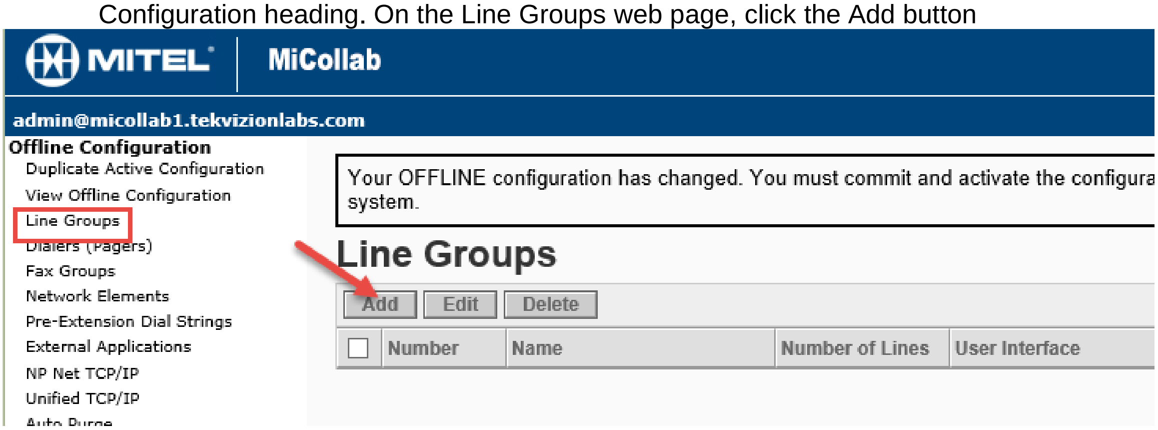


Figure 47 – Voicemail Line Group Configuration

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

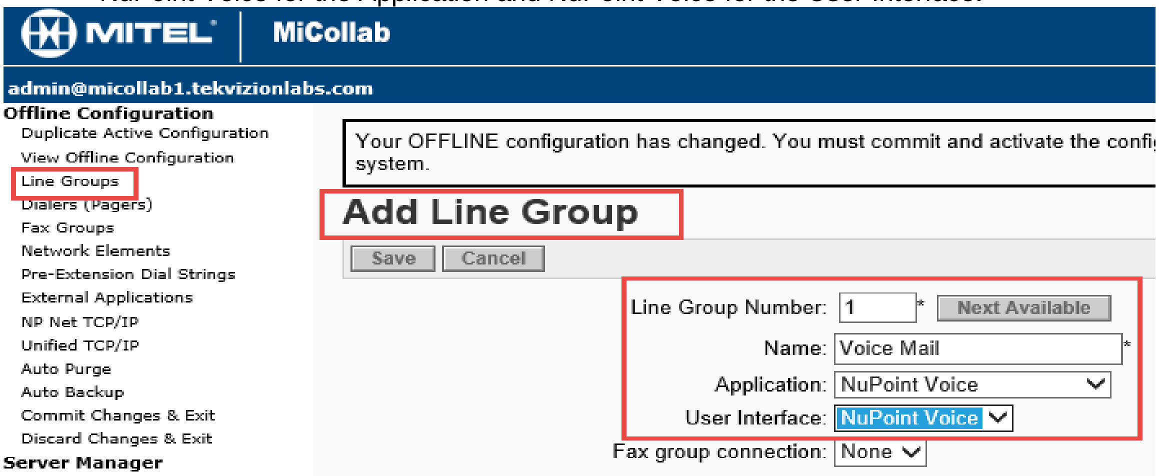


Figure 48 – Adding Line Group

Next click the Add Button under the Lines heading. This will bring up the Line Triplet dialogue box. Click the Next Available button 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 3300. Enter the first extension number that was created in the section [IP Endpoints] used for NuPoint Ports in the Mapping field.

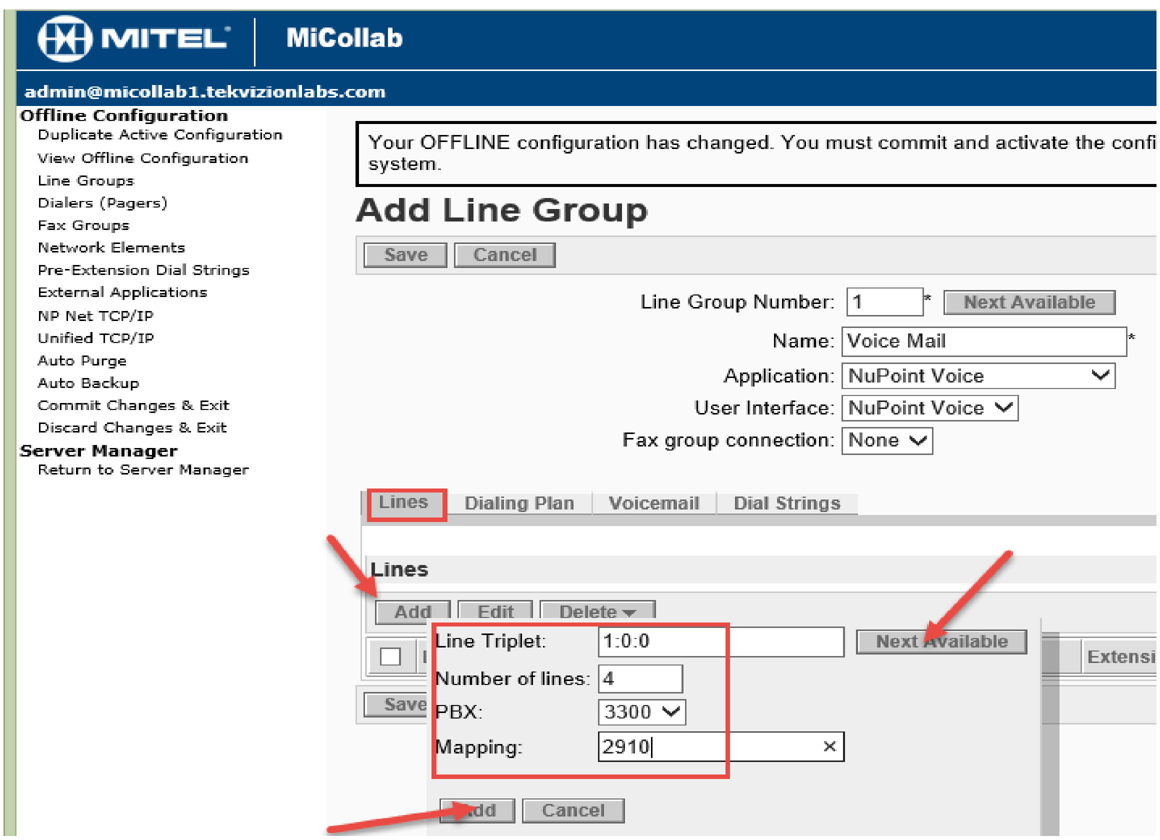


Figure 49 – Adding Lines

Next, click on the **Dialing Plan** tab on the Add Line Group page. This will bring up the Dialing Plan web 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 due to by default, mailboxes 999 and 998 are created, 998 is the default administrative mailbox and 999 is the default attendant mailbox.

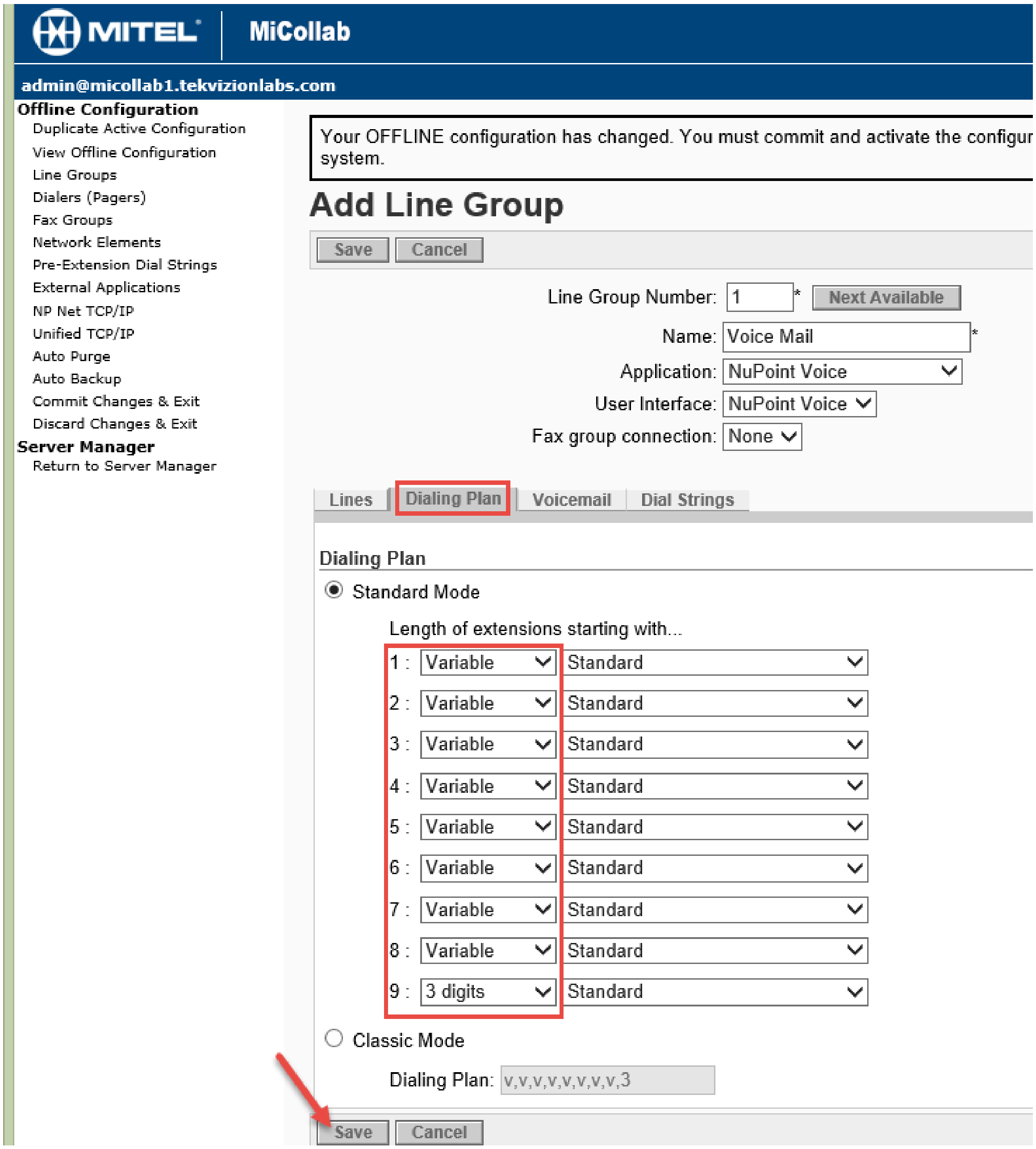


Figure 50 – Adding Dial Plan

Click **Save** button to save the configuration

The next step is to commit the changes that have been made to the offline configuration. Click on the Commit changes & Exit link under the Offline Configuration heading. Click on the Commit button.

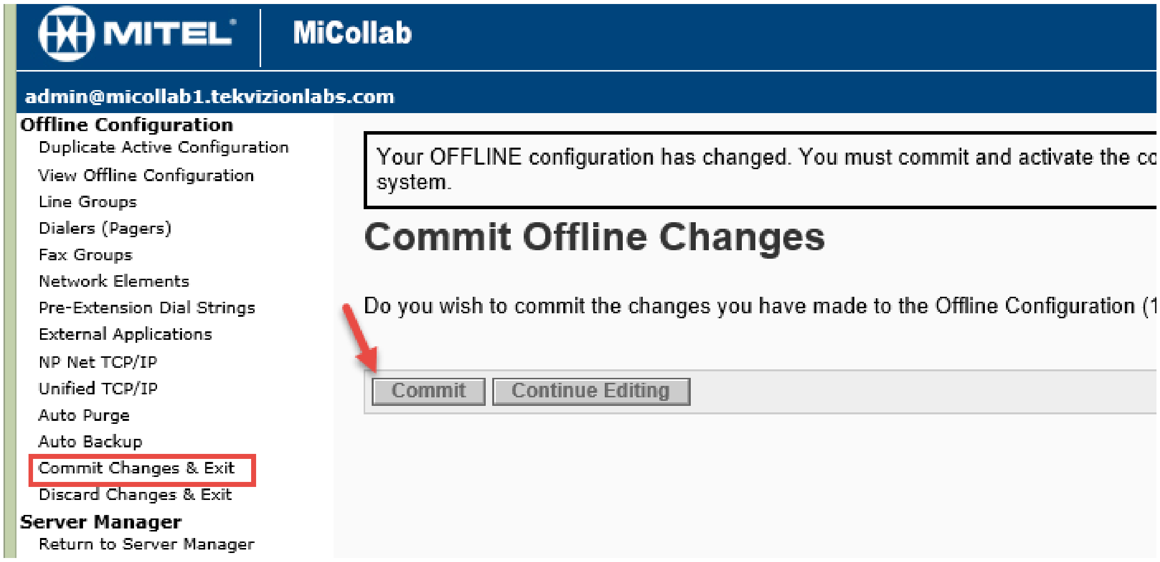


Figure 51 – - Committing Offline Changes

Next click 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 the Activate button.

Adding Mailboxes

NuPoint with MAS and Single Point Provisioning allows for programming 3300 phones, users and NuPoint Mailboxes from the MAS interface. We assume 3300 phones and users were configured in MiVoice Business Configuration Notes Section and this chapter only cover adding mailboxes.

Navigate to **Mailbox**, click **ADD**

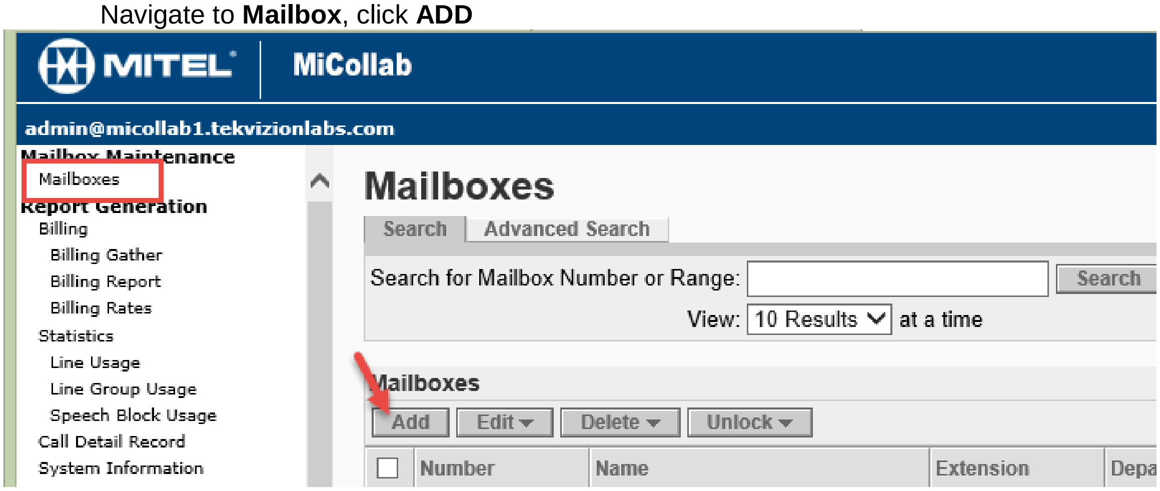


Figure 52 – Add Mailbox

Mailbox 1029 is created for this test. Under **General** tab, set the proper **Name**, **Passcode** and associated 3300 phone/user as **Extension**.

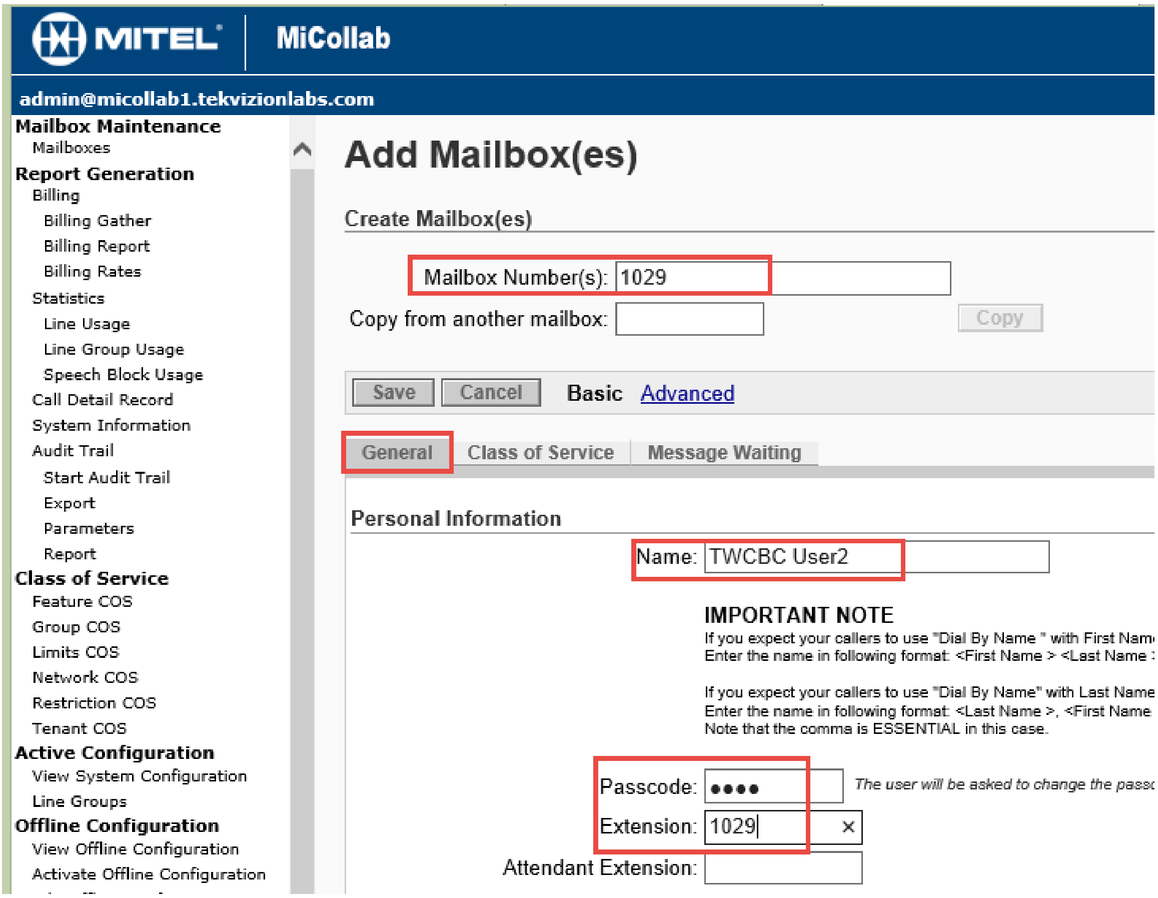


Figure 53 – Add Mailbox cont.

Under **Message Waiting** tab, select **Mitai Messaging** as **Type** then click **Save**.

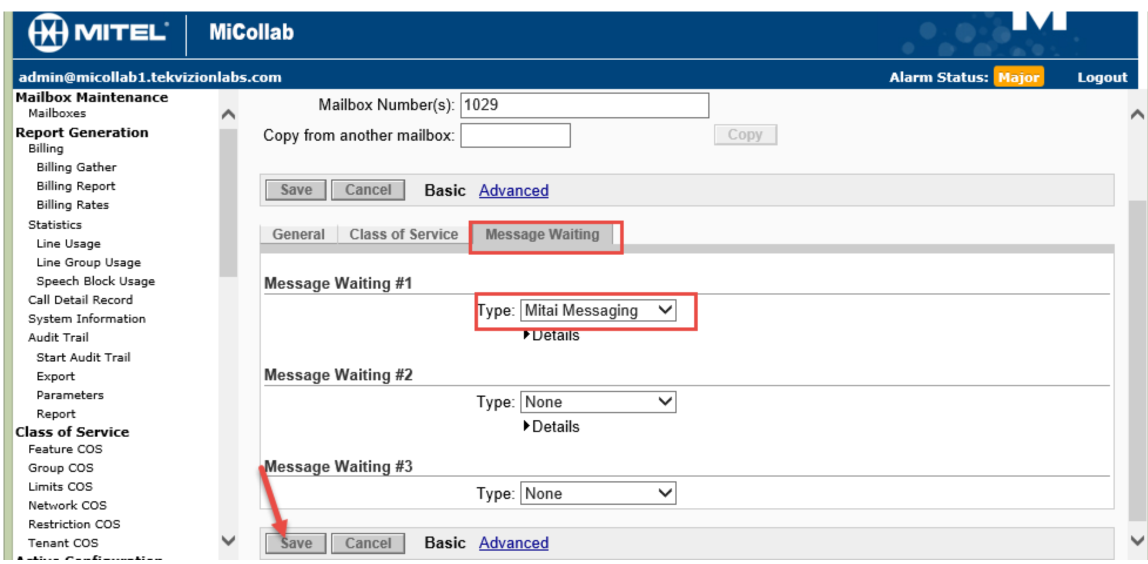


Figure 54 – Add Mailbox cont.

Click done when pop-up window shows the mailbox added successfully.

MiVoice Border Gateway Configuration Notes

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

Navigate to: MiVoice Border Gateway -> Configuration -> Network Profiles

Click the "→" beside Sever-gateway configuration on the network edge then click Apply.

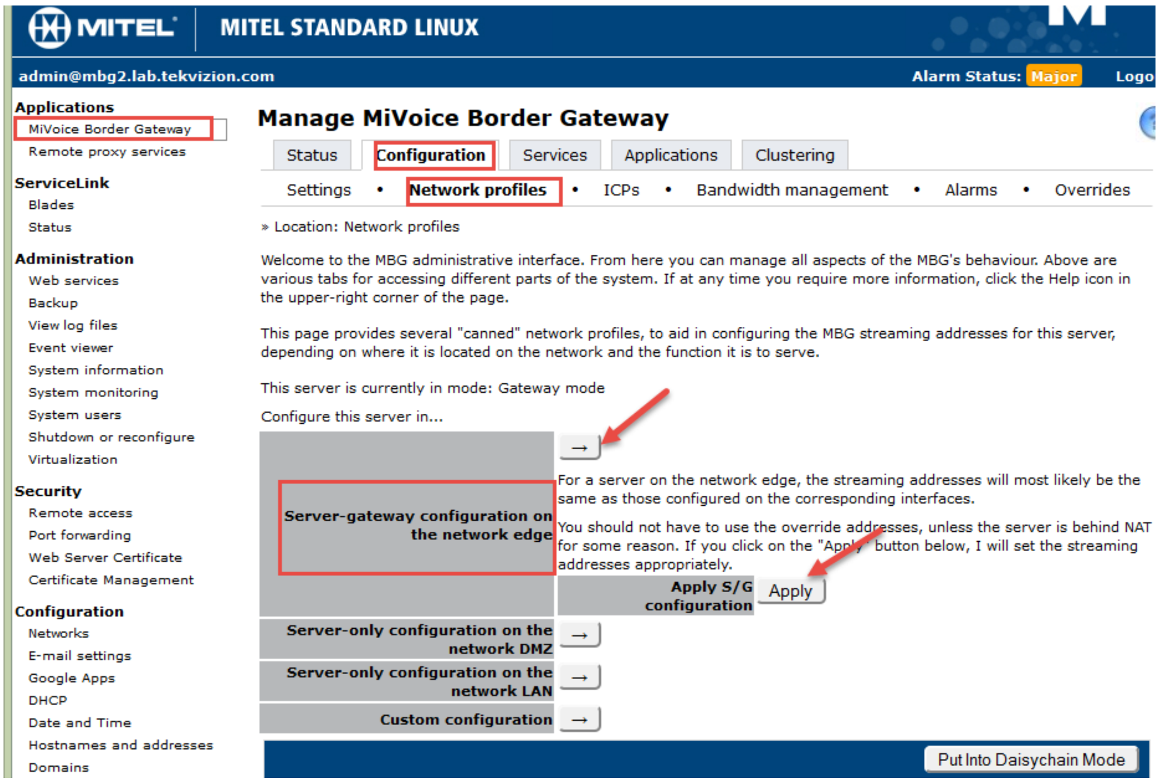


Figure 55 – Network Profiles

In order to make the mid-call feature works for External Hot Desk User, need to setup KPML username and password under Configuration -> Settings, click Edit.

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

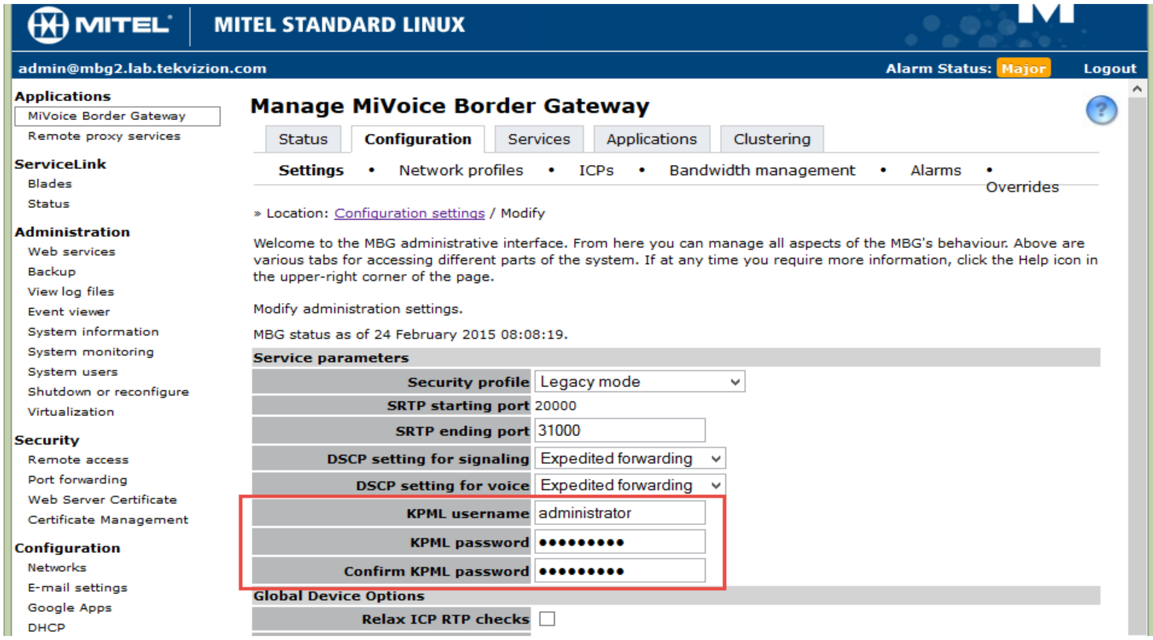


Figure 56 – MBG Settings

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

Navigate to **MiVoice Border Gateway** -> **Configuration** -> **ICPs**

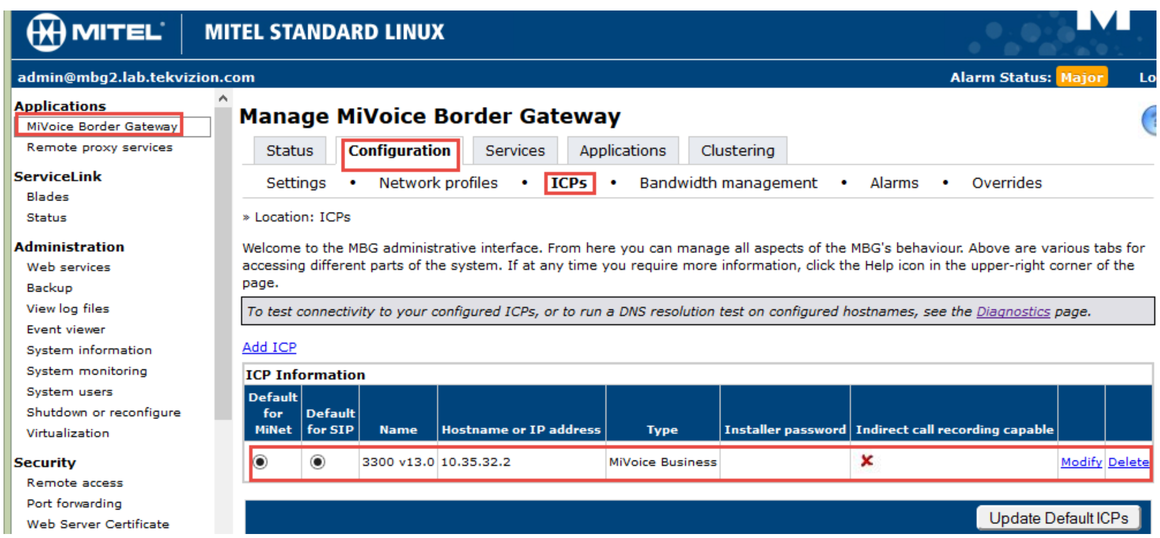


Figure 57 – MIVOICE BORDER GATEWAY's Configuration page

- 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 button.

To add a new SIP trunk:

- Click **Services** tab and then click **SIP trunking**
- Click **Add a SIP trunk** link.

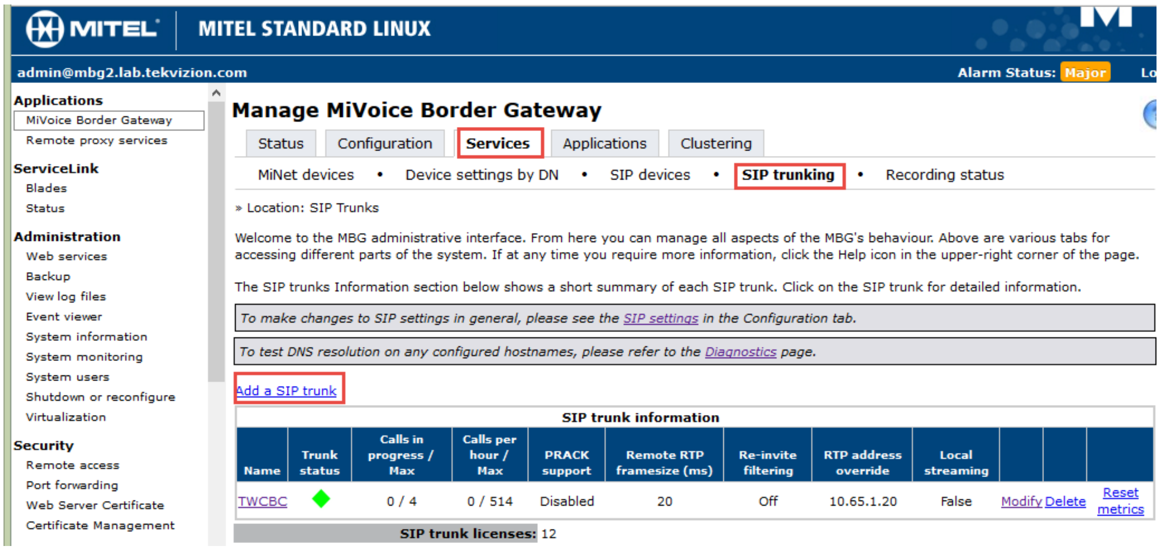


Figure 58 – SIP trunking configuration page

Enter the SIP trunk details as follow:

Set **Name**: TWCBC is given in this setup

Set **Remote trunk endpoint address**: 10.65.1.200 is given in this lab setup. This is the LAN IP Address of the TWCBC ESG, Please contact TWCBC for the IP address for your deployment.

Set **Remote trunk endpoint port**: 5060 is used as suggested by TWCBC.

Set **Remote RTP framesize (ms)**: This is the packetization rate you want to set on this trunk, TWCBC only supports 20ms packetization rate.

Set **RTP address override**: LAN Interface is select from drop- down. MBG send/receive all SIP/RTP packets to/from TWCBC ESG via LAN interface as the WAN of MBG is setup with public IP address for Teleworker.

Set **PRACK**: Select Disabled from drop-down as TWCBC does not support PRACK.

Set **Routing rules:** It allows routing of any digits to the selected MiVoice Business ICP.

The rest of the settings are optional and could be configured if required.

Click **Save** button.

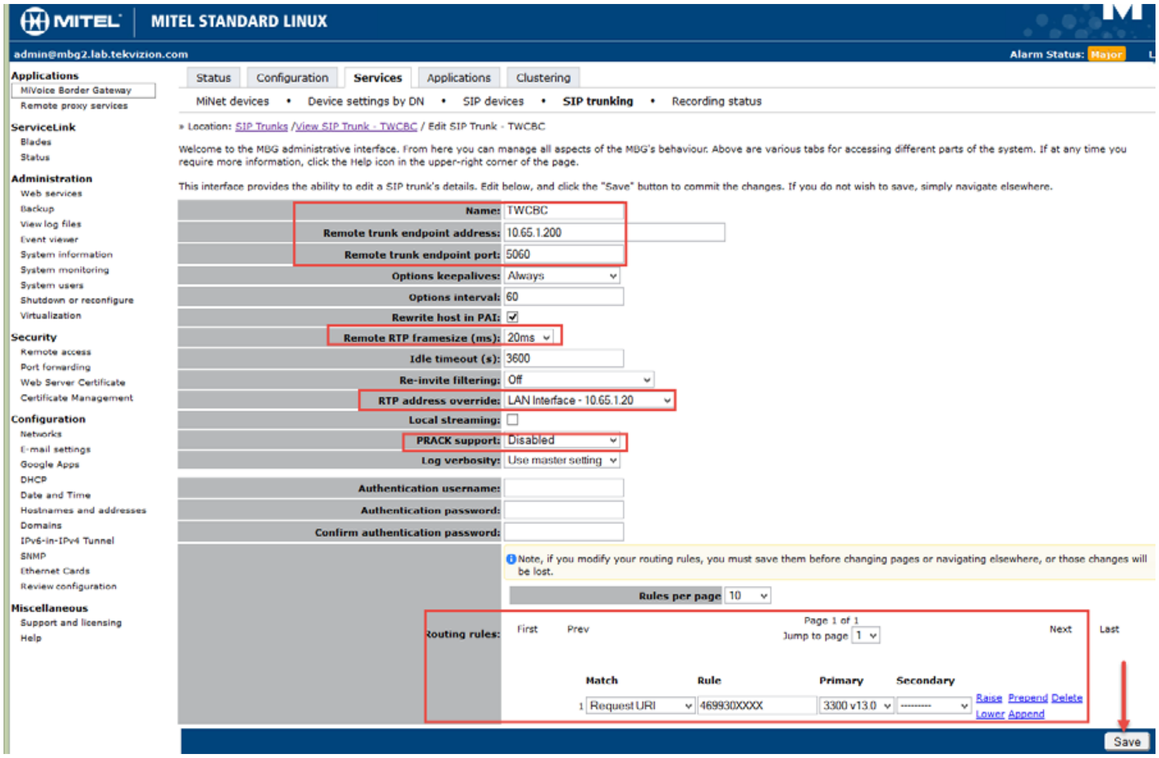


Figure 59 – SIP Trunk configuration settings

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

www.mitel.com

For more information on our worldwide office locations, visit our website at www.mitel.com/offices