

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

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



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

March 2015 15-4940-00363

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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:

	<p>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.</p>
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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 Plans (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.

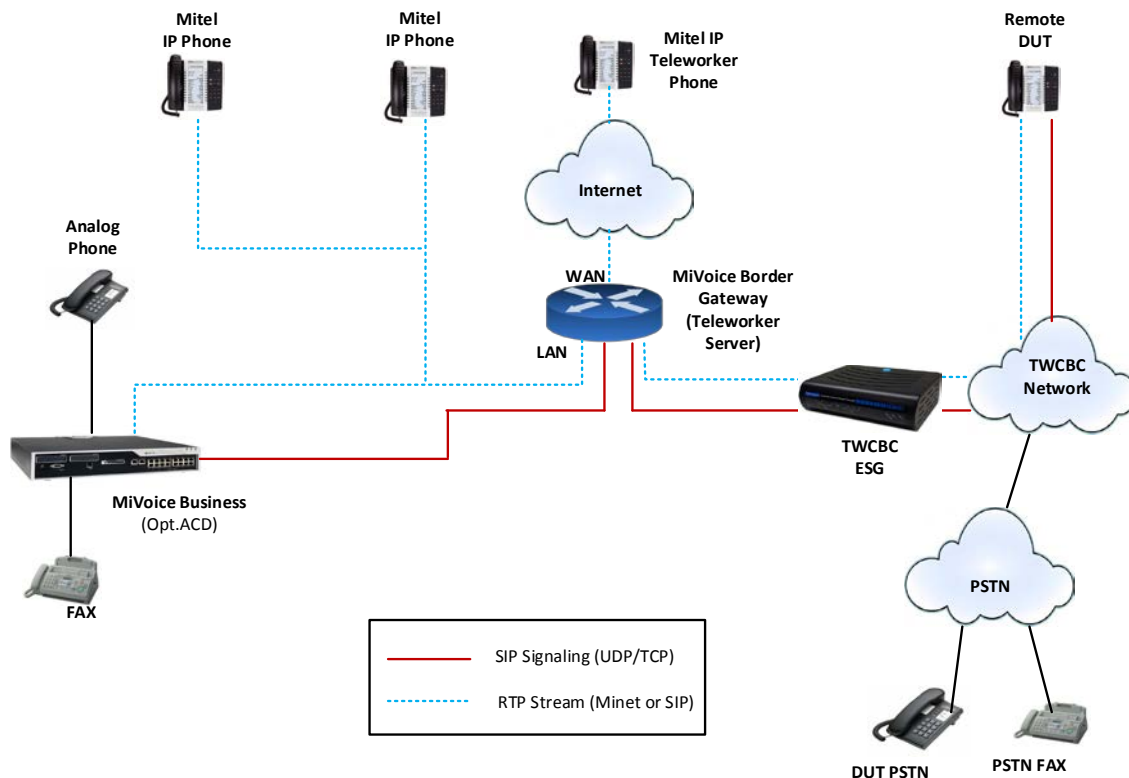


Figure 1 – Network Topology

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.

Local_2 License and Option Selection on Local_2

View by Category SDS Share

Change Print... Import... Export... Data Refresh

License and Option Selection

Online Licensing with the Application Management Center

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	Can be Over Allocated
Users						
IP Users	11	16	0	16	Unrestricted	Yes
External Hot Desk Users	1	5	5	10	Unrestricted	Yes
ACD Active Agents	1	10	0	10	Unrestricted	No
HTML Applications	0	0	20	0	Unrestricted	Yes
Analog Lines	0	16	0	16	Unrestricted	Yes
MiVoice Business Console Active Operators	0	0	20	0	Unrestricted	No
Multi-device Users	0	5	0	5	Unrestricted	Yes
Multi-device Suites	0	0	5	5	0	No
Messaging						
Embedded Voice Mail	1	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	10	353	0	353	Unrestricted	Yes
Others						
IDS Connection	1	Yes	0	1	Unrestricted	Yes
MLPP	0	No	0	0	Unrestricted	No
Configuration Options						
Country		North America				
Extended Agent Skill Group		No				
Maximum Elements per Cluster		30				
Maximum Configurable IP Users and Devices		700				
Extended Hunt Group		Yes				
5560 IPT Device Extended Key Lines		No				

Figure 2 – License and Option Selection

Class of Service Assignment

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

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

Local_2
View by Category SDS Share
Class of Service Options on Local_2
DN to search
Show form on Not Accessible
Change Copy Print... Import... Export... Data Refresh
Page 1 of 10
Go to: value: Go

Class Of Service Number	Comment
1	TWCBC
2	Mitel Phone
3	Embedded VM
4	RAD_ANN
5	NuPoint VM
6	
7	
8	
9	
10	

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 Work Timer	No
	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 Adjustment 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
Ringling	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringling Line Select	No
Ringling 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 Work Timer	No
	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
Ringling	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringling Line Select	No
Ringling 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.

Network Elements	
Name	TWCBC
Type	Other
FQDN or IP Address	10.65.1.200
Local	False
Version	
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	10.65.1.200
External SIP Proxy Transport	UDP
External SIP Proxy Port	5060
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

Save Cancel

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).

The screenshot shows a configuration window titled "Network Elements". The interface includes several input fields and a dropdown menu, with red boxes highlighting specific areas:

- Name:** TWCBC_MBG x
- Type:** Outbound Proxy (dropdown menu)
- FQDN or IP Address:** 10.65.1.20
- Local:** False
- Version:** (empty field)
- Zone:** 1
- ARID:** (empty field)
- Outbound Proxy Specific:**
 - Outbound Proxy Transport Type:** default (dropdown menu)
 - Outbound Proxy Port:** 0

At the bottom right of the window, there are "Save" and "Cancel" buttons.

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.

Trunk Attributes	
Trunk Service Number	1
Release Link Trunk	No
Call Recognition Service	Trusted
Direct Inward Dialing Service	<input type="radio"/> Off <input checked="" type="radio"/> On
Class of Service	1
Class of Restriction	1
Baud Rate	9600
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Label	TWCBC

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 EHCU in the PRG (Personal Ring Groups).

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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
SIP Peer Profile Label		TWCBC		
Network Element		TWCBC		
Local Account Information				
Registration User Name				
Address Type		IP Address: 10.35.32.2		
Administration Options				
Interconnect Restriction		1		
Maximum Simultaneous Calls		10		
Minimum Reserved Call Licenses		10		
Administration Options				
Outbound Proxy Server		TWCBC_MBG		
SMDR Tag		0		
Trunk Service		1		
Zone		1		
User Name				
Password		*****		
Confirm Password		*****		
Authentication Option for Incoming Calls		No Authentication		
Subscription User Name		administrator		
Subscription Password		*****		
Subscription Confirm Password		*****		

Figure 19 – SIP Peer Profile - Basic

Call Routing (figure 20):

Leave the default settings as shown.

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 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.

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	
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 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.

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				
Yes				
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 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.

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
Trunk Group Label				
Allow Display Update				
				No
Build Contact Using Request URI Address				
				No
De-register Using Contact Address not *				
				Yes
Disable Reliable Provisional Responses				
				Yes
Disable Use of User-Agent and Server Headers				
				No
E.164: Enable sending '+'				
				No
E.164: Add '+' if digit length > N digits				
				0
E.164: Do not add '+' to Emergency Called Party				
				No
E.164: Do not add '+' to Called Party				
				No
Force Max-Forward: 70 on Outgoing Calls				
				No
If TLS use 'sips:' Scheme				
				No
Ignore Incoming Loose Routing Indication				
				No
Only use SDP to decide 180 or 183				
				Yes
Prefer From Header for Caller ID				
				No
Require Reliable Provisional Responses on Outgoing Calls				
				No
Signal Privacy (if enabled) on Emergency Calls				
				No
Suppress Redirection Headers				
				No
Use Fixed Retry Time for 491				
				No
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-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

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.

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	120			
Session Timer: Local as Refresher	No			
Subscription Period	3600			
Subscription Period Minimum	300			
Subscription Period Refresh (%)	80			
Invite Ringing Response Timer	0			

Figure 24 – SIP Peer Profile Assignment- Timers

Key Press Event (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**

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation
Timers	Key Press Event	Outgoing DID Ranges	Profile Information	
Allow Inc Subscriptions for Local Digit Monitoring	Yes			
Allow Out Subscriptions for Remote Digit Monitoring	Yes			
Force Out Subscriptions for Remote Digit Monitoring	No			
Request Outbound Proxy to Handle Out Subscriptions	Yes			
KPML Transport	UDP			
KPML Port	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.

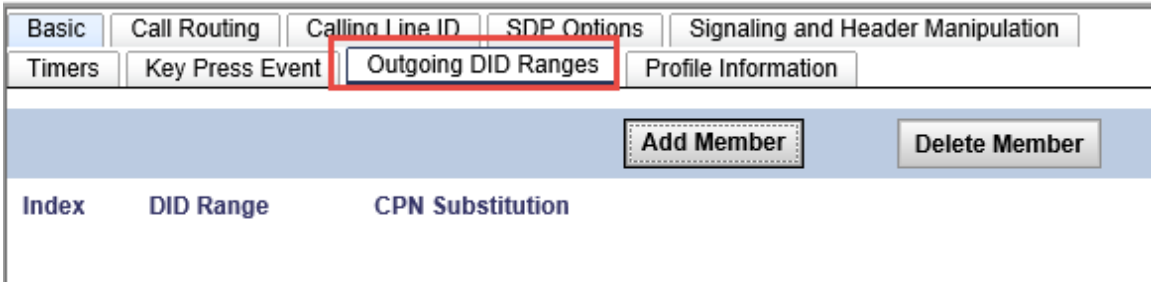


Figure 26 – SIP Peer Profile Assignment- Outgoing DID Ranges

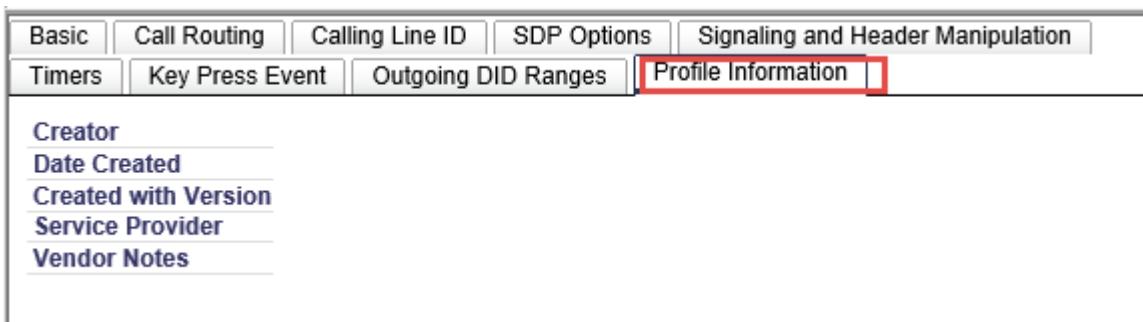


Figure 27 – SIP Peer Profile Assignment – Profile Information

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).

The screenshot displays the configuration interface for ARS Digit Modification Plans. On the left, a navigation menu is visible with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, Users and Devices, Integrated Directory Services, Voice Mail, Call Routing (expanded), Automatic Route Selection (ARS) (expanded), ARS Call Progress Tone Detectio..., ARS Digit Modification Plans (selected), ARS Maximum Dialed Digits, ARS Routes, ARS Route Lists, ARS Route Plans, and ARS Digits Dialed. The main content area shows a table with the following data:

Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan/Information Marker
1	1		

A 'Webpage Dialog' is open over the table, showing a form for editing the selected plan. The form contains the following fields:

- Digit Modification Number: 1
- Number of Digits to Absorb: 1
- Digits to be Inserted: (empty)
- Final Tone Plan/Information Marker: (empty)

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.

The screenshot displays the ARS Routes configuration page. On the left is a navigation tree with 'ARS Routes' highlighted. The main area shows a table of routes and a configuration form for Route Number 1.

Route Number	Routing Medium	Trunk Group Number	SIP Peer Profile	PBX Number / Cluster Element ID	COR Group Number	Digit Modification Number	Digits Before Outpulsing	Route Type
1	SIP Trunk		TWCBC		1	1		Non-verified Account

Route Number	1
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	TWCBC
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	1
Digits Before Outpulsing	
Route Type	Non-verified Account
Compression	Off

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.

The screenshot shows the configuration interface for ARS Digits Dialed. On the left is a navigation tree with the following items:

- Licenses
- LAN/WAN Configuration
- Voice Network
- System Properties
- Hardware
- Trunks
- Users and Devices
- Integrated Directory Services
- Voice Mail
- Call Routing
 - Automatic Route Selection (ARS)
 - ARS Call Progress Tone Detectic
 - ARS Digit Modification Plans
 - ARS Maximum Dialed Digits
 - ARS Routes
 - ARS Route Lists
 - ARS Route Plans
 - ARS Digits Dialed**
 - ARS Leading Digits
 - ARS Day and Time Zones
 - ARS Node Identities
 - Call Handling
- Music On Hold
- Emergency Services Management
- Property Management
- Maintenance and Diagnostics

The main content area shows a table titled "ARS Digits Dialed" with the following data:

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
8	9	Route	1
9	10	Route	1
91	10	Route	1

Below the table is a "Range Programming -- Webpage Dialog" window titled "Change Range Programming - ARS Digits Dialed". It contains the following information:

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

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

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:

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

Buttons at the bottom of the dialog include "Preview", "Save", and "Cancel".

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).

The screenshot displays the 'Voice Network' configuration page. The left sidebar shows a tree view with 'Voice Network' and 'Fax Service Profiles' highlighted. The main area shows the 'Inter-Zone Fax Profile' settings and a table of 'Intra-Zone Fax Service Profiles'.

Inter-Zone Fax Profile Settings:

- Maximum Fax Rate: 14400 (V.17, 14400bps)
- High Speed Redundancy: 1
- Low Speed Redundancy: 3
- Error Correction Mode (ECM): Disabled
- Override Non-Standard Facilities (NSF): Disabled
- Label: Inter-zone

Intra-Zone Fax Service Profiles Table:

Profile	Maximum Fax Rate	High Speed Redundancy	Low Speed Redundancy	Error Correction Mode	NSF Override	NSF Vendor Code Value	NSF Country Code Value	Label
1	-	-	-	-	-	-	-	G.711
2	14400 (V.17, 14400bps)	1	3	Disabled	Disabled	.	.	T.38
3
4

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.

The screenshot displays the 'Network Zones' configuration page. On the left is a navigation tree with 'Voice Network' and 'Network Zones' highlighted. The main area shows a table of zones:

Zone ID	Intra-zone Compression	Intra-zone Fax Profile	Label	SMDR Tag	Time Zone	LBN Prefix	Zone CESID	Default Billing Number	Default CPN
1	No	1	TWCBC		America/Chicago				
2	No	1							
3	No	1							

An overlay dialog titled '-- Webpage Dialog' shows the configuration for Zone 1:

- Zone ID: 1
- Intra-zone Compression: No Yes
- Intra-zone Fax Profile: 1
- Label: TWCBC
- SMDR Tag: (empty)
- Time Zone: America/Chicago
- LBN Prefix: (empty)
- Zone CESID: (empty)
- Default Billing Number: (empty)
- Default CPN: (empty)

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

Personal Ring Group	One Busy All Busy	Prime Member Name	Home Element	Secondary Element
1029	No	TWCBC,TWO	Local_2	Not Assigned

Personal Ring Group	1029
Local-only DN	False
One Busy All Busy	No
Prime Member Name	TWCBC,TWO
Home Element	Local_2
Secondary Element	Not Assigned

Member Index	Number	Presence	Name	Home Element	Secondary Element
1	1029	Present	TWCBC,TWO	Local_2	Not Assigned
2	2030	Present	TWCBC,hotdesk	Local_2	Not Assigned

Figure 33 – Personal Ring Groups

Multiline IP sets 1029 and 2030 are configured as following.

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

Device Id	Hot Desk	Device Type	Auxiliary Module	ACD Number	Enabled	Line Type	Interconnect Number	Hot Desk User External Dialing Prefix	Hot Desk External User Number	Max Call History Records	Tenant Language	Service Number Level
1	No	5320 IP	None	0442	Yes	Single Line 1				0	English	1 Full
2	No	5312 IP	None	1029	No	Single Line 1				0	English	1 Full
3	No	5360 IP	None	0487	Yes	Single Line 1				0	English	1 Full
4	No	5360 IP	None	1045	No	Single Line 1				0	English	1 Full
5	Yes		None	2030	No	Single Line 1	9	14699300487		0	English	1 Full
6	No	5020 IP	None	2910	No	Single Line 1				0	English	1 Full

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:

2. Define the Change Range Programming Pattern:

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

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**.

System Options	
AC system	No
ACD Event Statistics Refresh Rate	4
ACD Make Busy On Login Reason Code	0
ACD Make Busy Walk Away Codes	No
ACD Make Last Agent Unavailable on No Answer	No
ACD Number of Threshold Alert Indicators Rate	30
ACD Real Time Events Feature Level	0
Advice of Charge - Multiplier	0
Advice of Charge - Surcharge	0
Advice of Charge Feature Active	No
Alpha Tagging Enabled	No
Att Cancel-All Feature Access	None
Battery Backup	No
Battery Cabinet Alarm Information	No
BLF - Busy Indication based on set enabled	No
BLF - CFA Indication based on set enabled	No
Call Forwarding Always - Line Status Indicator ON	No
Call History - Default Call History Records	20
Call History - Disable Record Generation	No
Call Rerouting Timer	22
Callback Activation	Group
Callback Cancel Timer	8
Campon Repetitive Tone Timer	0
Conference/Call Intrusion Repetitive Tone Timer	0
Data Line Error Threshold	100
Default Language	English
Dialed Number Editing For Trunks	Yes
DISA Failed Attempts before Lock-Out	3
DISA Number Lock-Out Timer	15
Disable End of Dial Character (#)	No
Do Not Override DND for Public Network DID/DDI Callers	No
DTRX Autobaud Timeout	60
DTRX DSA Response Format	Comment
DTRX Herald Message	SX2000:
DTRX Inactivity Timeout	60
Email Server	
Email - Sender's Address	
External Hot Desking - Single Digit Mid Call Features	Yes
Feature Active Dial Tone - Call Forwarding	No

Figure 36 – System Option

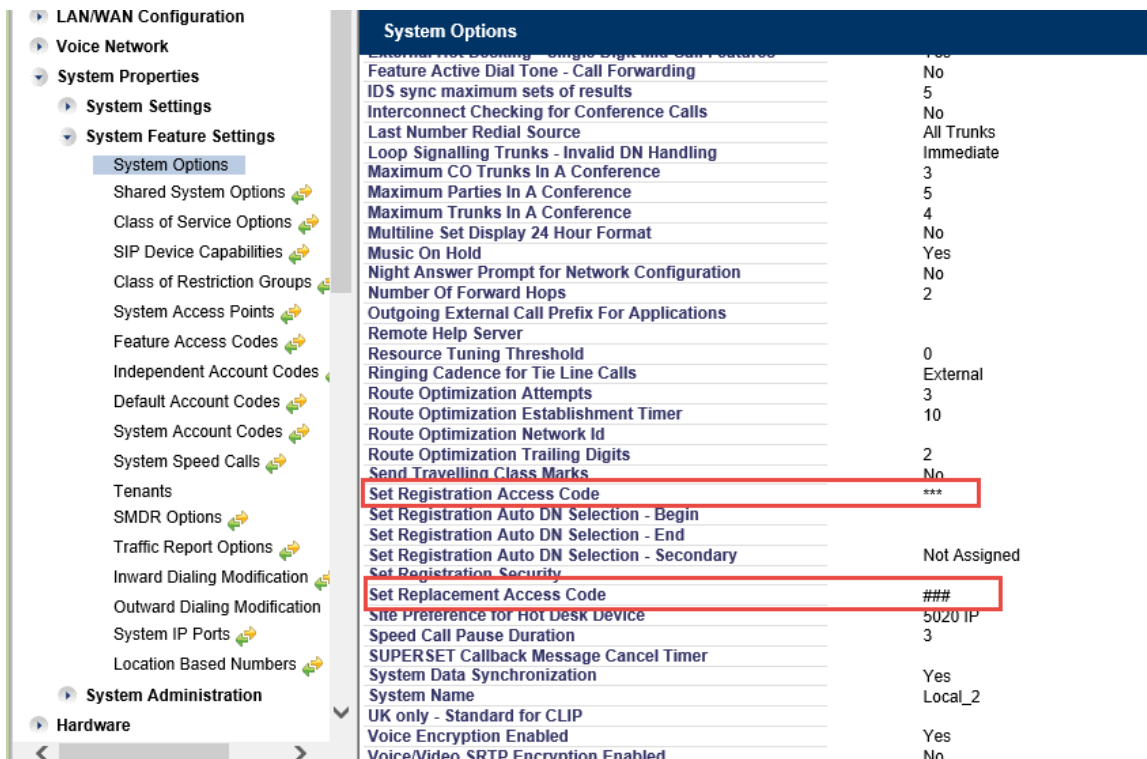


Figure 37 – System Options cont.

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.

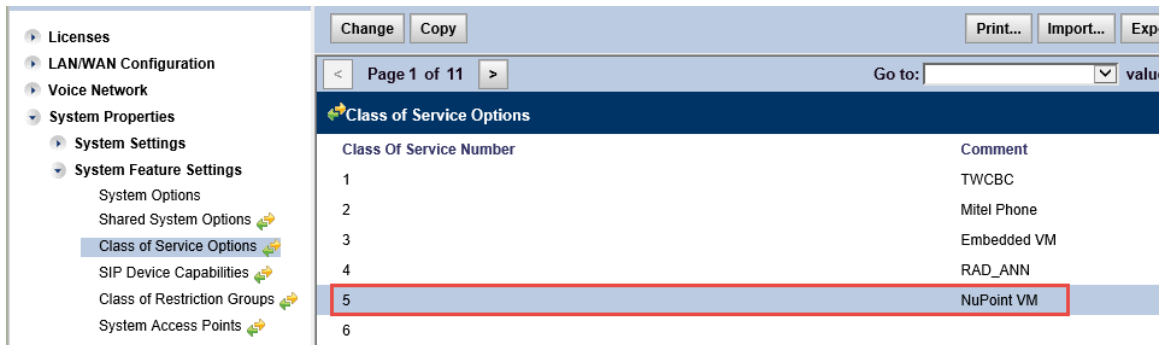


Figure 38 – Class of Service Option for NuPoint Voice Ports

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.

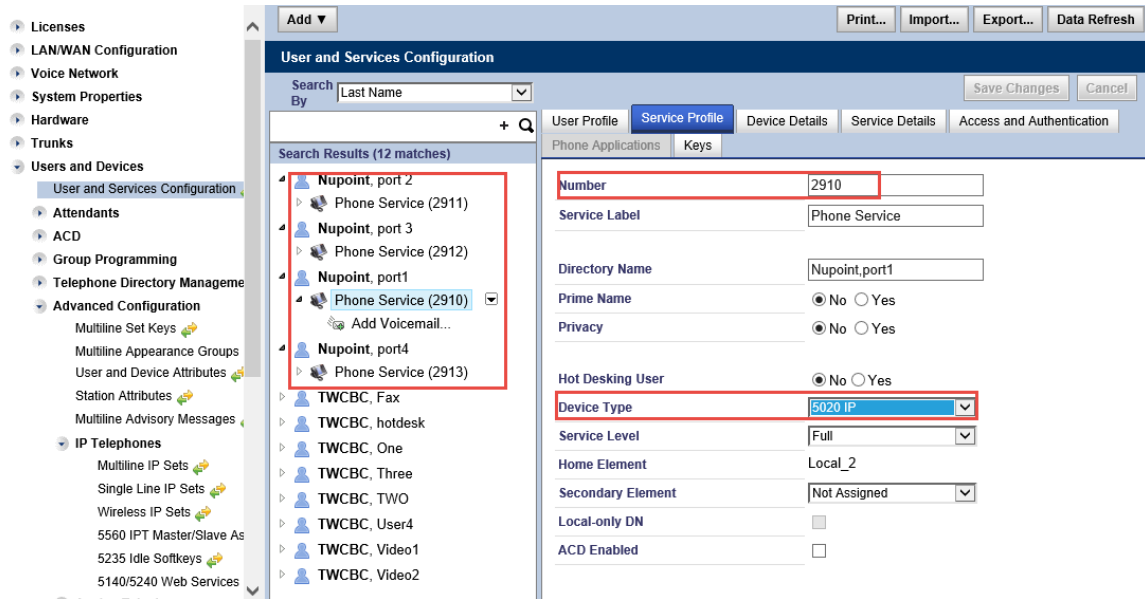


Figure 39 – IP Endpoints configuration

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.

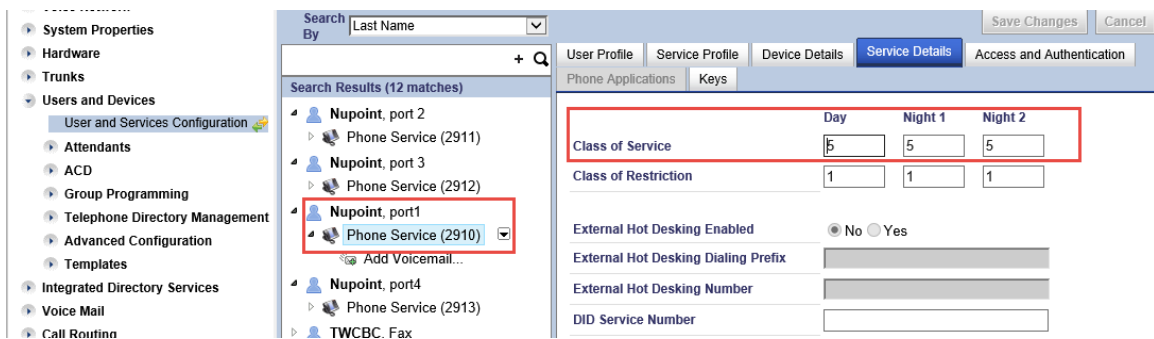


Figure 40 – IP Endpoints Class of Service

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.

The screenshot displays the Cisco Unified Communications Manager (CUCM) configuration page for Hunt Groups on the 'Local_2' domain. The left-hand navigation pane shows the tree structure with 'Users and Devices' and 'Group Programming' expanded, and 'Hunt Groups' selected. The main content area is divided into two sections: a table of existing Hunt Groups and a configuration form for the selected group (2900).

Hunt Groups Table:

Hunt Group	Hunt Group Mode	Hunt Group Name	Hunt Group Priority	Hunt Group Type	Home Element	Secondary Element
2100	Circular		64	VoiceMail	Local_2	Not Assigned
2900	Circular		64	VoiceMail	Local_2	Not Assigned
2999	Circular		64	HCIRoute	Local_2	Not Assigned

Hunt Group Configuration Form (2900):

- Hunt Group: 2900
- Local-only DN: False
- Hunt Group Mode: Circular
- Hunt Group Name: (empty)
- Class of Service - Day: (empty)
- Class of Service - Night1: (empty)
- Class of Service - Night2: (empty)
- Home Element: Local_2
- Secondary Element: Not Assigned

Below the configuration form is a 'Page 1 of 1' indicator and a 'Go to:' field with an 'Add Member' button.

Hunt Group Members Table:

Member Index	Number	Presence	Name	Home Element
1	2910	Present	Nupoint,port1	Local_2
2	2911	Present	Nupoint,port 2	Local_2
3	2912	Present	Nupoint,port 3	Local_2
4	2913	Present	Nupoint,port4	Local_2

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

Hunt Group	Hunt Group Mode	Hunt Group Name	Hunt Group Priority	Hunt Group Type	Home Element	Secondary Element
2100	Circular		64	VoiceMail	Local_2	Not Assigned
2900	Circular		64	VoiceMail	Local_2	Not Assigned
2999	Circular		64	HCIReroute	Local_2	Not Assigned

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

Figure 42 – HCIReroute Hunt Group

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

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

Figure 43 – Call Rerouting Always Alternative

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

Number	Call Rerouting - Day	Call Rerouting - Night1	Call Rerouting - Night2	Call Rerouting DND Type	Call Rerouting - 1st Alt.	Call Rerouting - 2nd Alt.
2912	1	1	1	All	1	1
2913	1	1	1	All	1	1
2999	2	2	2	All	2	2

Figure 44 – Call Rerouting

MiCollab NuPoint Configuration

Network Elements

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

The Users and Services directory allows you to maintain user data and assign or remove user s usernames and office numbers of the MiCollab users, and shows the services that have been available if they have been installed on the server as an application blade and are licensed.

Users Network Element User Templates User Roles Locations Departments B

Add Edit Delete

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**.

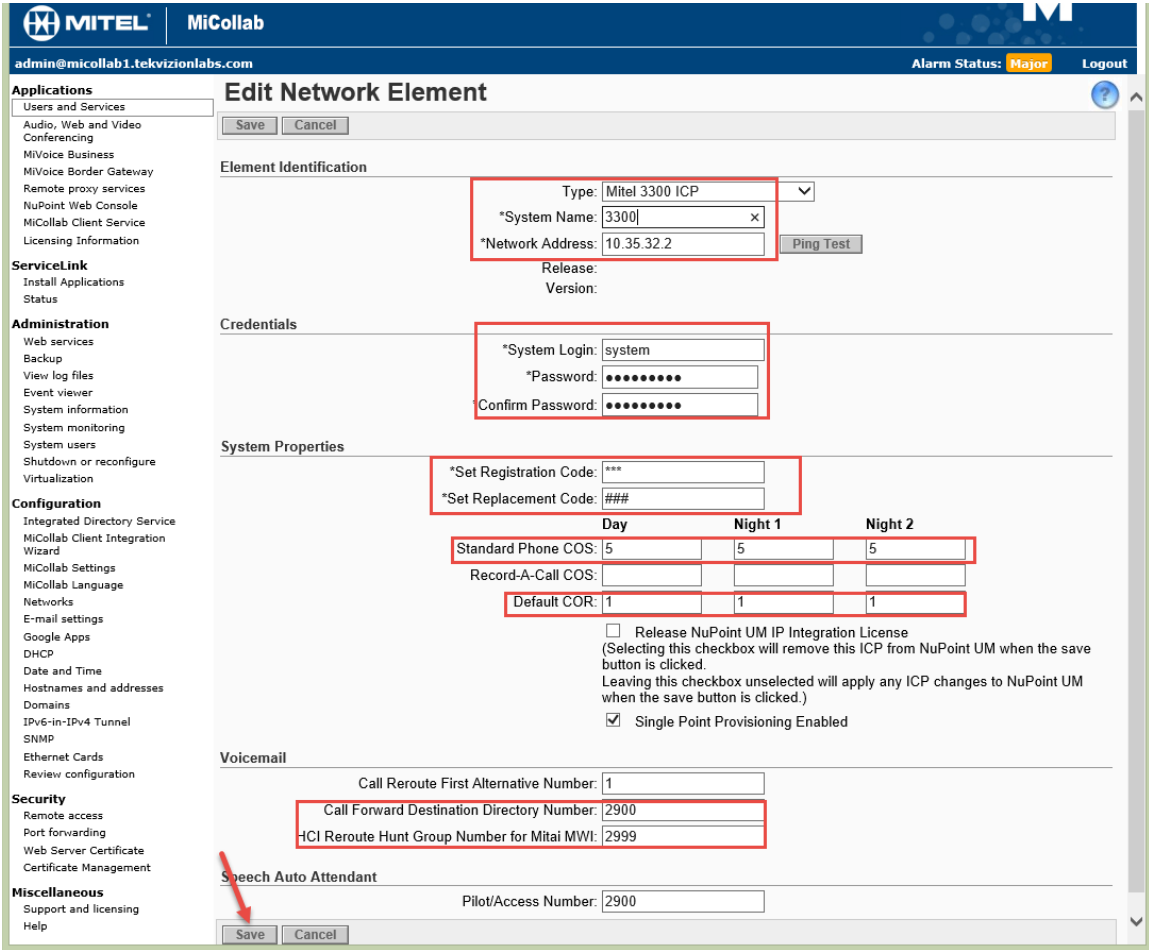


Figure 46 – Network Element cont.

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

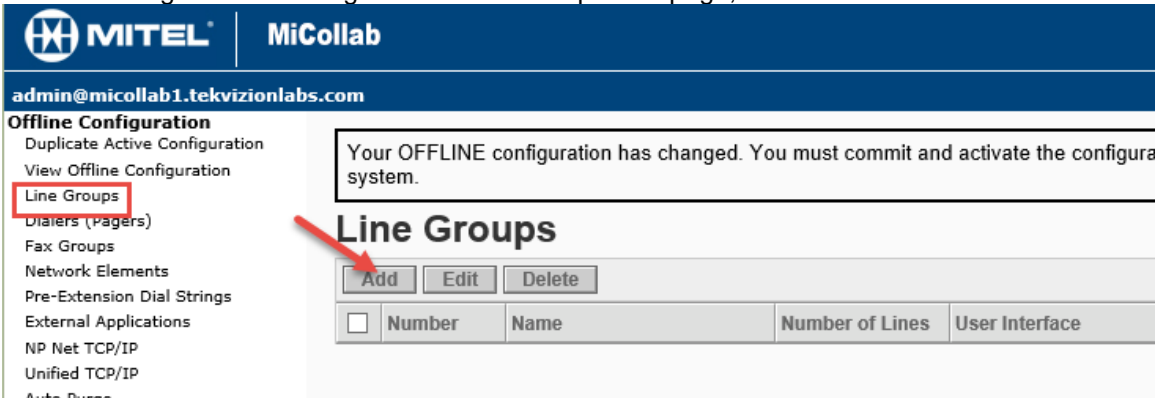


Figure 47 – Voicemail Line Group Configuration

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.

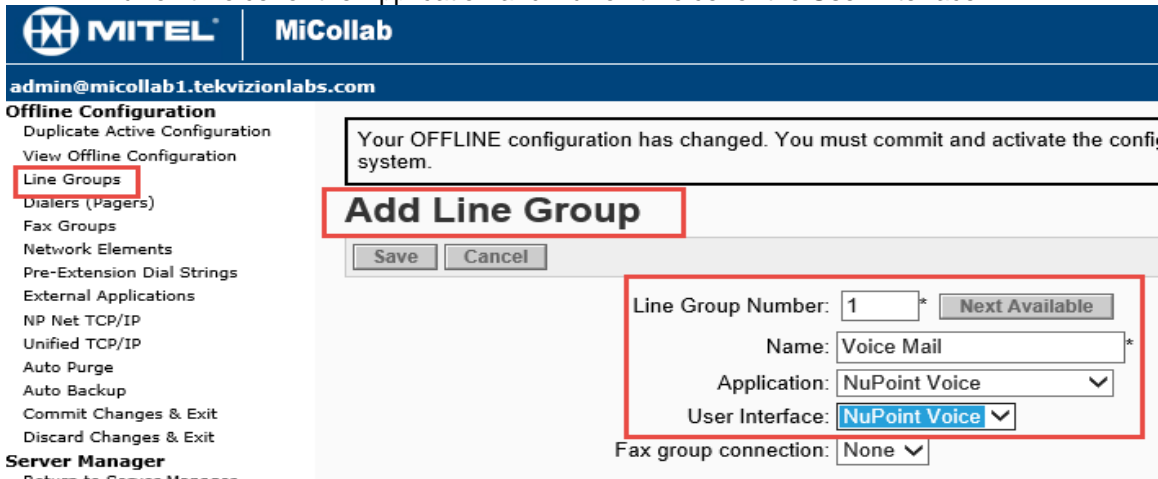


Figure 48 – Adding Line Group

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.

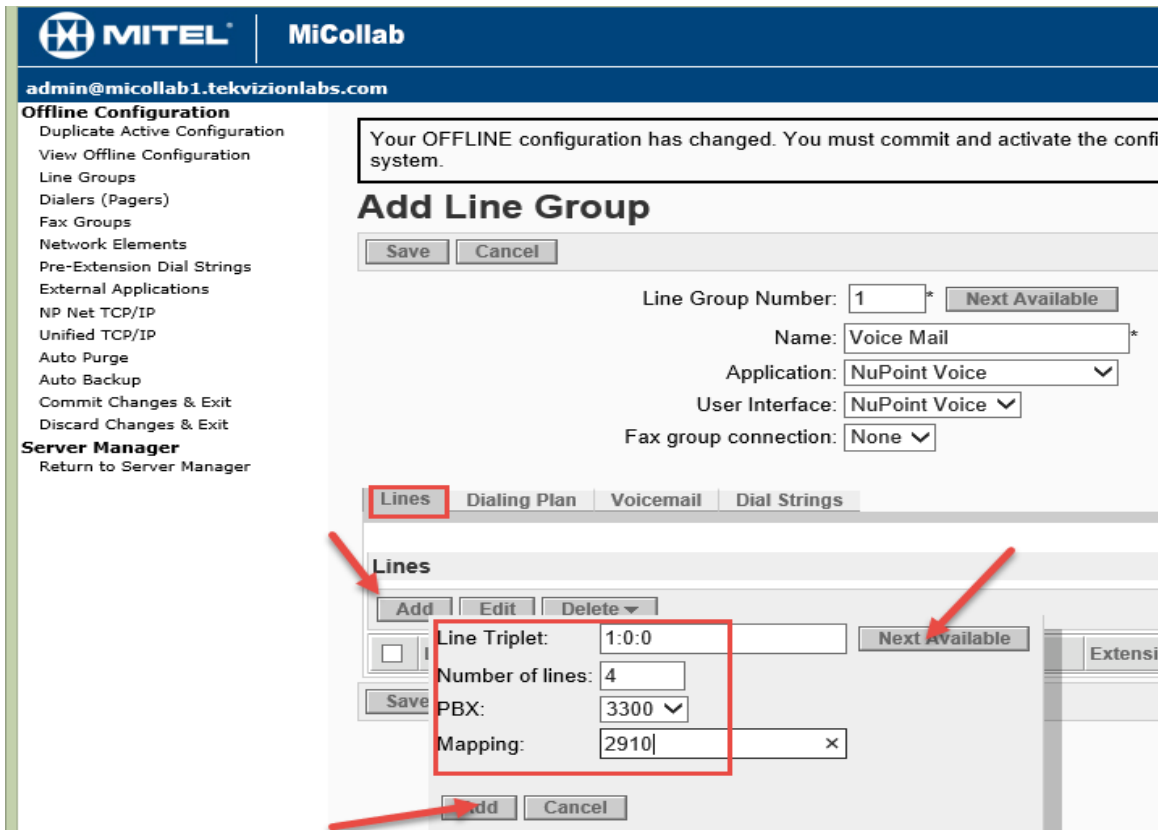


Figure 49 – Adding Lines

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.

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Offline Configuration

- Duplicate Active Configuration
- View Offline Configuration
- Line Groups
- Dialers (Paggers)
- Fax Groups
- Network Elements
- Pre-Extension Dial Strings
- External Applications
- NP Net TCP/IP
- Unified TCP/IP
- Auto Purge
- Auto Backup
- Commit Changes & Exit
- Discard Changes & Exit

Server Manager

- Return to Server Manager

Your OFFLINE configuration has changed. You must commit and activate the configur system.

Add Line Group

Save Cancel

Line Group Number: 1 * Next Available

Name: Voice Mail *

Application: NuPoint Voice

User Interface: NuPoint Voice

Fax group connection: None

Lines **Dialing Plan** Voicemail Dial Strings

Dialing Plan

Standard Mode

Length of extensions starting with...

1:	Variable	Standard
2:	Variable	Standard
3:	Variable	Standard
4:	Variable	Standard
5:	Variable	Standard
6:	Variable	Standard
7:	Variable	Standard
8:	Variable	Standard
9:	3 digits	Standard

Classic Mode

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

Save Cancel

Figure 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.

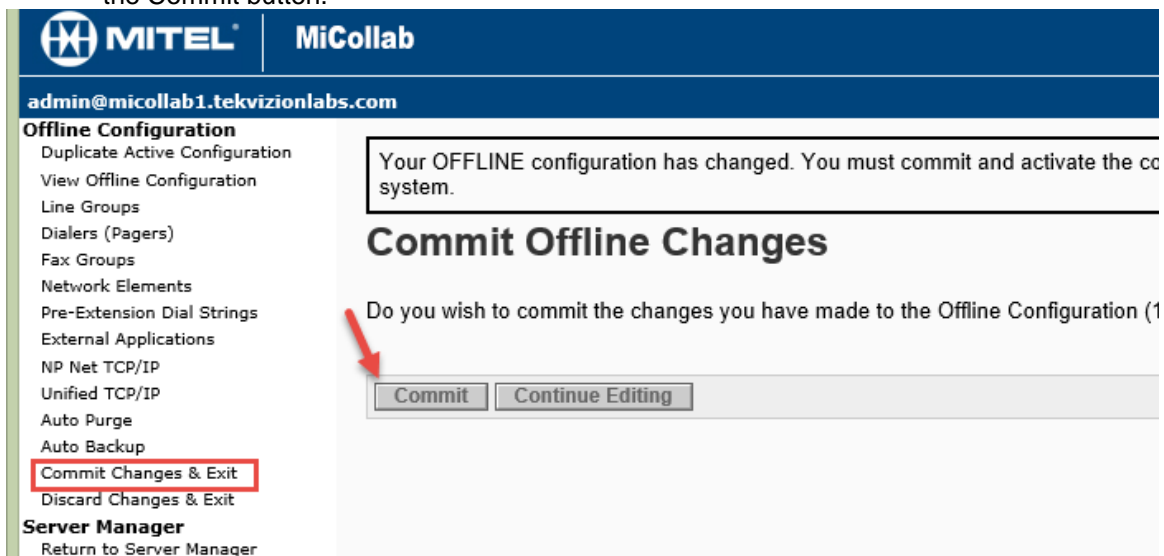


Figure 51 -- Committing Offline Changes

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**

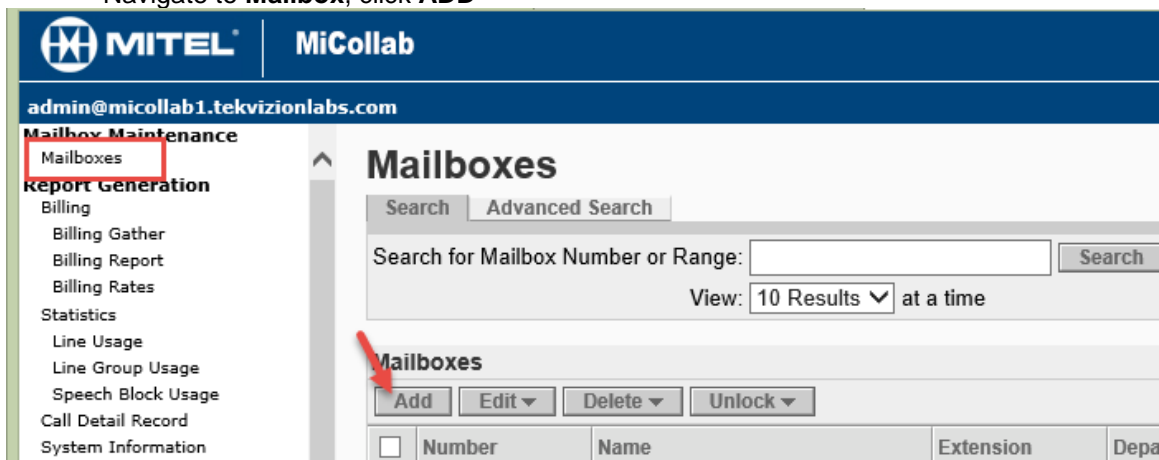


Figure 52 -- Add Mailbox

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

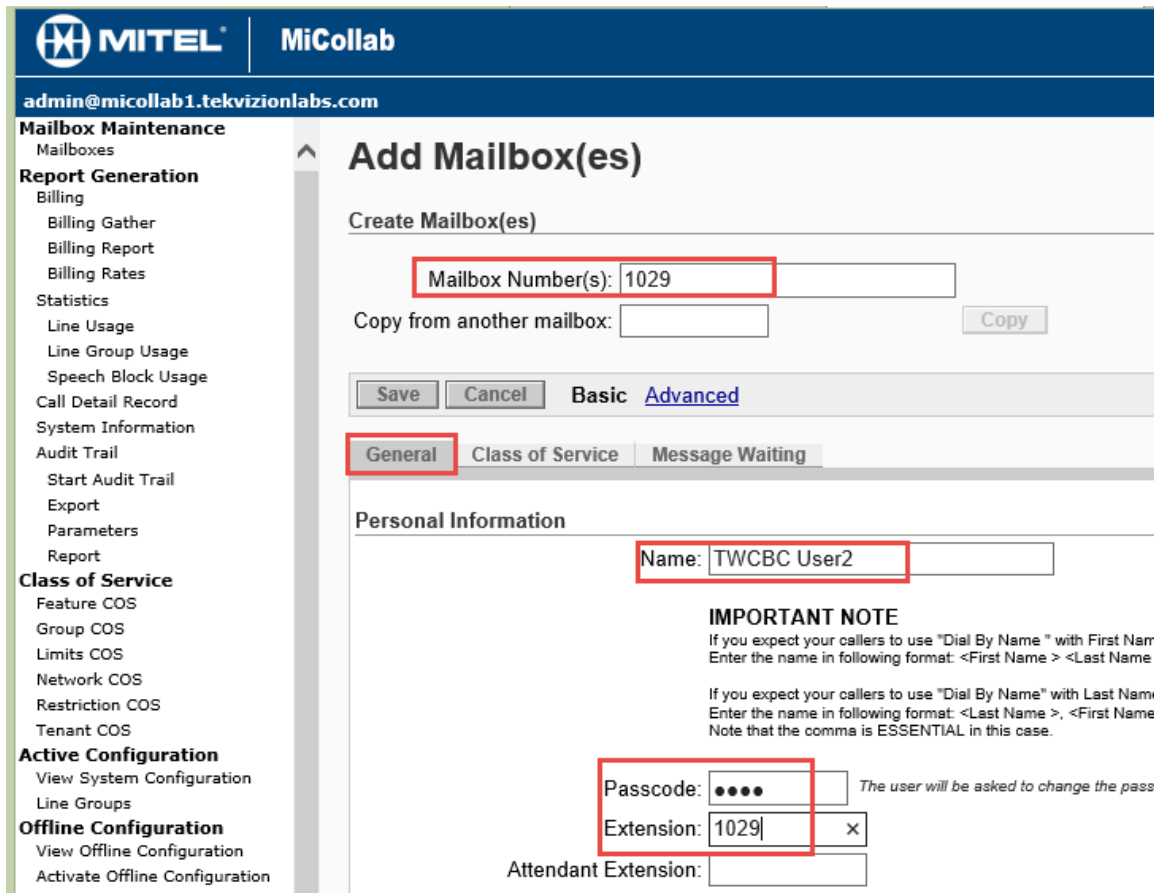


Figure 53 – Add Mailbox cont.

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

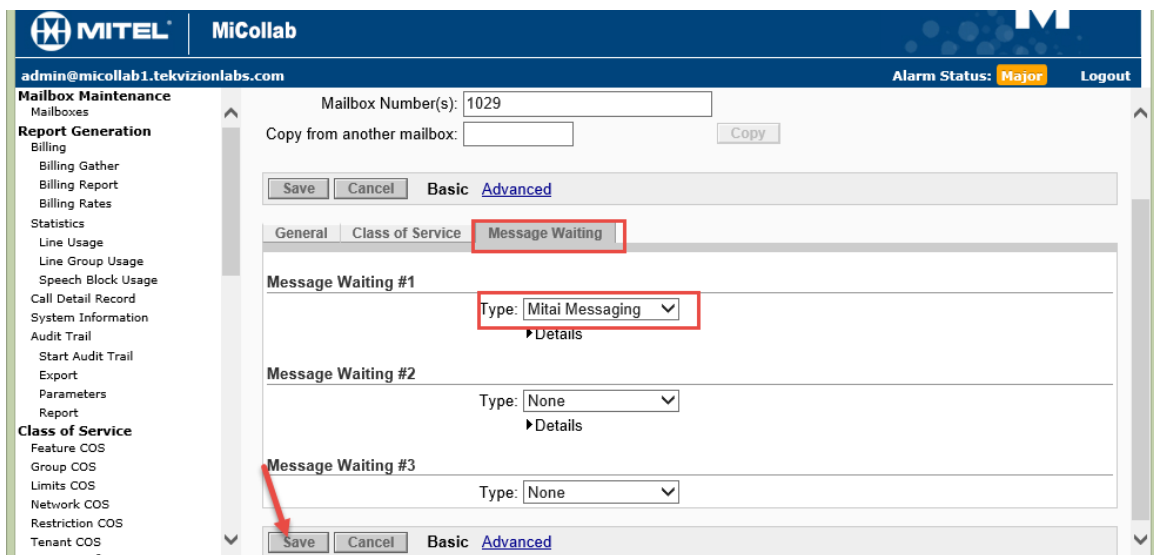


Figure 54 – Add Mailbox cont.

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 Server-gateway configuration on the network edge then click Apply.

The screenshot displays the Mitel Standard Linux administrative interface for a MiVoice Border Gateway. The left sidebar contains navigation menus for Applications, ServiceLink, Administration, Security, and Configuration. The main content area is titled 'Manage MiVoice Border Gateway' and includes tabs for Status, Configuration, Services, Applications, and Clustering. Under the Configuration tab, there are sub-tabs for Settings, Network profiles, ICPs, Bandwidth management, Alarms, and Overrides. The 'Network profiles' sub-tab is active, showing a list of configuration options: 'Server-gateway configuration on the network edge', 'Server-only configuration on the network DMZ', 'Server-only configuration on the network LAN', and 'Custom configuration'. Each option has a right-pointing arrow button. A red box highlights the 'Server-gateway configuration on the network edge' option, and another red box highlights the 'Apply S/G configuration' button. A red arrow points to the 'Apply' button. Below the configuration options, there is a 'Put Into Daisychain Mode' button.

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](#).

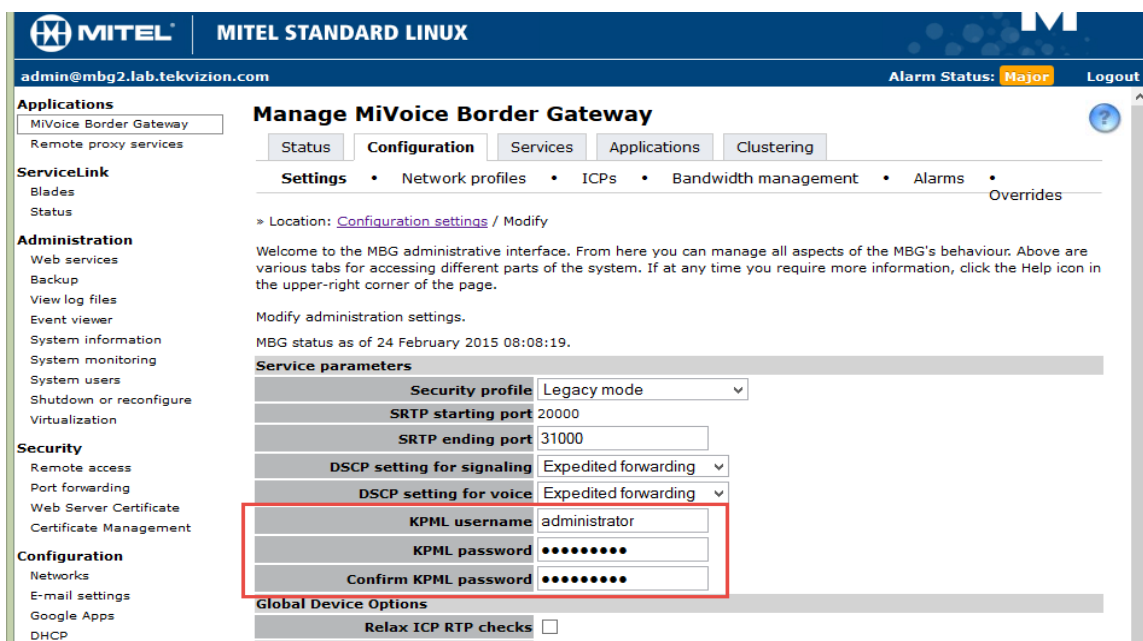


Figure 56 – MBG Settings

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**

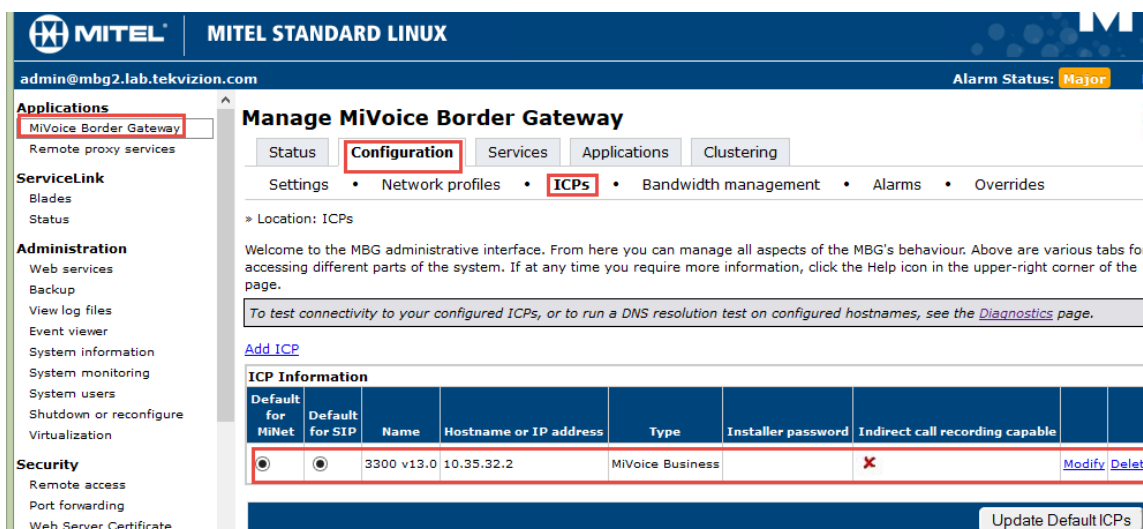


Figure 57 – MIVOICE BORDER GATEWAY's Configuration page

- 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.

The screenshot displays the Mitel Standard Linux web interface for managing the MiVoice Border Gateway. The 'Services' tab is active, and the 'SIP trunking' sub-tab is selected. A table titled 'SIP trunk information' lists the following details for the 'TWCBC' trunk:

Name	Trunk status	Calls in progress / Max	Calls per hour / Max	PRACK support	Remote RTP framesize (ms)	Re-invite filtering	RTP address override	Local streaming
TWCBC	Up	0 / 4	0 / 514	Disabled	20	Off	10.65.1.20	False

Below the table, there are links for 'Add a SIP trunk', 'Modify', 'Delete', and 'Reset metrics'. The page also shows a navigation menu on the left and a top status bar with 'Alarm Status: Major'.

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.

MITEL STANDARD LINUX

admin@mbg2.lab.tekvision.com Alarm Status: Major

Applications | Status | Configuration | **Services** | Applications | Clustering

MI-Net devices • Device settings by DN • SIP devices • **SIP trunking** • Recording status

Location: SIP Trunks / View SIP Trunk - TWCBC / Edit SIP Trunk - TWCBC

Welcome to the MBG administrative interface. From here you can manage all aspects of the MBG's behaviour. Above are various tabs for accessing different parts of the system. If at any time you require more information, click the Help icon in the upper-right corner of the page.

This interface provides the ability to edit a SIP trunk's details. Edit below, and click the "Save" button to commit the changes. If you do not wish to save, simply navigate elsewhere.

Name:	TWCBC
Remote trunk endpoint address:	10.65.1.200
Remote trunk endpoint port:	5060
Options keepalives:	Always
Options interval:	60
Rewrite host in PAI:	<input checked="" type="checkbox"/>
Remote RTP framesize (ms):	20ms
Idle timeout (s):	3600
Re-invite filtering:	Off
RTP address override:	LAN Interface - 10.65.1.20
Local streaming:	<input type="checkbox"/>
PRACK support:	Disabled
Log verbosity:	Use master setting
Authentication usernames:	
Authentication password:	
Confirm authentication password:	

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

Rules per page: 10

Routing rules: First Prev Page 1 of 1 Jump to page 1 Next Last

Match	Rule	Primary	Secondary	
1 Request URI	46993XXXX	3300 v13.0	-----	Raise Presend Delete Lower Append

Save

Figure 59 – SIP Trunk configuration settings



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