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

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

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

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

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Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVoice Business to connect to First Communications SIP Trunking. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic setup with required option setup.

Interop History

Version	Date	Reason
1	05-September-2018	Initial Interop with Mitel MiVoice Business Release 8.0 SP3 PR1 Software Load 14.0.3.22 and First Communications SIP Trunking

Interop Status

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

	The most common certification which means First Communications SIP Trunking has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.
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Software & Hardware Setup

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

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 8.0 SP3 PR1 Software Load 14.0.3.22
Mitel	MBG – Teleworker	v10.0.2.14
Mitel	Minet Sets: 5312, 5320, 5330	Minet 5312 (06.05.00.11) Minet 5320 (06.05.00.11) Minet 5330 (06.03.03.08)
Service Provider	First Communications	N/A

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Trunk Side Interoperability Test Plans (08-4940-00034) for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through First Communications and their PSTN gateway, call holding, transferring, conferencing, busy calls, DTMF RFC 2833, In-band, long calls durations, variable codec, G.711 and G.729 codec, Privacy, Loop back Calling, Long Ringing	☑
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
NuPoint Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	☑
Packetization	Forcing the Mitel MiVoice Business to stream RTP packets through its E2T card at different intervals, from 10ms to 90ms	⚠
Personal Ring Groups	Receiving calls through First Communications and their PSTN gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑
External Hot Desking	Receiving calls through First Communications and their PSTN gateway to PRG with EHDU. Including moving calls to/from the prime member of the PRG with the EHDU. Also placing calls from the EHDU and using mid call features with EHDU.	☑
Teleworker	Making and receiving a call First Communications and their PSTN gateway to and from Teleworker extensions.	☑
Fax	T.38 and G711 Fax Calls	⚠

☑ - No issues found

☒ - Issues found, cannot recommend to use

⚠ - Issues found

Device Limitations and Known Issues

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

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

Network Topology

This diagram shows how the testing network is configured for reference

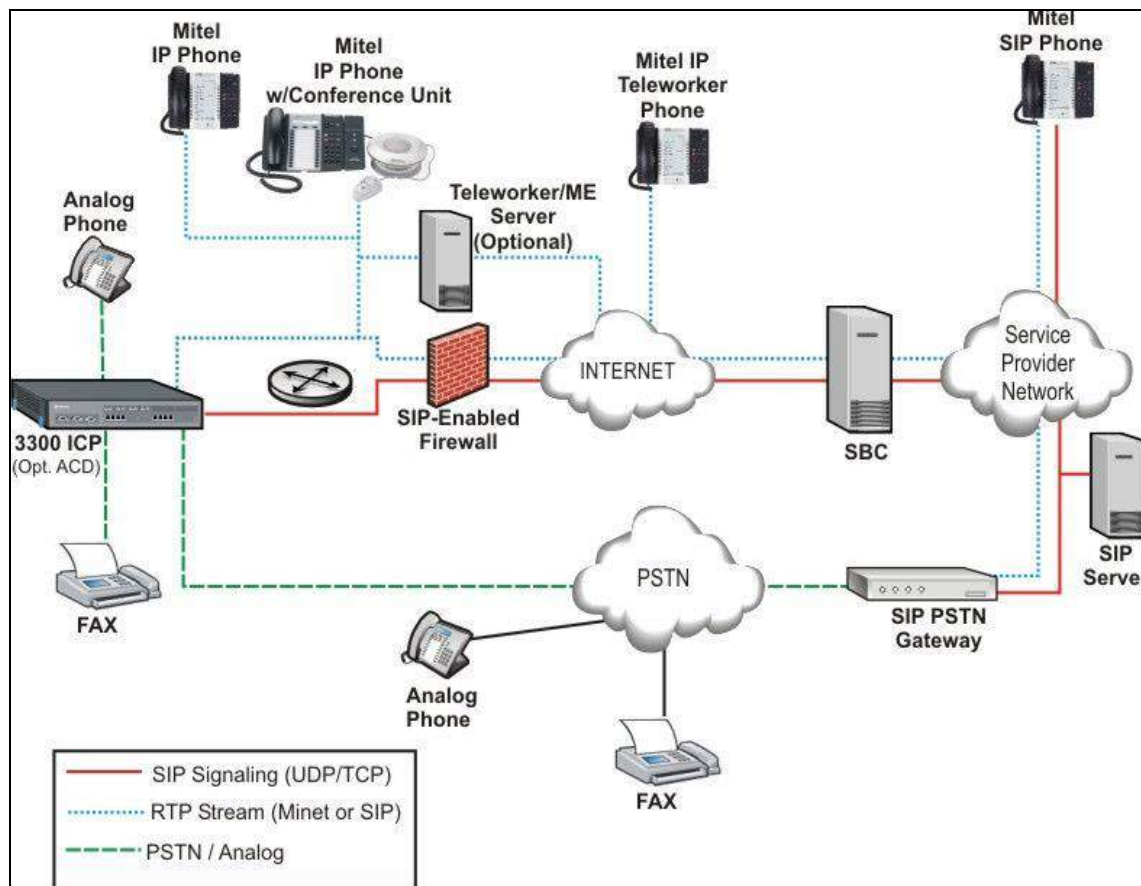


Figure 1: Network Topology

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline how a device can be configured in a customer environment and how First Communications SIP Trunking MiVoice Business programming was configured in our test environment.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

MIVOICE BUSINESS Configuration Notes

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

Configuration Template

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

Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx. 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms Packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx. 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the MiVoice Business Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

Assumptions for MiVoice Business Programming

The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

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

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

License and Option Selection

Application Record ID 26682859

System Type	License Sharing		Hardware Identifier			
Enterprise	No		0000003a1a4f			
Local Limits						
Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Licenses Allowed	Can be Over Allocated
Users						
IP Users	16	16	0	16	Unrestricted	Yes
External Hot Desk Users	3	7	3	10	Unrestricted	Yes
ACD Active Agents	2	10	0	10	Unrestricted	No
HTML Applications	0	0	20	0	Unrestricted	Yes
Single Line Users	0	16	0	16	Unrestricted	Yes
MiVoice Business Console Active Operators	0	0	20	0	Unrestricted	No
Multi-device Users	3	5	0	5	Unrestricted	Yes
Multi-device Suites	0	0	5	5	0	No
Messaging						
Embedded Voice Mail	2	16	0	16	Unrestricted	Yes
Embedded Voice Mail PMS	1	Yes	0	1	Unrestricted	Yes
Trunking / Networking						
Digital Links	0	1	0	1	Unrestricted	Yes
Compression		8	0	8	Unrestricted	Yes
FAX Over IP (T.38)		4	0	4	Unrestricted	Yes
SIP Trunks	5	353	0	353	Unrestricted	Yes
Others						
IDS Connection	1	Yes	0	1	Unrestricted	Yes
MLPP	0	No	0	0	Unrestricted	No
Configuration Options						
Country	North America					

Figure 2 – License and Option Selection

Extended Agent Skill Group	No
Maximum Elements per Cluster	30
Maximum Configurable IP Users and Devices	700
Extended Hunt Group	Yes
5560 IPT Device Extended Key Lines	No

Figure 3 - License and Option Selection – Contd.

Class of Service Assignment

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

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

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

The screenshot displays the 'Class of Service Options' configuration page for 'Local_2'. The left-hand navigation pane includes sections for Licenses, LAN/WAN Configuration, and Voice Network, with 'System Properties', 'System Feature Settings', and 'Class of Service Options' highlighted. The main area shows a table of Class of Service Options with the following entries:

13	JT_Global
14	SIP Trunk_IPC
15	CableONE
16	FirstComm
17	Crash

Below the table, the 'General' tab is selected, showing the following configuration details for Class of Service Number 16:

Class Of Service Number	16
Comment	FirstComm
ACD	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
ACD Make Busy on Login	No
ACD Silent Monitor Accept	No

Figure 4 – Class of Service

Class of Service for SIP Trunk

General	Advanced
Class Of Service Number	16
Comment	FirstComm
ACD	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
ACD Make Busy on Login	No
ACD Silent Monitor Accept	No
ACD Silent Monitor Accept Monitoring Non-Prime Lines	No
ACD Silent Monitor Allowed	No
ACD Silent Monitor Notification	No
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	No
Work Timer	0
Announce	
Call Announce Line	No
Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack Allowed	No
Busy Override	
Busy Override Security	No
Disable Executive Busy Override Tone	No
Executive Busy Override	No
Call Control Timer	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15

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

Inter Digit Timer	10
Lockout Timer	45
Call Duration	
Call Duration	10
Call Duration Forced Cleardown Timer	0
Enable Call Duration Limit on External Calls	No
Enable Call Duration Limit on Internal Calls	No
Call Forwarding/Rerouting	
Call Forward - Delay	0
Call Forward No Answer Timer	15
Call Forward Override	No
Call Forwarding (External Destination)	Yes
Call Forwarding (Internal Destination)	Yes
Call Forwarding Accept	Yes
Call Reroute after CFFM to Busy Destination	No
Call Forwarding Reminder Ring (CFFM and CFIAH only)	No
Disable Call Reroute Chaining On Diversion	No
Follow Reroute on Disabled Forwarding	No
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No
Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes

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

Call Hold	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	Yes
Call Hold Remote Retrieve	Yes
Call Hold Timer	
Local Music On Hold source	Yes
Music on Hold on Transfer	No
Use Called Party Call Hold Timer	No
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	Yes
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Pickup Display	No
Call Privacy	
Call Privacy	No
Calling Party Name Substitution	Yes
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	Yes
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes

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

Campon

Auto Campon Timer

Campon Recall Timer 0

Direct Voice Call

Direct Voice Call - Accept No

Direct Voice Call - Allow No

Direct Voice Call - Maximize Volume No

Display

After Answer Display Time

Calling Name Display - Internal - ONS Yes

Calling Number Display - Internal - ONS Yes

Display ANI/DNIS/ISDN Calling/Called Number No

Display ANI/ISDN Calling Number Only No

Display Caller ID on multicall/keylines No

Display Caller ID On Multicall/Keylines Timer 5

Display Caller ID On Single Line Displays For Forwarded Calls No

Display Dialed Digits during Outgoing Calls No

Display DNIS/Called Number Before Digit Modification No

Display DNIS on Key Label No

Display Held Call ID on Transfer No

Display Transfer Destination on Recall No

Hot Desk External User - Display Internal Calling ID No

Maintain Ringing Party During Recall No

Non-Prime Public Network Identity No

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

Originator's Display Update In Call Forwarding/Rerouting	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	No
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No
Fax	
Campan Tone Security	Yes
External Trunk Standard Ringback	No
Fax Capable	Yes
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
Hot Desk	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk Auto Logout Timer	0
Hot Desk External User - Allow Mid-Call Features	Yes
Hot Desk External User - Answer Confirmation	No
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	Yes
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No

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

Miscellaneous

Backlighting - Enabled	Yes
Clear All Features Remote	No
Enbloc Dialing - Enabled	No
Force Device Busy If Any Line In Use	No
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Integrated DECT High Power - Enabled	Yes
Integrated DECT Wideband - Enabled	Yes
Long Key Press Timer	0
Multi-Color LED Support - Disable	No
Phone Lock	No
Reseize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0

Paging

Group Page Accept	No
Group Page Allow	No
Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes
Pager Access All Zones	Yes
Pager Access Individual Zones	No

PC Port

PC Port On IP Device - Disable	No
---------------------------------------	----

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

RAD

Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No

Ringin

Delay Ring Timer	10
No Answer Recall Timer	17
Ringin Line Select	No
Ringin Timer	180

SMDR

SMDR External	No
SMDR Internal	No

Trunk

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

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

Trunk Flash Allowed	Yes
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5

Figure 12: Class of Service (General) – Contd.

Account Code

Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes

Attendant

Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No

Call Screening

BLF Screening Allow	No
BLF Screening Accept	No

Conference

Conference Call	Yes
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

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

Group Presence	
Group Presence Control	No
Group Presence Third Party Control	No
Hotel	
Display VIP	No
Hotel Room Monitor Setup Allowed	No
Hotel Room Monitoring Allowed	No
Hotel/Motel Room Personal Wakeup Call Allowed	No
Hotel/Motel Room Remote Wakeup Call Allowed	No
Message Waiting	
Message Waiting	Yes
Message Waiting - Disable Ringing Lamp Notification	No
Message Waiting Audible Tone Notification	No
Message Waiting Deactivate On Off-Hook	Yes
Message Waiting Inquire	Yes
Message Waiting Ringing Start Time Hour	
Message Waiting Ringing Start Time Minute	
Message Waiting Ringing Stop Time Hour	
Message Waiting Ringing Stop Time Minute	
Multiline Set Voice Mail Callback Message Erasure Allowed	No
ONS CLASS/CLIP: Message Waiting Activate/Deactivate	No
Miscellaneous	
Auto Answer Allowed	Yes
Auto Release on Key Select	No
Brokers Call	No
Called Party Features Override	No

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

Check COR after PSTN Dial Tone	No
Dialled Night Service	Yes
Disable Send Message	No
Flexible Answer Point	No
Individual Trunk Access	Yes
Key A	
Key B	
Key C	
Key D	
Multiline Set Loop Test	No
Multiline Set Message Center Remote Read Allowed	No
Multiline Set Music	No
Multiline Set On-hook Dialing	Yes
Multiline Set Phonebook Allowed	Yes
Non DID Extension	No
ONS CLASS/CLIP: Set	No
ONS/OPS Internal Ring Cadence for External Callers	No
Override Interconnect Restriction on Transfer	No
Recall If Transferred to Original Call Destination	No
Redial Facilities	Yes
Use Default Billable Number For Trunk Calls	No
Voice Dial Preferred	No
Voice Mail Softkey	No

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

Phonebook

Phonebook Lookup - Default to User Location	No
Phonebook Lookup - Display User Location	No

Record A Call

Record-A-Call - Save Recording on Hang-up	No
Record-A-Call - Start Automatic Incoming Call Recording	No
Record-A-Call - Start Automatic Outgoing External Call Recording	No
Record-A-Call Active	No

Figure 16: Class of Service (Advanced) – Contd.

Class of Service for Phones

General	Advanced
Class Of Service Number	11
Comment	MitelPhone_FC
ACD	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
ACD Make Busy on Login	Yes
ACD Silent Monitor Accept	No
ACD Silent Monitor Accept Monitoring Non-Prime Lines	No
ACD Silent Monitor Allowed	No
ACD Silent Monitor Notification	No
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	No
Work Timer	0

Figure 17: Class of Service (General)

Announce	
Call Announce Line	No
Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack Allowed	No
Busy Override	
Busy Override Security	No
Disable Executive Busy Override Tone	No
Executive Busy Override	No
Call Control Timer	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10
Lockout Timer	45
Call Duration	
Call Duration	10
Call Duration Forced Cleardown Timer	0
Enable Call Duration Limit on External Calls	No
Enable Call Duration Limit on Internal Calls	No
Call Forwarding/Rerouting	
Call Forward - Delay	0
Call Forward No Answer Timer	15
Call Forward Override	Yes
Call Forwarding (External Destination)	Yes
Call Forwarding (Internal Destination)	Yes
Call Forwarding Accept	Yes

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

Call Forwarding Accept	Yes
Call Reroute after CFFM to Busy Destination	No
Call Forwarding Reminder Ring (CFFM and CFIAH only)	No
Disable Call Reroute Chaining On Diversion	No
Follow Reroute on Disabled Forwarding	Yes
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No
Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes
Call Hold	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	Yes
Call Hold Remote Retrieve	Yes
Call Hold Timer	30
Local Music On Hold source	Yes
Music on Hold on Transfer	Yes
Use Called Party Call Hold Timer	No
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	Yes
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Pickup Display	No

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

Call Privacy	
Call Privacy	No
Calling Party Name Substitution	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	Yes
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes
Campon	
Auto Campon Timer	
Campon Recall Timer	0
Direct Voice Call	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
Display	
After Answer Display Time	
Calling Name Display - Internal - ONS	Yes
Calling Number Display - Internal - ONS	Yes
Display ANI/DNIS/ISDN Calling/Called Number	Yes
Display ANI/ISDN Calling Number Only	Yes
Display Caller ID on multicall/keylines	Yes
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	No
Display Dialed Digits during Outgoing Calls	Yes

Figure 20: Class of Service (General) – Contd.

Display DNIS/Called Number Before Digit Modification	Yes
Display DNIS on Key Label	No
Display Held Call ID on Transfer	No
Display Transfer Destination on Recall	No
Hot Desk External User - Display Internal Calling ID	Yes
Maintain Ringing Party During Recall	No
Non-Prime Public Network Identity	No
Originator's Display Update In Call Forwarding/Rerouting	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	No
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No
Fax	
Campon Tone Security	No
External Trunk Standard Ringback	No
Fax Capable	No
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes
Hot Desk	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk Auto Logout Timer	0

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

Hot Desk External User - Allow Mid-Call Features	No
Hot Desk External User - Answer Confirmation	No
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	No
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No
Miscellaneous	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Enbloc Dialing - Enabled	No
Force Device Busy If Any Line In Use	Yes
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Integrated DECT High Power - Enabled	Yes
Integrated DECT Wideband - Enabled	Yes
Long Key Press Timer	0
Multi-Color LED Support - Disable	No
Phone Lock	No
Reseize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0
Paging	
Group Page Accept	No
Group Page Allow	No

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

Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes
Pager Access All Zones	Yes
Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
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

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

Public Trunk	Yes
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	Yes
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	Yes
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	Yes
ONS VMail-Delay Dial Tone Timer	5

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

General	Advanced
Account Code	
Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes
Attendant	
Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No
Call Screening	
BLF Screening Allow	No
BLF Screening Accept	No
Conference	
Conference Call	Yes

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

Disable Conference Join Tone	No
DND	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No
Emergency	
Emergency Call - Audio Level for Set	Ringer
Emergency Call Notification - Audio	No
Emergency Call Notification - Visual	No
Group Presence	
Group Presence Control	No
Group Presence Third Party Control	No
Hotel	
Display VIP	No
Hotel Room Monitor Setup Allowed	No
Hotel Room Monitoring Allowed	No
Hotel/Motel Room Personal Wakeup Call Allowed	No
Hotel/Motel Room Remote Wakeup Call Allowed	No
Message Waiting	
Message Waiting	Yes
Message Waiting - Disable Ringing Lamp Notification	No
Message Waiting Audible Tone Notification	No
Message Waiting Deactivate On Off-Hook	Yes
Message Waiting Inquire	Yes
Message Waiting Ringing Start Time Hour	
Message Waiting Ringing Start Time Minute	

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

Message Waiting Ringing Stop Time Hour	
Message Waiting Ringing Stop Time Minute	
Multiline Set Voice Mail Callback Message Erasure Allowed	Yes
ONS CLASS/CLIP: Message Waiting Activate/Deactivate	Yes
Miscellaneous	
Auto Answer Allowed	Yes
Auto Release on Key Select	No
Brokers Call	No
Called Party Features Override	No
Check COR after PSTN Dial Tone	No
Dialled Night Service	Yes
Disable Send Message	No
Flexible Answer Point	No
Individual Trunk Access	Yes
Key A	
Key B	
Key C	
Key D	
Multiline Set Loop Test	No
Multiline Set Message Center Remote Read Allowed	No
Multiline Set Music	No
Multiline Set On-hook Dialing	Yes
Multiline Set Phonebook Allowed	Yes

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

Non DID Extension	No
ONS CLASS/CLIP: Set	No
ONS/OPS Internal Ring Cadence for External Callers	No
Override Interconnect Restriction on Transfer	No
Recall If Transferred to Original Call Destination	No
Redial Facilities	Yes
Use Default Billable Number For Trunk Calls	No
Voice Dial Preferred	No
Voice Mail Softkey	Yes
Phonebook	
Phonebook Lookup - Default to User Location	No
Phonebook Lookup - Display User Location	No
Record A Call	
Record-A-Call - Save Recording on Hang-up	No
Record-A-Call - Start Automatic Incoming Call Recording	No
Record-A-Call - Start Automatic Outgoing External Call Recording	No
Record-A-Call Active	No

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

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

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

➔ **Network Elements**

Name	FirstComm
Type	Other
FQDN or IP Address	21 <input style="width: 100px;" type="text" value="0"/>
Data Sharing	NO
Local	False
Version	
Zone	2
ARID	
SIP Peer Specific	
SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	21 <input style="width: 100px;" type="text" value="0"/>
External SIP Proxy Transport	default
External SIP Proxy Port	0
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	default
SIP Registrar Port	0
SIP Peer Status	Auto-Detect/Normal

Figure 29 – Network Element Assignment

Network Element Assignment (Proxy)

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

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

Figure 30 – Network Element Assignment (Proxy)

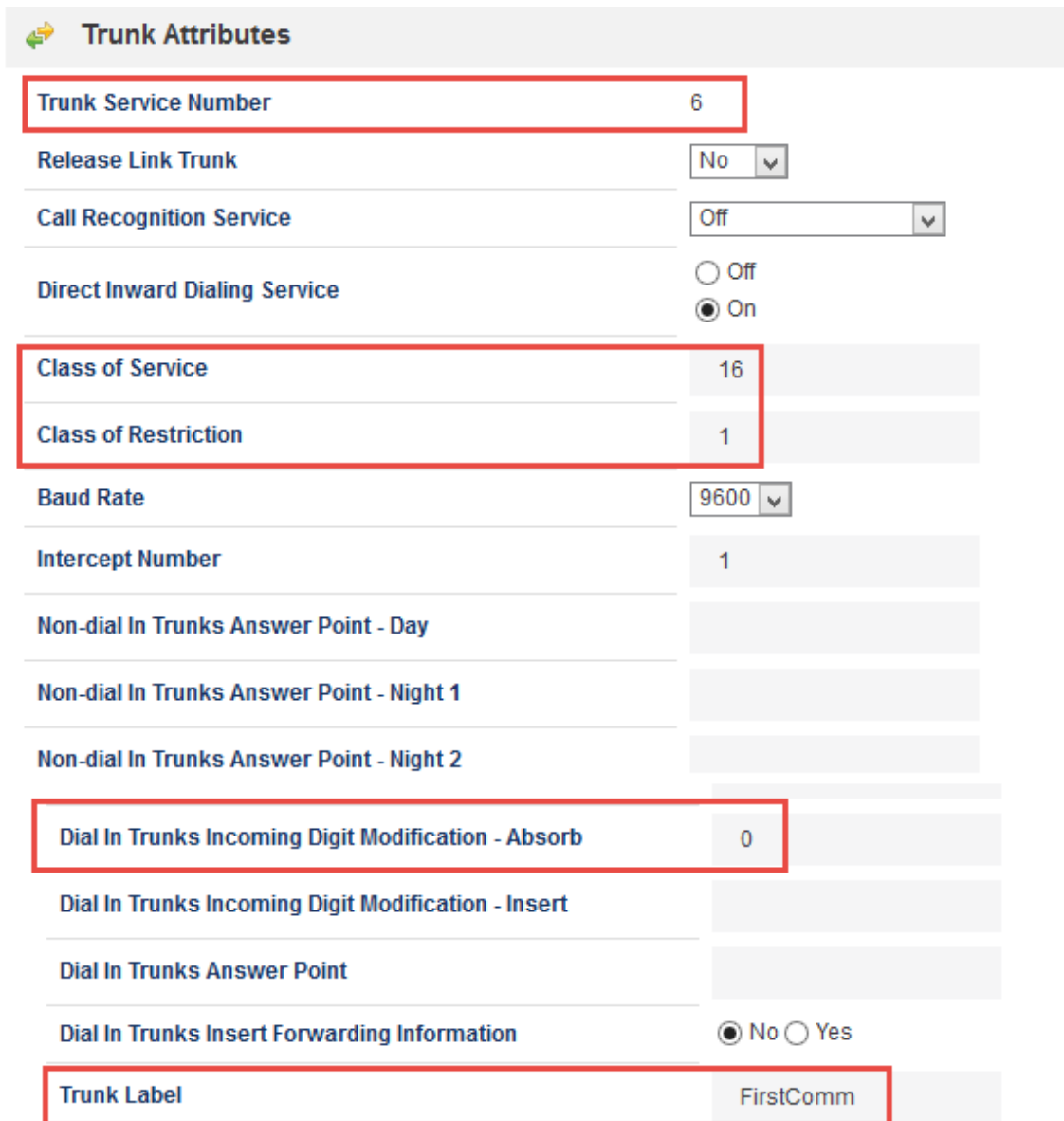
Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number 6 which will be used to direct incoming calls to an answer point in the Mitel MiVoice Business.

Program the Non-dial In or Dial in Trunks (DID) according to the site requirements and what type of service was ordered from your service provider.

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

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



Trunk Attributes	
Trunk Service Number	6
Release Link Trunk	No
Call Recognition Service	Off
Direct Inward Dialing Service	<input checked="" type="radio"/> On <input type="radio"/> Off
Class of Service	16
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	FirstComm

Figure 31 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces.

IP/Ethernet connectivity is part of the base MiVoice Business Platform. The SIP Peer Profile should be configured with the following options:

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

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

Address Type: Select IP address.

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

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

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

SMDR: If Call Detail Records are required for SIP Trunking, the SMDR Tag should be configured (by default there is no SMDR and this field is left blank).

Subscription User Name/Password: Enter user name and password which will be matched in later MBG configuration for KPML credentials under Configuration > Settings > Service Parameter. This is part of configuration for Mid Call features to function with KPML such as pressing 5 to handoff from the EHDU in the PRG (Personal Ring Groups).

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

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

SIP Peer Profile Label	FirstComm
Network Element	FirstComm

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	0
Outbound Proxy Server	FC_MBG
SMDR Tag	0
Trunk Service	6
Zone	1

Authentication Options

User Name	
Password	*****
Confirm Password	*****
Authentication Option for Incoming Calls	No Authentication
Subscription User Name	administrator
Subscription Password	*****
Subscription Confirm Password	*****

Figure 32 – SIP Peer Profile Assignment- Basic

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers
Key Press Event	Outgoing DID Ranges	Profile Information			
Alternate Destination Domain Enabled					No
Alternate Destination Domain FQDN or IP Address					
Enable Special Re-invite Collision Handling					No
Only Allow Outgoing Calls					No
Private SIP Trunk					No
Reject Incoming Anonymous Calls					No
Route Call Using P-Called-Party-ID (if present)					Yes
Route Call Using To Header					No

Figure 33: SIP Peer Profile Assignment- Call Routing

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers
Key Press Event	Outgoing DID Ranges	Profile Information			
Default CPN					
Default CPN Name					
CPN Restriction					No
Override From Header with Default CPN					No
Public Calling Party Number Passthrough					No
Strip PNI					No
Use Diverting Party Number as Calling Party Number					No
Use Original Calling Party Number If Available					No

Figure 34: SIP Peer Profile Assignment- Calling Line ID

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

Figure 35: SIP Peer Profile Assignment- SDP Options

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					No
Disable Use of User-Agent and Server Headers					No
Domain for Trunk Context					
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
Include Diversion Header for EHDU					No
Multilingual Name Display					No
Only use SDP to decide 180 or 183					Yes
Prefer From Header for Caller ID					No
Require Reliable Provisional Responses on Outgoing Calls					Yes
Signal Privacy (if enabled) on Emergency Calls					No
Suppress Redirection Headers					No
Use Fixed Retry Time for 491					No

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

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

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

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

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

Figure 38: SIP Peer Profile Assignment- Timers

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	No
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	default
KPML Port	0

Figure 39: SIP Peer Profile Assignment – Key Press Event

Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers

Key Press Event Outgoing DID Ranges Profile Information

Add Member Delete Member

Index DID Range CPN Substitution

Figure 40: SIP Peer Profile Assignment – Outgoing DID Ranges

Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers

Key Press Event Outgoing DID Ranges Profile Information

Creator

Date Created

Created with Version

Service Provider

Vendor Notes

Figure 41: SIP Peer Profile Assignment – Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from First Communications SIP trunk to a particular SIP Peer profile. The configured here settings help matching the incoming DID numbers with the SIP Peer Profile when call is arriving from anonymous caller.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, "2247175110 - 2247175114, and 2247175116"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 characters maximum, you can create a new entry for the same profile.

Use a '*' to reduce the number of entries that need to be programmed. This is a type of "prefix identifier", and cannot be used as a range with '-'. For example, the string "11*" would be used to associate a peer with any number in the range from 110 up to the maximum digits per telephone number (In this case, 11999999999999999999999999999999.) Note that the string "11" by itself would not count as a match, as the '*' represents 1 or more digits.

The screenshot shows a web form titled "Change SIP Peer Profile Assignment by Incoming DID". It contains three rows of input fields:

Incoming DID Range	2247175110-22471751*
SIP Peer Profile Label	FirstComm
Comment	FirstComm

Figure 42: SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

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

ARS Digit Modification Plans

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

Figure 43: Digit Modification Assignment

ARS Routes

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

ARS Routes

Route Number	5
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	FirstComm
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	1
Digits Before Outpulsing	
Route Type	PSTN Access Via DPNSS
Compression	Off

Figure 44: SIP Trunk Route Assignment

ARS Digits Dialed

ARS initiates the routing of trunk calls when certain digits are dialed from a station. In this

example, when a user dials (8214XXXXXX), access code 8 will be removed and the call will be routed to First Communications (i.e. Route 5).

Change Range Programming - ARS Digits Dialed Help

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

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
8	Unknown	Route	5

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 ▼	8	
Number of Digits to Follow	Change to ▼	Unknown ▼	-
Termination Type	Change to ▼	Route ▼	-
Termination Number	Change to ▼	5	

Preview
Save
Cancel

Figure 45: ARS Digit Dialed Assignment

T.38 Fax Configuration

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

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

Fax Service Profiles on **Local_2**

DN to search

Show form on

Change

Print...

Import...

Export...

Data Refresh

Inter-Zone Fax Profile

Maximum Fax Rate	14400 (V.17, 14400bps)
High Speed Redundancy	0
Low Speed Redundancy	0
Error Correction Mode (ECM)	Disabled

< Page 1 of 7 >

Go to Value

Go

Change Member

Change Page Members

Change All Members

Clear Member

Intra-Zone Fax Service Profiles

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 46: Fax Configuration

Zone Assignment

By default, all zones are set to Intra-zone FAX Profile 1.

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

Network Zones on **Local_2** DN to search Show form on

Change Change Page Clear Print... Import... Export... Data Refresh

< Page 1 of 50 > Go to Value Go

Network Zones

Zone ID	Intra-zone Compression	Group Zone	Intra-zone Fax Profile	Label	SMDR Tag	Time Zone	LBN Prefix	Zone CESID	Default Billing Number	Default CPN
1	No		1			America/Chicago				
2	Yes		2	T.38 Fax		America/Chicago				
3	No		1			America/Chicago				

Figure 47: Zone Assignment

NuPoint Configuration

MiVoice Business Setup for Connecting NuPoint

Licensing and Option Selection – SIP Licensing

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

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

Local_2

System Options on Local_2

DN to search

Show form on

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

System Options

Route Optimization Trailing Digits	2
Send Travelling Class Marks	No
Send Welcome Email	No
Set Registration Access Code	***
Set Registration Auto DN Selection - Begin	
Set Registration Auto DN Selection - End	
Set Registration Auto DN Selection - Secondary	Not Assigned
Set Registration Security	
Set Replacement Access Code	###
Site Preference for Hot Desk Device	5020 IP
Speed Call Pause Duration	3
SUPERSET Callback Message Cancel Timer	
System Data Synchronization	Yes

Figure 48: system options

Class of Service Options

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

In Class of Service for NuPoint Voicemail enable the following:

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

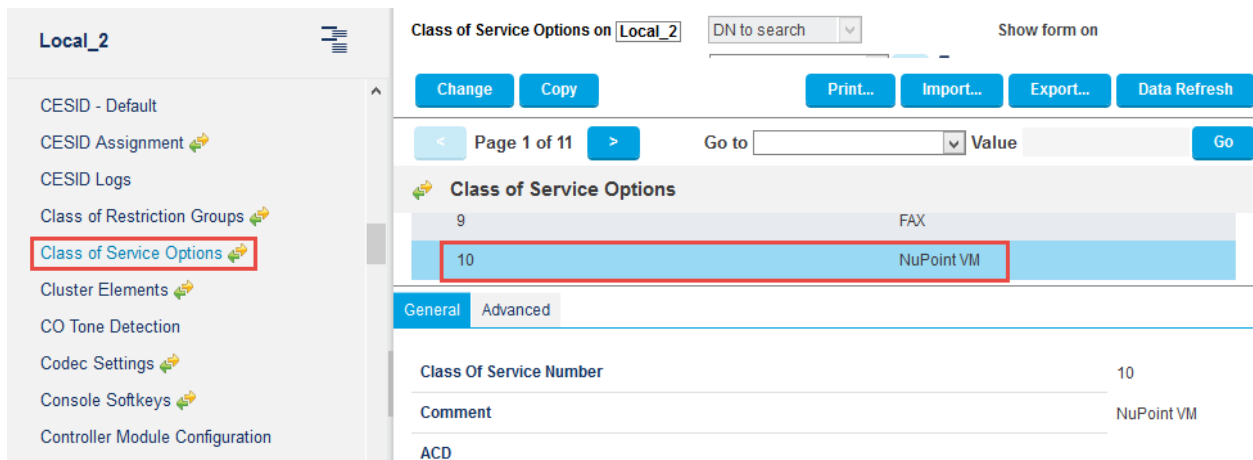


Figure 49: NuPoint Class of Service

IP Endpoints used for NuPoint Ports

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

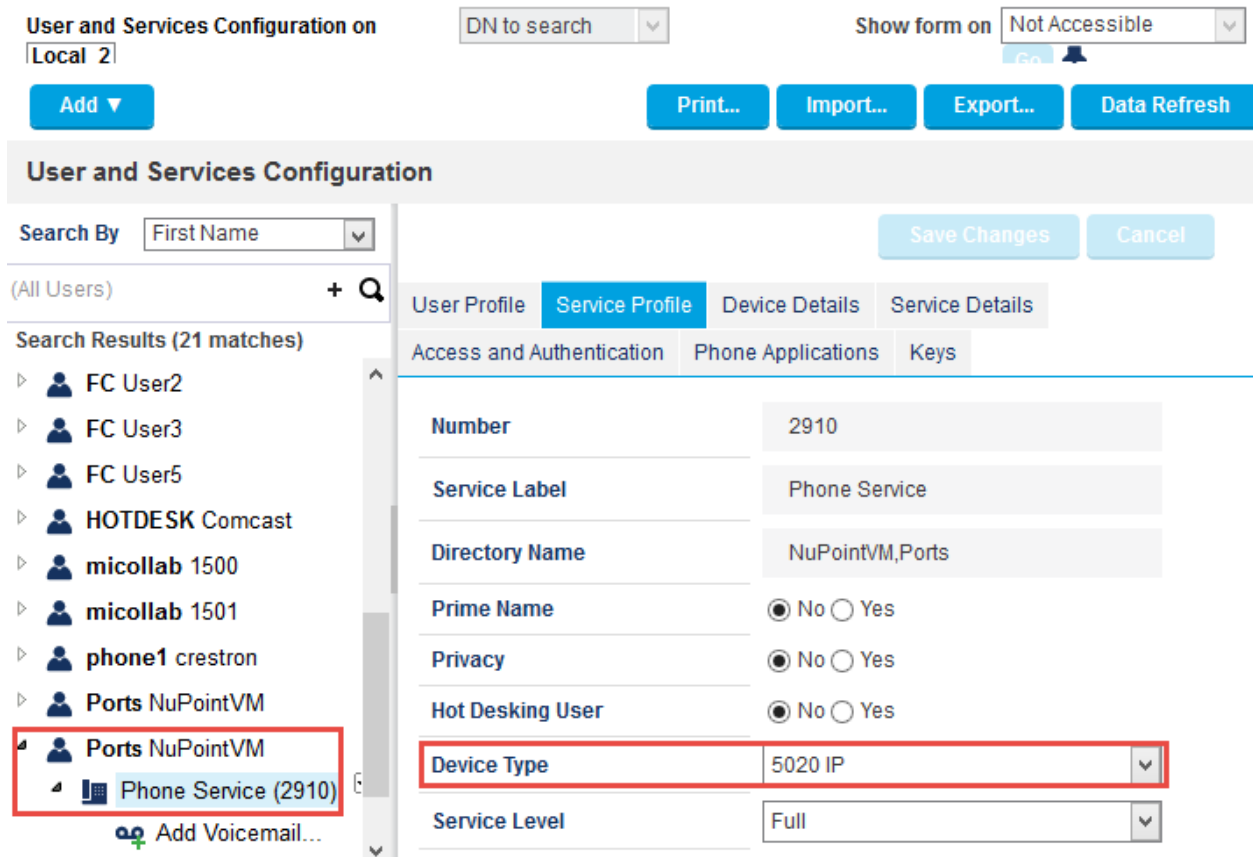


Figure 50: NuPoint Endpoint Configuration

Class of Service value for Day, Night 1 and Night 2 of the IP end point should be given the

Class of Service of incoming ports created earlier, which is 10.

User and Services Configuration on Local 2

DN to search [v] Show form on Not Accessible [v]

Add [v] Print... Import... Export... Data Refresh

User and Services Configuration

Search By First Name [v] Save Changes Cancel

(All Users) + Q

Search Results (21 matches)

- FC User5
- HOTDESK Comcast
- micollab 1500
- micollab 1501
- phone1 crestron
- Ports NuPointVM
- Ports NuPointVM**
- Phone Service (2910)

Access and Authentication Phone Applications Keys

	Day	Night 1	Night 2
Class of Service	10	10	10
Class of Restriction	1	1	1
External Hot Desking Enabled	<input checked="" type="radio"/> No <input type="radio"/> Yes		
External Hot Desking Dialing Prefix			
External Hot Desking Number			

Figure 51: NuPoint Endpoint Configuration – Contd.

Voicemail Hunt Group

Create a Voicemail Hunt Group that will be used to call voicemail. All of the endpoints created in the section above will be added to this hunt group. Enter the hunt group number that will be used for voicemail and change the Hunt Group type to Voicemail. Here, Hunt Group 3000 is created.

Hunt Groups on **Local_2** DN to search Show form on **Not Accessible**

[Go](#)

[Add](#) [Change](#) [Copy](#) [Delete](#) [Print...](#) [Import...](#) [Export...](#) [Data Refresh](#)

Hunt Groups

Number	Mode	Size	Mode	Location	Status
2900	Circular	64	Voice	Local_2	Not Assigne
3000	Circular	64	VoiceMail	Local_2	Not Assigne
3500	Circular	64	Voice	Local_2	Not Assigne

Hunt Group 3000

Local-only DN False

Hunt Group Mode Circular

Hunt Group Name

< Page 1 of 1 > Go to Value [Go](#)

[Add Member](#) [Change Member](#) [Delete Member](#)

Hunt Group Members

Member Index	Number	Presence	Name	Home Element	Secondary Element
1	2910	Present	NuPointVM,Ports	Local_2	
2	2911	Present	NuPointVM,Ports	Local_2	

Figure 52: Voicemail Hunt Group Configuration

HCI Reroute Hunt Group

Program the HCIReroute Hunt Group and set it to always route to the NuPoint Voicemail Hunt Group. The primary reason for setting up an HCIReroute is to enable MiTAI for MWI. 3999 is configured as HCIReroute Hunt Group in this test and Call Rerouting Always Alternative number 2 was modified to reroute everything to the Voicemail Hunt Group.

Hunt Groups on **Local_2** DN to search Show form on **Not Accessible**

Add Change Copy Delete Print... Import... Export... Data Refresh

Hunt Groups

Number	Mode	Members	Home Element	Local-only DN	Mode	Assigned To
3600	Circular	64	VoiceMail	Local_2		Not Assigned
3999	Circular	64	HCIReroute	Local_2		Not Assigned

Hunt Group 3999

Local-only DN False

Hunt Group Mode Circular

Hunt Group Name

Page 1 of 1 Go to Value Go

Add Member Change Member Delete Member

Figure 53: HCIReroute Hunt Group

Local_2 Call Rerouting Always Alternatives on **Local_2** DN to search Show form on

Change Change Page Change All Clear

Page 1 of 12 Go to Value

Call Rerouting Always Alternatives

Always Alternative Number	Originating Device DID	Originating Device TIE	Originating Device CO	Originating Device INT	Directory Number
1	No Reroute	No Reroute	No Reroute	No Reroute	
2	Reroute	Reroute	Reroute	Reroute	3000
3	No Reroute	No Reroute	No Reroute	No Reroute	
4	No Reroute	No Reroute	No Reroute	No Reroute	

Figure 54: Call Rerouting Always Alternative

Local_2

Call Rerouting on Local_2

DN to search

Show form on

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

Page 2 of 2 Go to Value

Call Rerouting

3600	1	1	1	All	1	1
3999	2	2	2	All	2	2
5001	1	1	1	All	1	1
5005	1	1	1	All	1	1
5006	1	1	1	All	1	1

Figure 55: Call Rerouting

MiCollab NuPoint Configuration

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

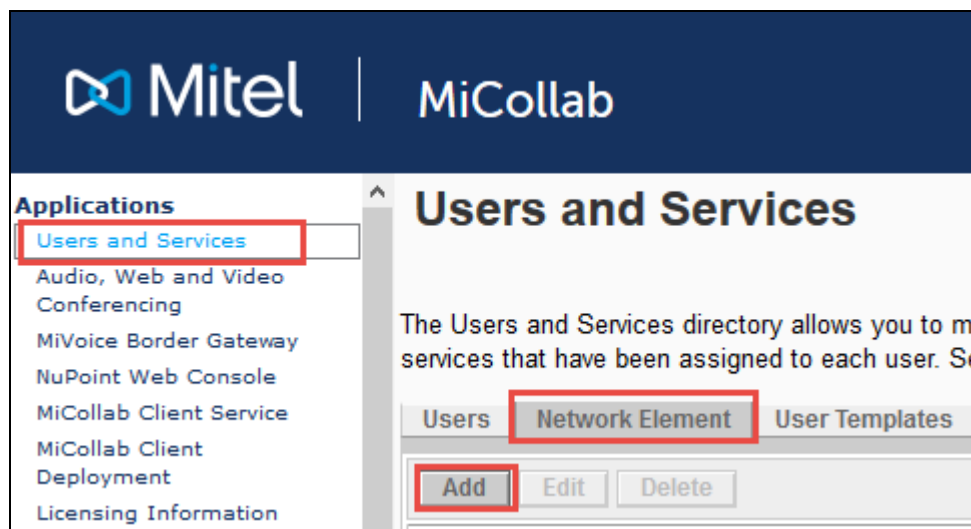


Figure 56: Add Network Element

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

Create Network Element



Element Identification

Type: MiVoice Business

*System Name: 3300

*IP Address/FQDN: 10.35.32.2

*Zone: 1

Network Element Settings

SIP Conference FAC: *40

Credentials

*System Login: system

*Password:

*Confirm Password:

System Properties

*Set Registration Code: ***

*Set Replacement Code: ###

Maximum IP Inteartion Licenses Reached

Voicemail

Call Reroute First Alternative Number: 1

Call Forward Destination Directory Number: 3000

HCI Reroute Hunt Group Number for Mitai MWI:

Figure 57: Network Element – Contd.

- Click On **NuPoint Web Console**

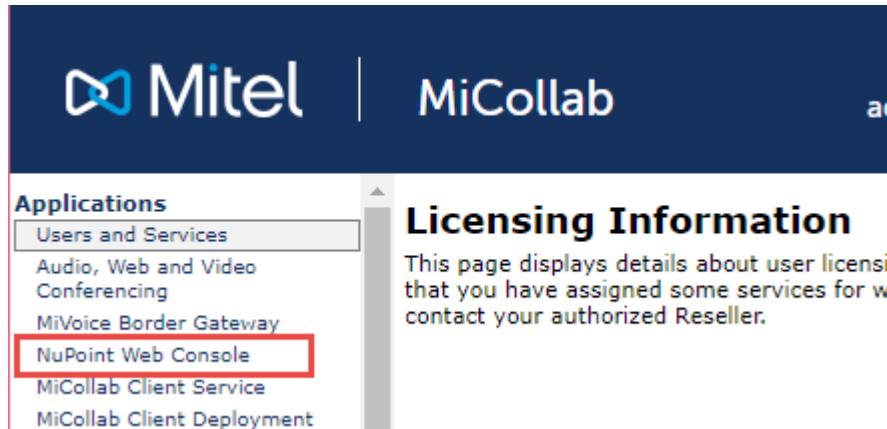


Figure 58: Voicemail Line Group Configuration

- Click **Add**



Figure 59: Voicemail Line Group Configuration – Contd.

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

Save Cancel

Line Group Number: 1

Name: VM *

Application: NuPoint Voice

User Interface: NuPoint Voice

Fax group connection: None

Lines Dialing Plan Voicemail Dial Strings

Lines

Add Edit Delete

Line Triplet	Device	Extension or Port
--------------	--------	-------------------

Figure 60: Adding Line Group

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

Lines Dialing Plan Voicemail Dial Strings

Lines

Add Edit Delete

Line Triplet	Device	Extension or Port
<input checked="" type="checkbox"/> 1:0:0	3300	2910
<input type="checkbox"/> 1:0:0		2911

Line Triplet: 1:0:0

PBX: 3300

Mapping: 2910

Save Save Cancel

Figure 61: Adding Lines

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

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

- Click **Save**

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 62: Adding Dial Plan

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

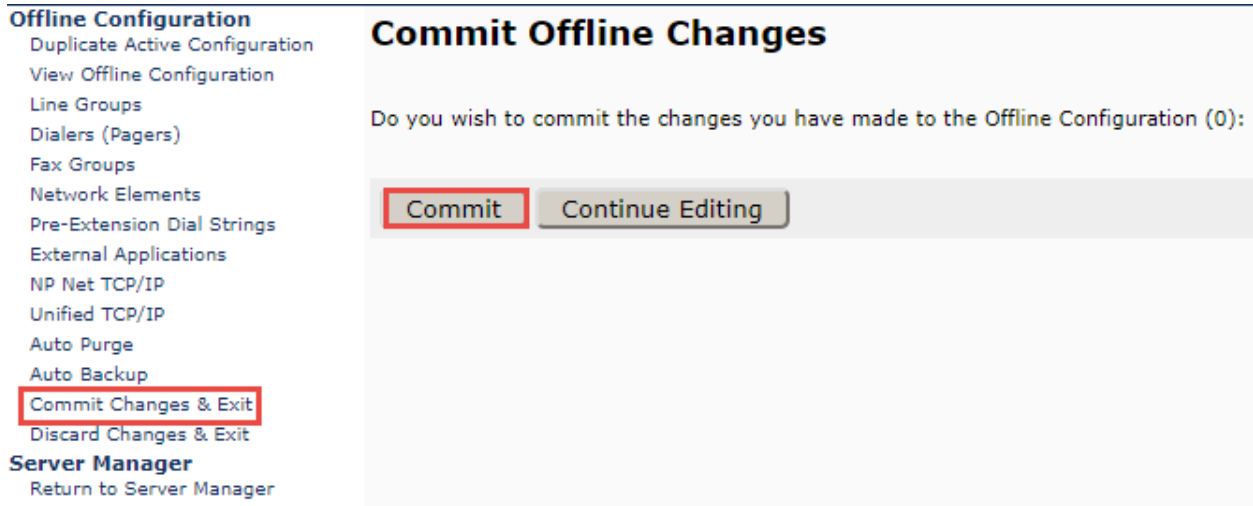


Figure 63: Committing Offline Changes

Adding Mailboxes

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

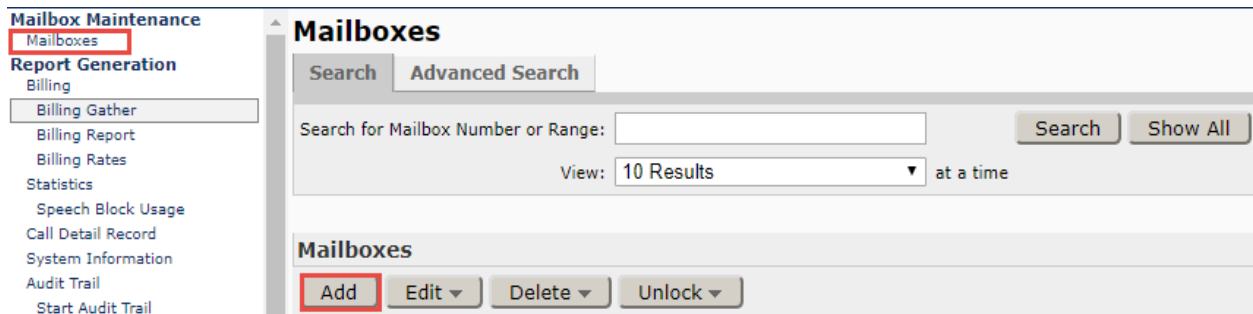


Figure 64: Add Mailbox

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

Add Mailbox(es)



Create Mailbox(es)

Mailbox Number(s): 5112

Copy from another mailbox:

Copy

Save

Cancel

Basic

[Advanced](#)

General

Class of Service

Message Waiting

Personal Information

Name: NuPoint,FC

IMPORTANT NOTE

Please enter names using the following syntax:

<First Name> <Last Name>

or

<Last Name>, <First Name>

The space or the comma between the two names is essential.

Passcode:

The user will be asked to change the passcode on the next TUI login

Extension:

5112

Attendant Extension:

Unified Messaging Information

UM Audio Encoding:

ADPCM (Smallest files, default \ ▼

UM-SMTP Email Address:

UM-Web View Email Address:

Save

Cancel

Basic

[Advanced](#)

Figure 65: Add Mailbox – Contd.

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

Add Mailbox(es)

Create Mailbox(es)

Mailbox Number(s):

Copy from another mailbox:

Basic [Advanced](#)

General **Class of Service** **Message Waiting**

Message Waiting #1

Type: ▾

▸ Details

Message Waiting #2

Type: ▾

▸ Details

Message Waiting #3

Type: ▾

Basic [Advanced](#)

Figure 66: Add Mailbox – Cont.

MiCollab Client Configuration

MiCollab Client Setup on MiVoice Business

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

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

SIP Device Capabilities on **Local_2** Show form on

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

SIP Device Capabilities

71	UC Endpoint
72	612 SIP-DECT

Basic | SDP Options | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press Event

Called Party Inward Dialing Modification | Record Information | Advanced

SIP Device Capabilities Number 71

Comment UC Endpoint

Call Routing and Administration Options

Outbound Proxy Server

Replace System based with Device based In-Call Features Yes

Allow MWI Notifications without Subscription No

Enable Digit Collection In Busy Or Alerting State No

Figure 67: SIP Device Capabilities – Basic

Ring Groups Scheduler

SDS Distribution Errors - All

SDS Distribution Errors - Sys

SDS Distribution Errors - Use

SDS Form Comparison

SDS Form Sharing

Shared System Options

Single Line DNI Sets

Single Line IP Sets

SIP Device Capabilities

SIP Peer Profile

SIP Peer Profile Assignment

SIP Peer Profile Called Party

SIP Peer Profile Calling Party

SMDR Options

SNMP Configuration

SIP Device Capabilities

Basic | **SDP Options** | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press E

Called Party Inward Dialing Modification | Record Information | Advanced

Allow Device To Use Multiple Active M-Lines Yes

Allow Using UPDATE For Early Media Renegotiation Yes

AVP Only Device Yes

Enable Mitel Proprietary SDP No

Force sending SDP in initial Invite message No

Ignore SDP Answers in Provisional Responses No

Limit to one Offer/Answer per INVITE Yes

Prevent SDP Renegotiation If Peer Initiated Hold No

Prevent the Use of IP Address 0.0.0.0 in SDP Messages Yes

Renegotiate SDP To Enforce Symmetric Codec Yes

Repeat SDP Answer If Duplicate Offer Is Received Yes

Send Answer only after renegotiation is complete Yes

Suppress Use of SDP Inactive Media Streams No

Figure 68: SIP Device Capabilities - SDP Options

The screenshot shows the 'SIP Device Capabilities' configuration page. The left sidebar lists various configuration categories, with 'SIP Device Capabilities' selected and highlighted. The main content area is titled 'SIP Device Capabilities' and has several tabs: 'Basic', 'SDP Options', 'Signaling and Header Manipulation' (which is active), 'Distinctive Ring Tones', and 'Timers'. Under the 'Signaling and Header Manipulation' tab, there are sub-tabs: 'Key Press Event', 'Called Party Inward Dialing Modification', 'Record Information', and 'Advanced'. A table of capabilities is displayed, with two rows highlighted by red boxes:

Capability	Value
Allow Display Update	Yes
Disable Reliable Provisional Responses	No
Disable Use of User-Agent and Server Headers	No
Fail REFER To Keep Call Active On Mid-Call Feature	No
If TLS use 'sips:' Scheme	No
Multilingual Name Display	No
Override Auto-Answer Headers	No
Override Auto-Answer Headers With	
Remove Anonymous User	No
Require Reliable Provisional Responses on Outgoing Calls	Yes
Suppress Redirection Headers	No
Use P-Asserted Identity Header	Yes

Figure 69: SIP Device Capabilities - Signaling and Header Manipulation

MiCollab Client Setup on MiCollab

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

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

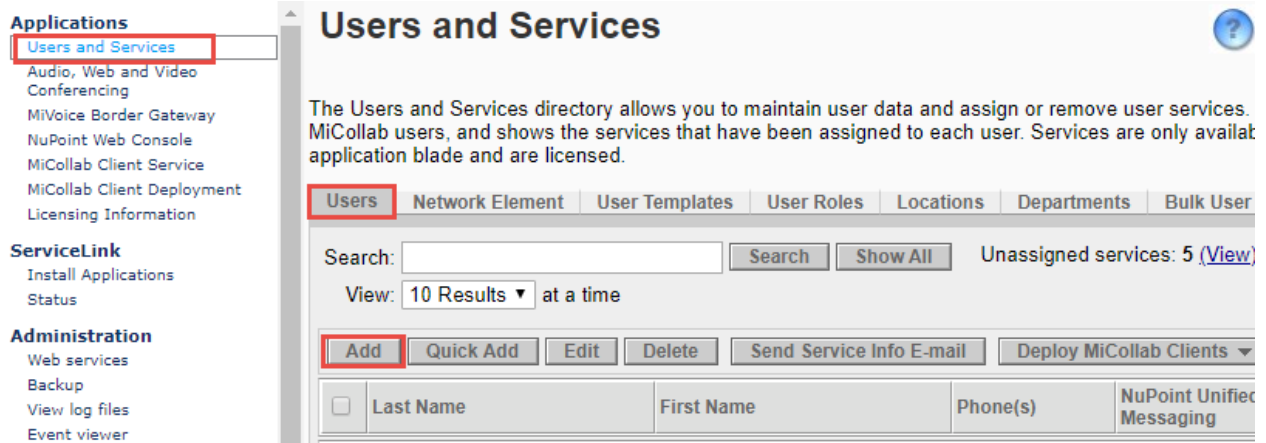


Figure 70: Add MiCollab User Account

Configure the following User fields as follows

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

User Phones NuPoint Unified Messaging MiCollab Client Audio, Web and Video Conferenc

User

First Name: Last Name:

Display Name:

UCC Bundle:

Department:

Location:

Prompt Language:

Primary Email Address:

Distinguished Name:

IDS Manageable

Authentication Section

Login:

Password:

Confirm Password:

TUI Passcode:

Confirm Passcode:

Figure 71: Add MiCollab User Account - Contd.

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

Desk Phone

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

Soft Phone

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

User	Phones	NuPoint Unified Messaging	MiCollab Client	Audio, Web and Video Conferencing	1
------	---------------	---------------------------	-----------------	-----------------------------------	---

1500 (on 3300)

*Number: 1500
 Service Label: DeskPhone

Secondary Element:

DID Service Number:
 Use as Outgoing DID

CESID:

Hot Desking User

Device Type: UC Endpoint
 Deployment Profile: default

Send Deployment Email

Status: Deployed

SIP Device Capabilities:

SIP Password:

Confirm SIP Password:

Figure 72: MiCollab Phone Configuration

1501 (on 3300)

*Number: 1501
Service Label: SoftPhone
Secondary Element:
DID Service Number: Use as Outgoing DID
CESID:
 Hot Desking User
Device Type: UC Endpoint
Deployment Profile: default Status: Deployed
 Send Deployment Email
SIP Device Capabilities: 71
SIP Password:
Confirm SIP Password:
▶ Advanced Phone Settings:
Save Cancel

Figure 73: MiCollab Phone Configuration – Contd.

Advanced Phone Settings for Desk and Soft Phones

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

Advanced Phone Settings:
Service Level: Full
Zone ID: 1
Call Coverage Service Number: 1

	Day	Night 1	Night 2
*Class Of Service:	1	1	1
*Class Of Restriction:	1	1	1

Figure 74: MiCollab Phone Configuration - Contd.

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

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

User Phones NuPoint Unified Messaging **MiCollab Client** Audio, Web and Video Conferencing

MiCollab Client for miclient client1

Feature Profile:

Desk phone extension:

Soft phone extension:

Mailbox number: Number:

Deployment Profile: Status: Deployed

Figure 75: MiCollab Phone Configuration - Contd.

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

Users and Services ?

The Users and Services directory allows you to maintain user data and assign or remove user services. The directory lists the usernames and office numbers of the MiCollab users, and shows the services that have been assigned to each user. Services are only 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 Bulk User Provisioning

Search: Unassigned services: 5 ([View](#)) Total number of users: 2

View: 10 Results at a time

<input type="checkbox"/>	Last Name	First Name	Phone(s)	NuPoint Unified Messaging	MiCollab Client	Audio, Web and Video Conferencing	Teleworker
<input type="checkbox"/>	1000	User			✓		
<input type="checkbox"/>	client1	miclient	1500 1501	✓	✓	✓	✓

Figure 76: MiCollab Phone Configuration – Contd.

MiVoice Border Gateway Configuration Notes

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

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

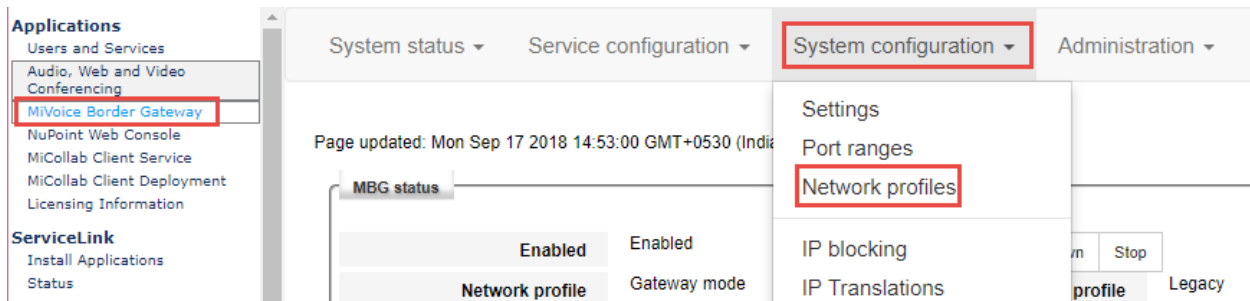


Figure 77: Network Profiles

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

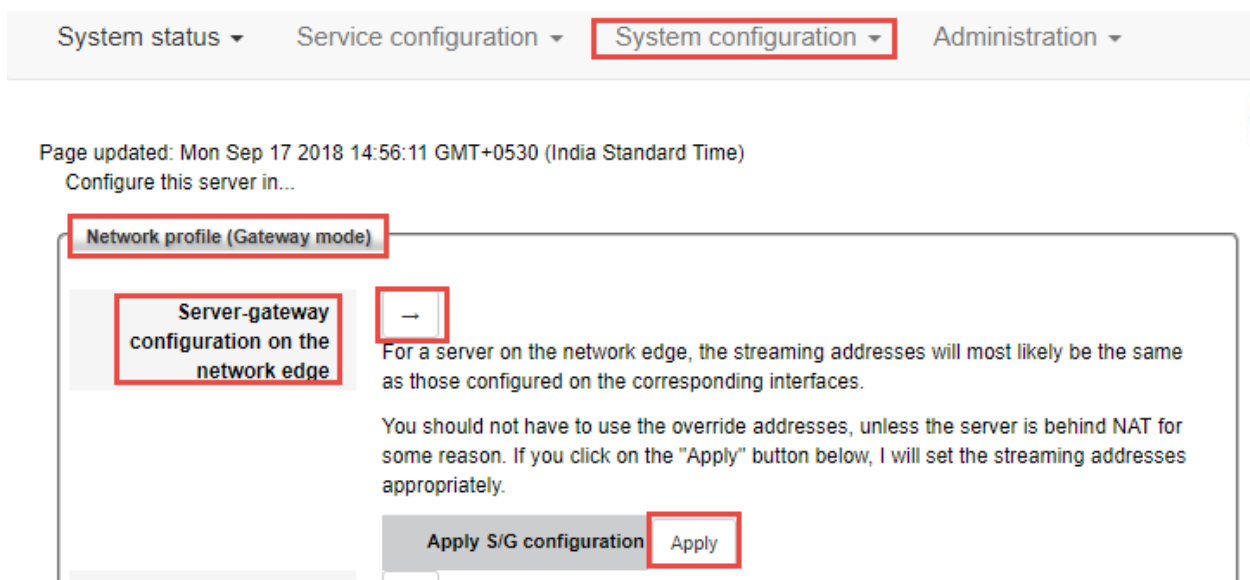


Figure 78: Network Profiles – Contd.

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

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

SIP options

SIP support		Protocols	Access profile
UDP	<input checked="" type="checkbox"/>	Pt	▼
TCP	<input type="checkbox"/>	Pt	▼
TCP/TLS	<input checked="" type="checkbox"/>	Pt	▼

Registration Mode	Pass-Through ▼
Set-side registration expiry time	<input type="text"/>
ICP-side registration expiry time	<input type="text"/>
Allowed URI names	Add another
	<input type="text"/>

Blank any field you no longer want.

Tone injection	Enabled <input type="checkbox"/>
-----------------------	----------------------------------

Device ↔ device local streaming	<input type="checkbox"/>
Device ↔ trunk local streaming	<input type="checkbox"/>
Codec support	Unrestricted ▼
RTP framesize	Dynamic ▼
Set-side RTP security	Allow ▼
ICP-side RTP security	Disable ▼

KPML username	administrator
KPML password	Change KPML password
Confirm KPML password	<input type="text"/>

Permit weak SIP passwords	<input type="checkbox"/>
----------------------------------	--------------------------

PRACK support	<input checked="" type="checkbox"/>
Send options keepalives	Always ▼

Figure 79: MBG Settings

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

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



Figure 80: MiVoice Border Gateway Configuration

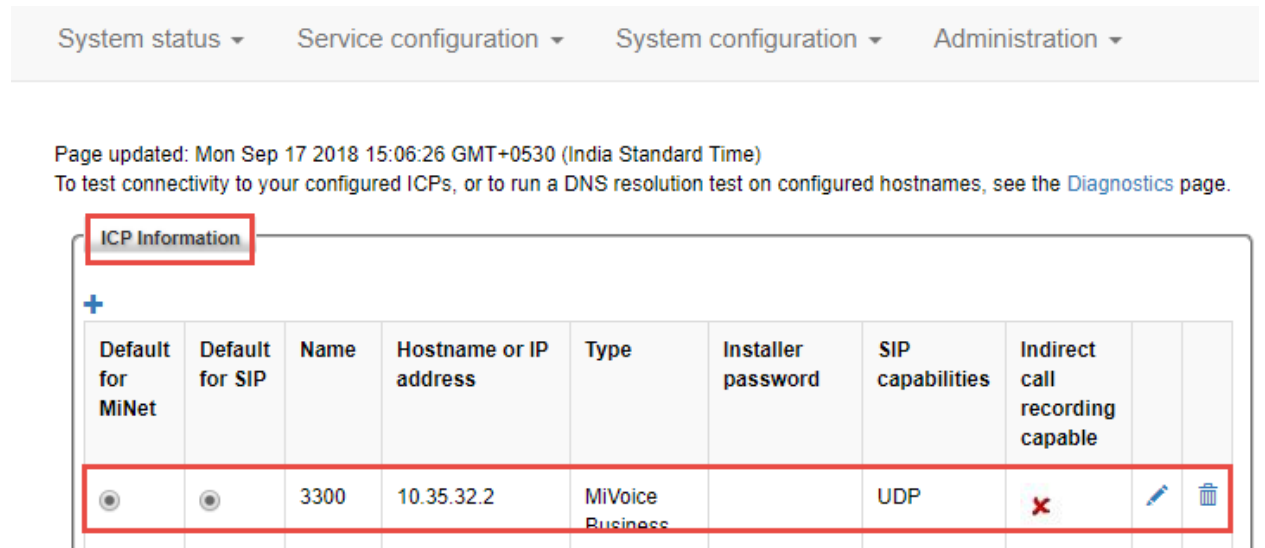


Figure 81: MiVoice Border Gateway Configuration – Contd.

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

Click **Update** Default ICPs

To add a new SIP trunk:

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

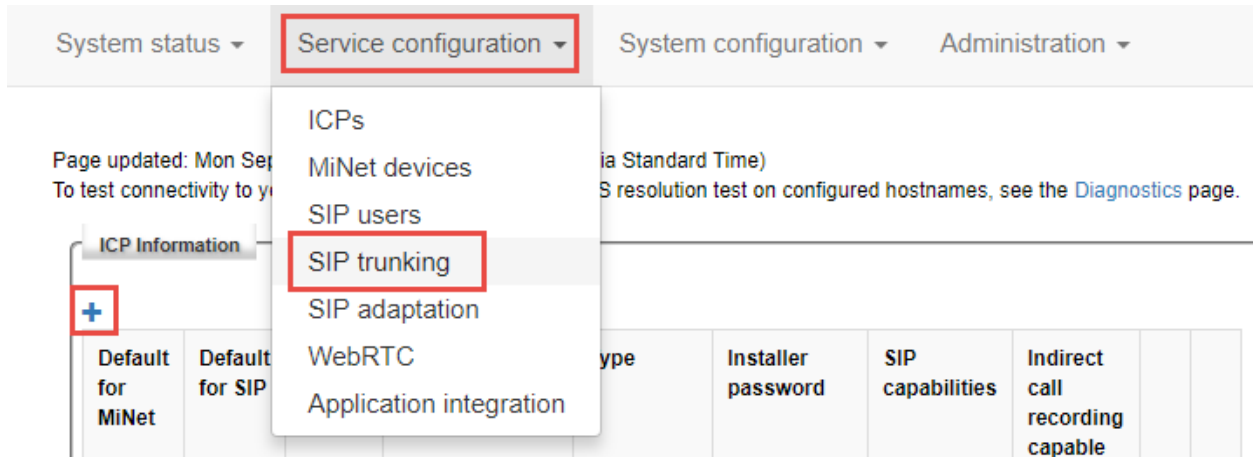


Figure 82: SIP Trunking Configuration

Enter the SIP trunk details as follows:

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

Manage SIP trunk

Enabled

Name

Remote trunk endpoint port

DNS SRV Support

DNS SRV resiliency timeout

Options keepalives

Rewrite host in PAI

Idle timeout (s)

Local streaming between trunk calls

Log verbosity

Authentication password

Set-side RTP security

SIP adaptation receive pipeline

Search routing rules

Remote trunk endpoint address

Accept traffic from any port

DNS SRV query domain

Re-invite conversion

Options interval

Remote RTP framesize (ms)

RTP address override

PRACK support

Authentication username

Confirm authentication password

Icp-side RTP security

SIP adaptation send pipeline

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

Page 1 of 1

Rules per page

Jump to page

Match	Rule	Primary	Secondary	Description	Next	Last
1 <input type="text" value="Request l"/>	<input type="text" value="22471XXXXX"/>	<input type="text" value="3300"/>	<input type="text" value="-----"/>	<input type="text" value="inbound"/>	<input type="button" value="↑"/>	<input type="button" value="↓"/>

Figure 83: SIP Trunk Configuration Settings