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Mitel Technical Configuration Notes – HO858

rev. 2018-12-12

Configure MiVoice Business 9.0 for use with TelNet Worldwide SIP Trunking Using MBG

Description:

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVoice Business to connect to Service Provider TelNet Worldwide SIP Trunking.

Environment:

MiVoice Business 9.0 (9.0.0.184), MiVoice Border Gateway 10.1.0.244, Mitel 69xx Phone 01.04.00.074 and Mitel 68XX Phone 5.1.0.227

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Mitel Technical Configuration Notes – Configure MiVoice Business for use with TelNet Worldwide SIP Trunking using MBG

Nov 2018, HO858

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[Configure MiVoice Business for use with TelNet Worldwide SIP Trunking Using MBG](#)

Table of Contents

Configure MiVoice Business 9.0 for use with TelNet Worldwide SIP Trunking Using MBG	i
Overview	1
Interop History	1
Interop Status.....	1
Software & Hardware Setup	1
Tested Features.....	2
Device Limitations and Known Issues	3
Network Topology.....	4
Configuration Notes.....	5
MiVoice Business Configuration Notes	5
MiVoice Border Gateway Configuration Notes (Optional)	30

Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to Service Provider TelNet Worldwide 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	09-Nov-2018	Initial Interop with Mitel MiVB 9.0 and Service Provider TelNet Worldwide SIP trunk using MBG

Interop Status

The Interop of Service Provider TelNet Worldwide 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 Service Provider TelNet Worldwide SIP Trunking achieved is:

	The most common certification which means Service Provider TelNet Worldwide 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 Service Provider TelNet Worldwide SIP Trunking and the MiVoice Business.

Manufacturer	Variant	Software Version
Mitel	MiVoice Business	Release 9.0 (9.0.0.184)
Mitel	MiVoice Border Gateway	10.1.0.244
Mitel	69XX	01.04.00.074
Mitel	68XX	5.1.0.227
BroadSoft	Broadworks	V20SP1
Oracle SBC	NetNet 4500	7.2.0 MR-6 Patch 9

Tested Features

This is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through Service Provider TelNet Worldwide and their PSTN gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	✓
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	✓
Nu-Point Voicemail	Terminating calls to a Nu-Point voicemail boxes and DTMF detection.	✓
Packetization	Forcing the Mitel MiVB to stream RTP packets through its E2T card at different intervals, from 10ms to 40ms	✓
Personal Ring Groups	Receiving calls through Service Provider TelNet Worldwide 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 Service Provider TelNet Worldwide 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 Service Provider TelNet Worldwide and their PSTN gateway to and from Teleworker extensions.	✓
Video	Making and receiving a call through Service Provider TelNet Worldwide with video capable devices.	✗
Fax	T.38 and G711 Fax Calls	✓
G722/Wideband Codec	Making and receiving a call through Service Provider TelNet Worldwide using G722 codec.	✗
E.164 calling	Make calls using E.164 format	✓

✓ - 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 Service Provider TelNet Worldwide SIP Trunking is connected to the MIVB.

Feature	Problem Description
Packetization	TelNet Worldwide does not support Packetization of 40 MS Recommendation: Contact TelNet Worldwide for more information.
TLS	TelNet Worldwide does not support TLS Recommendation: Contact TelNet Worldwide for more information.
Codec G722/G722.1	TelNet Worldwide does not support wideband G722/G722.1 Codec. Recommendation: Contact TelNet Worldwide for more information.
Video Calls	TelNet Worldwide does not support video calls. Recommendation: Contact TelNet Worldwide for more information.
DTMF (INFO)	DTMF via SIP INFO is not Supported by TelNet Worldwide Recommendation: Contact TelNet Worldwide for more information.
Blind Transfer	A delay of 2s is noticed for an inbound PSTN call transferred back to another PSTN number. Forcing the p-time value to 20 MS on MiVB reduces the delay

Network Topology

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

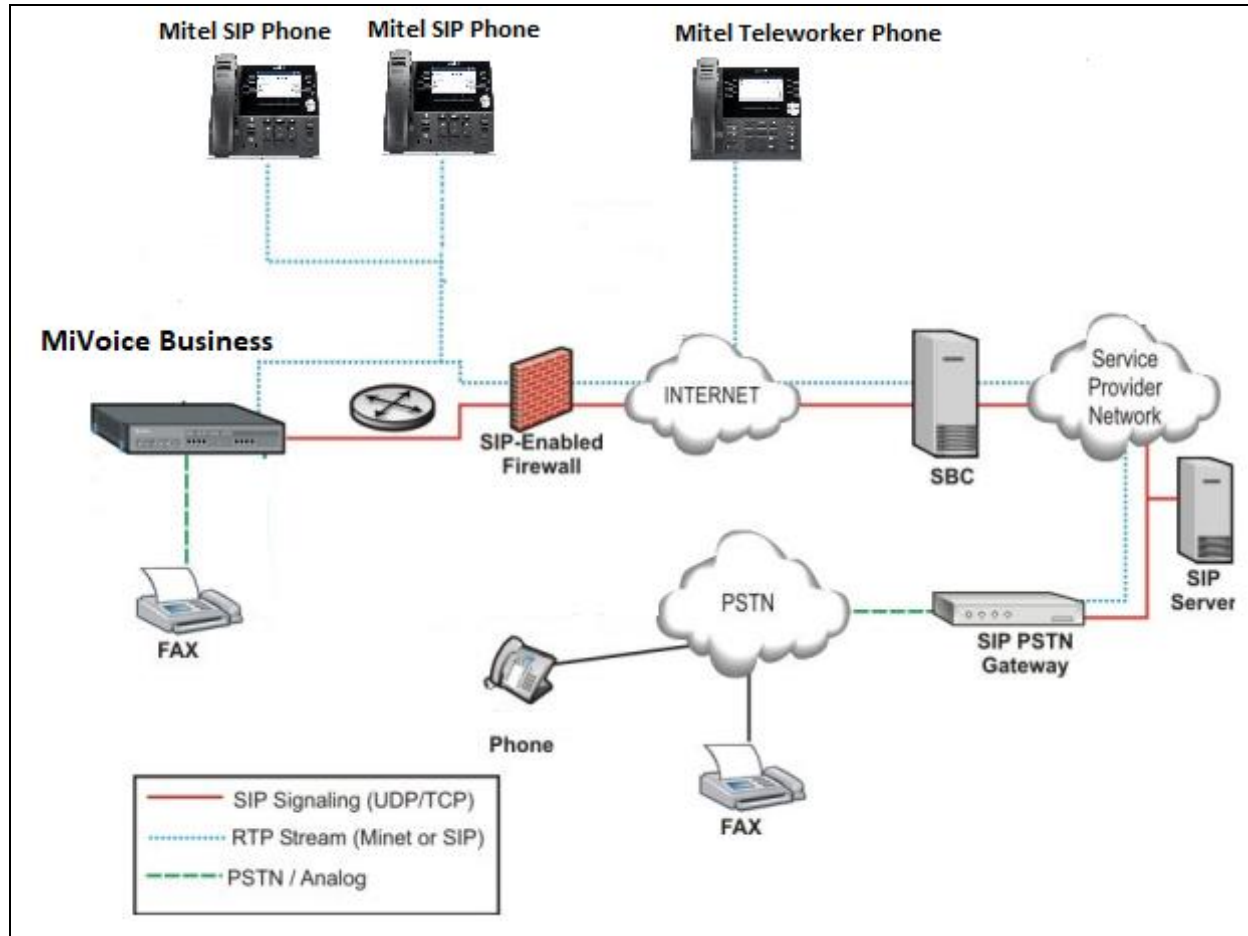


Figure 1 – Network Topology

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 Service Provider TelNet Worldwide SIP Trunking MiVB 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 MiVB to interconnect with Service Provider TelNet Worldwide 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 MIVB 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 MiVB 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 MIVB Programming

The SIP signaling connection uses UDP on Port 5060.

Licensing and Option Selection – SIP Licensing

Ensure that the MiVB is equipped with enough SIP trunking licenses for the connection to Service Provider TelNet Worldwide 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 MiVB to be used with all service providers, applications and SIP trunking devices.

The screenshot shows the Mitel MiVoice Business interface. The left sidebar contains a menu with 'Licenses' highlighted. The main area is titled 'License and Option Selection on Local_65'. It features a table with columns for various services and their configurations. The 'SIP Trunks' row is highlighted with a red box.

License and Option Selection							
Trunking / Networking							
Digital Links	0	1	1	0	Unrestricted	Yes	
Compression		40	0	40	Unrestricted	Yes	
FAX Over IP (T.38)		20	0	20	Unrestricted	Yes	
SIP Trunks	0	15	0	15	Unrestricted	Yes	
Others							
IDS Connection	0	No	1	0	Unrestricted	Yes	
MLPP	0	No	0	0	Unrestricted	No	
Configuration Options							

Figure 2 – License and Option Selection

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 MiVB.

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

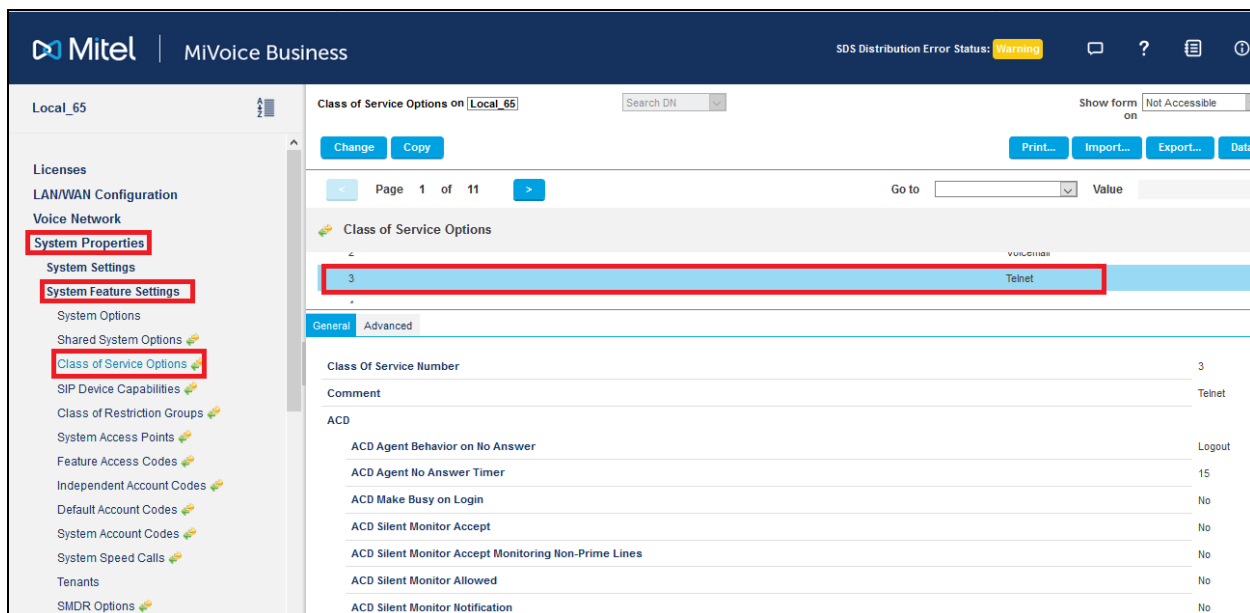


Figure 3 – Class of Service

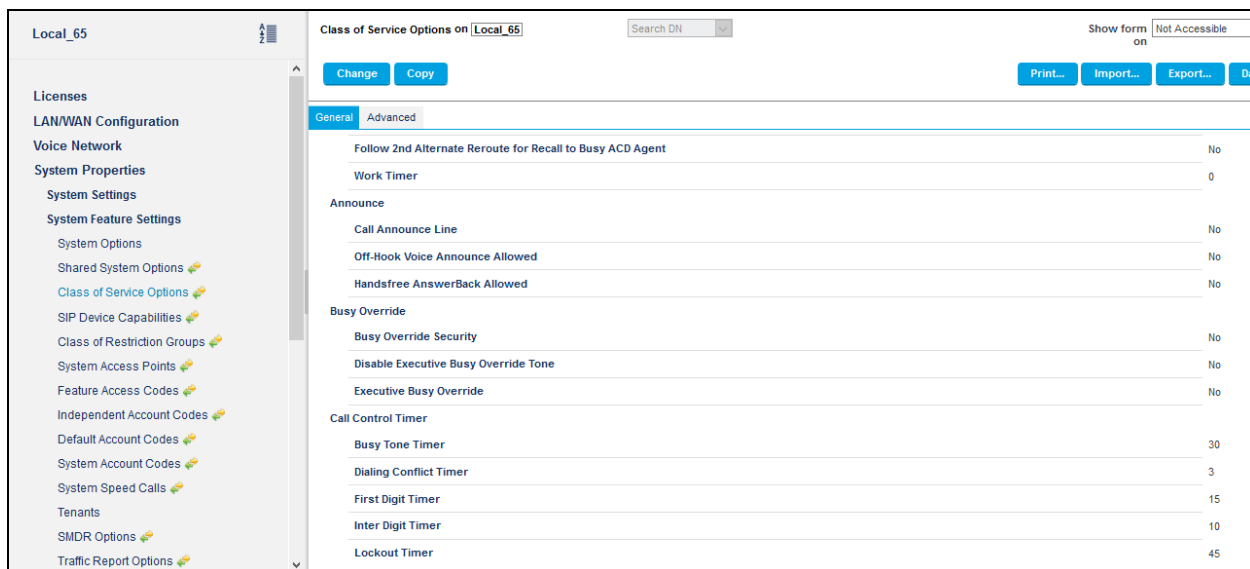


Figure 4 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

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

General Advanced

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

Figure 5 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

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

General Advanced

Group Call Forward Follow Me Accept

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	30
Local Music On Hold source	No
Music on Hold on Transfer	No
Use Called Party Call Hold Timer	No

Call Park

Call Park Timer	180
Call Park-Allowed To Park	No

Figure 6 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

Change Copy

Print... Import... Export... D

General Advanced

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	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	No

Call Waiting

Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes

Campon

Auto Campon Timer

Figure 7 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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

Figure 8 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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
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	Yes
Fax Capable	Yes
Return Disconnect Tone When Far End Party Clears	Yes

Figure 9 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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	No
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

Figure 10 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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
Enable Device Configuration	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

Figure 11 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
Ringing	
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	180

Figure 12 – Class of Service General

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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 DPHSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
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	Yes
ONS VMail-Delay Dial Tone Timer	5

Figure 13 – Class of Service General

Class of Service Advance Tab Configuration Value should be Default. As shown below

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

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
Disable Conference Join Tone	No
DND	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No

Figure 14 – Class of Service Advance

Configure MiVoice Business for use with TelNet Worldwide SIP Trunking 12
Using MBG

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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	

Figure 15 – Class of Service Advance

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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
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

Figure 16 – Class of Service Advance

Local_65

Class of Service Options on Local_65

Search DN

Show form on Not Accessible

General Advanced

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 17 – Class of Service Advance

Network Element Assignment

Create a network element for Service Provider TelNet Worldwide SIP Trunking. In this example, the soft switch is reachable by an IP Address and is defined as “Service Provider TelNet Worldwide” in the network element assignment form. **The FQDN or IP addresses of the SIP Peer (Network Element), the External SIP Proxy 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.

Change

 Network Elements

Name	<input type="text" value="Telnet"/>
Type	<div>Other</div>
FQDN or IP Address	<div>209.142.200.14</div>
Local	False
Version	
Zone	<div>1</div>
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	<div>UDP</div>

SIP Peer Specific	
SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	209.142.200.14
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 18 – 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, the MiVB 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).

Change


Network Elements

Name

MBG_100

Type

Outbound Proxy

FQDN or IP Address

192.168.10.100

Local

False

Version

Zone

1

ARID

Outbound Proxy Specific

Outbound Proxy Transport Type

UDP

Outbound Proxy Port

5060

Save

Cancel

Figure 19 – 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 2 which will be used to direct incoming calls to an answer point in the Mitel MiVB. 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 MiVB will absorb the first 6 digits of the DID number from Service Provider TelNet Worldwide leaving 4 digits for the MiVB to translate and ring the remaining 4-digit extension. For example, Service Provider TelNet Worldwide delivers 248-498-1136 through the SIP trunk to the MiVB. The MiVB will absorb the first 6 digits (248498) leaving the MiVB to ring extension 5000. Extension 5000 must be programmed as a valid dialable number in the MiVB. Please refer to the Mitel MiVB System Administration documentation for further programming information.

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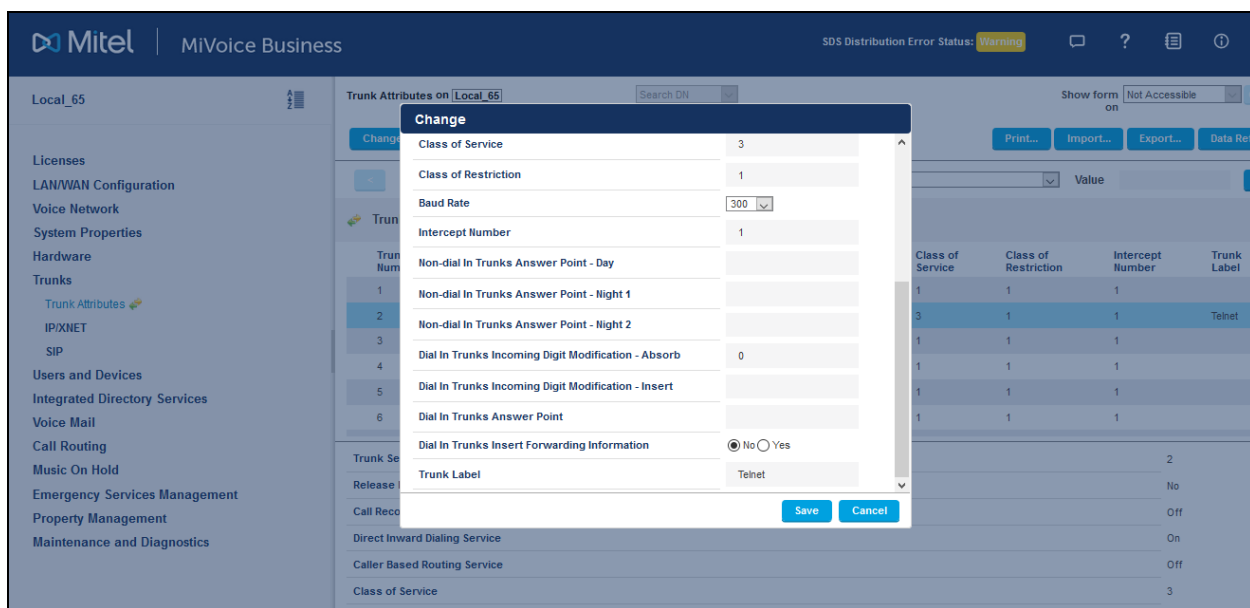
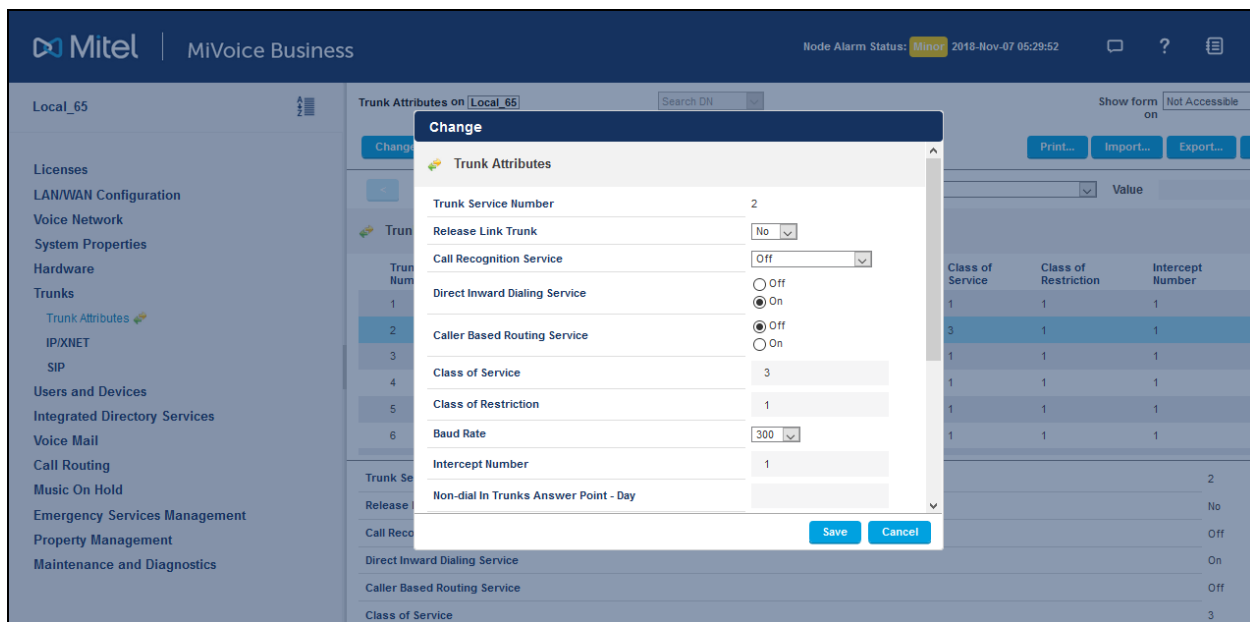


Figure 20 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base MiVB 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 "Service Provider TelNet Worldwide" Network Element.

Registration User Name: The Mitel MiVB does not support Bulk Registration; therefore, trunks will have to be registered individually. Enter the DIDs assigned by Service Provider TelNet Worldwide. 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, "6135554000-6135554400, 6135554500"). 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 Service Provider TelNet Worldwide.** 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).

Maximum Simultaneous Calls: This entry should be configured to maximum number of SIP trunks provided by Service Provider TelNet Worldwide.

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

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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		Telnet
Network Element		Telnet
Local Account Information		
Registration User Name		
Address Type		IP Address: 192.168.10.85
Administration Options		
Interconnect Restriction		1
Maximum Simultaneous Calls		5
Minimum Reserved Call Licenses		0
Outbound Proxy Server		MBG_100
SMDR Tag		0
Trunk Service		2
Zone		1

Figure 21 – SIP Peer Profile Assignment- Basic

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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-Caller-Party-ID (if present)		Yes
Route Call Using To Header		No

Figure 22 – SIP Peer Profile Assignment- Call Routing

Configure MiVoice Business for use with TelNet Worldwide SIP Trunking ²⁰
Using MBG

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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 23 – SIP Peer Profile Assignment- Calling Line ID

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

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
IP Media Default	ipv4
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

Figure 24 – SIP Peer Profile Assignment- SDP Options

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1
<div> Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event Outgoing DID Ranges Profile Information </div>						
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
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

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1
<div> Basic Call Routing Calling Line ID SDP Options Signaling and Header Manipulation Timers Key Press Event Outgoing DID Ranges Profile Information </div>						
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-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 25 – SIP Peer Profile Assignment- Signaling and Header Manipulation

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1
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					0	
Session Timer: Local as Refresher					No	
Subscription Period					3600	
Subscription Period Minimum					300	
Subscription Period Refresh (%)					80	
Invite Ringing Response Timer					0	

Figure 26 – SIP Peer Profile Assignment- Timers

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1
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					No	
Force Out Subscriptions for Remote Digit Monitoring					No	
Request Outbound Proxy to Handle Out Subscriptions					No	
KPML Transport					default	
KPML Port					0	

Figure 27 – SIP Peer Profile Assignment- Key Press Event

SIP Peer Profile

Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Telnet	Telnet	MBG_100	No	2	0	1

Basic

Call Routing

Calling Line ID

SDP Options

Signaling and Header Manipulation

Timers

Key Press Event

Outgoing DID Ranges

Profile Information

<

Figure 28 – SIP Peer Profile Assignment- Outgoing DID Ranges

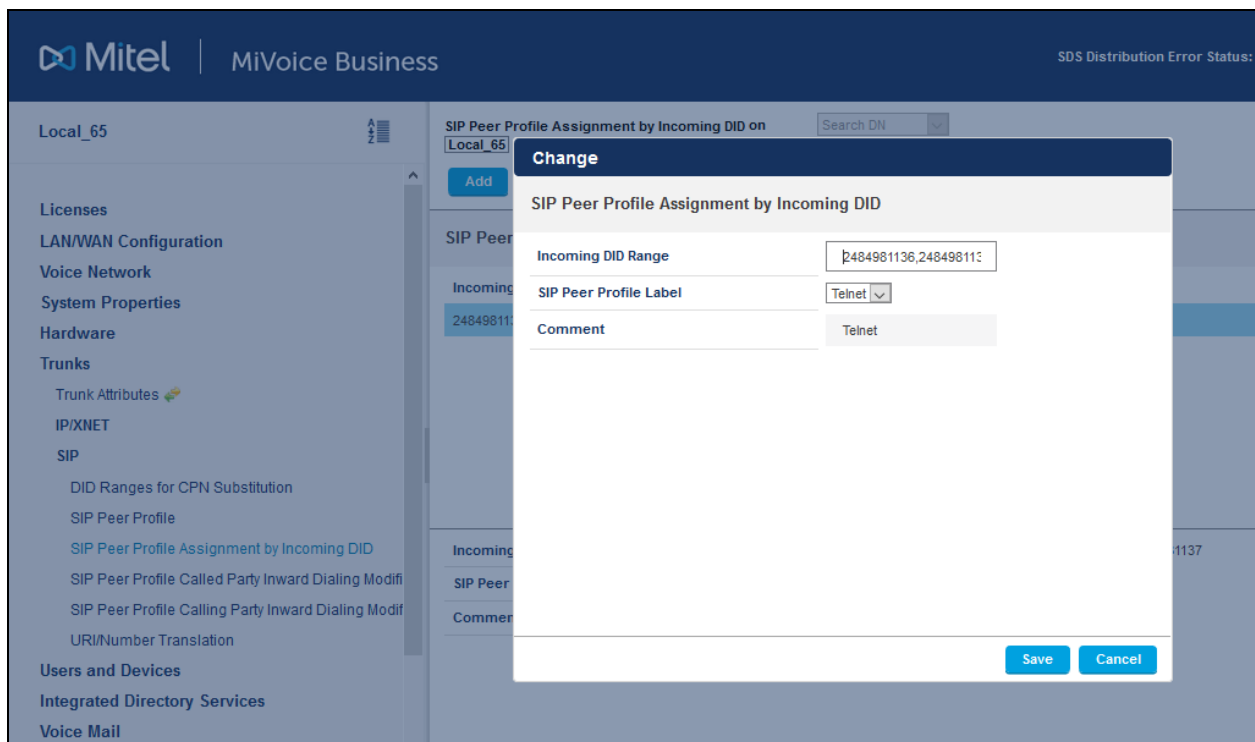


Figure 30 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to Service Provider TelNet Worldwide absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 3 digits (in this case will be 456 to dial out).

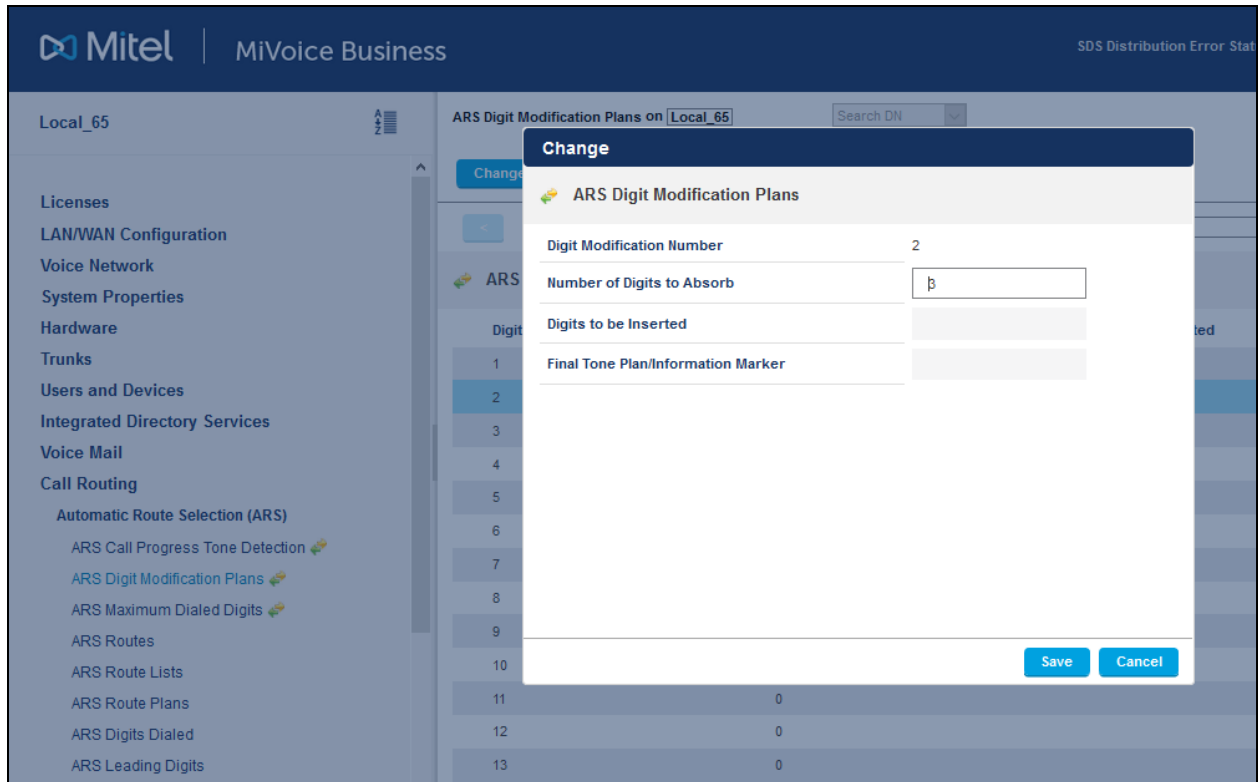


Figure 31 – Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to Service Provider TelNet Worldwide. In this example, the SIP trunk is assigned to Route Number 2. Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

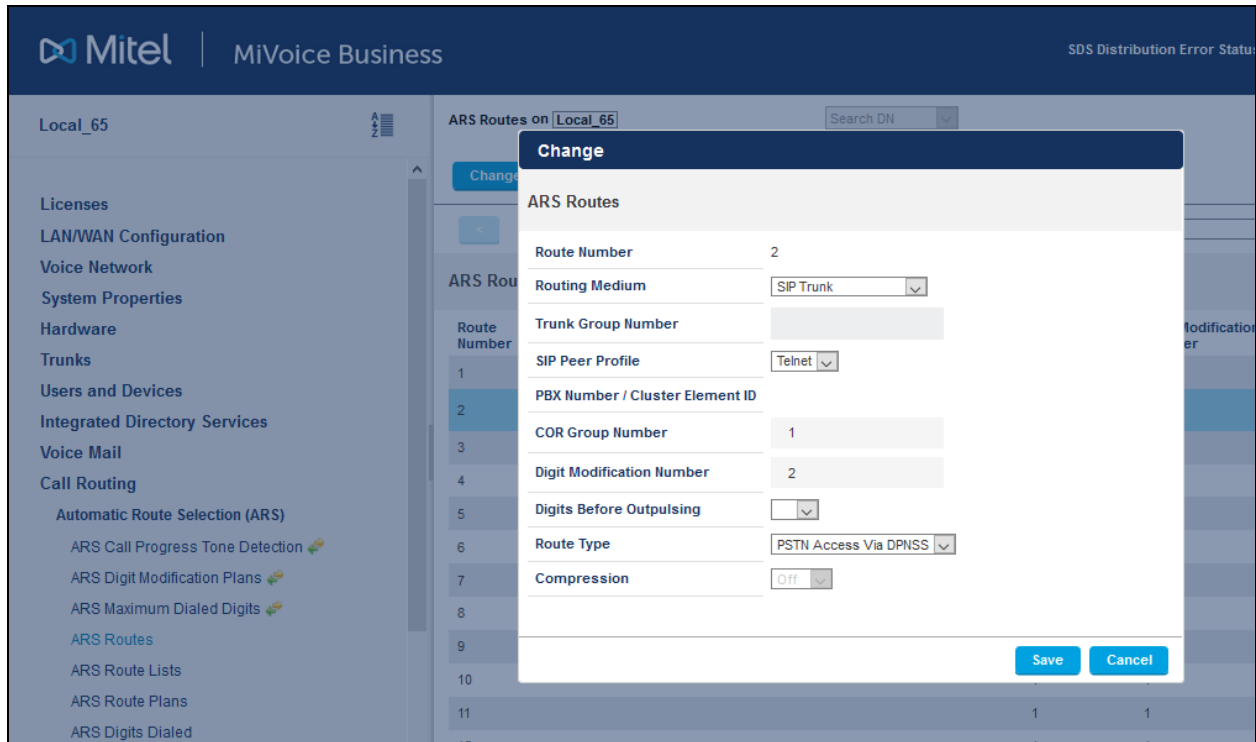


Figure 32 – SIP Trunk Route Assignment

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 456, the call will be routed to Service Provider TelNet Worldwide (i.e. Route 2).

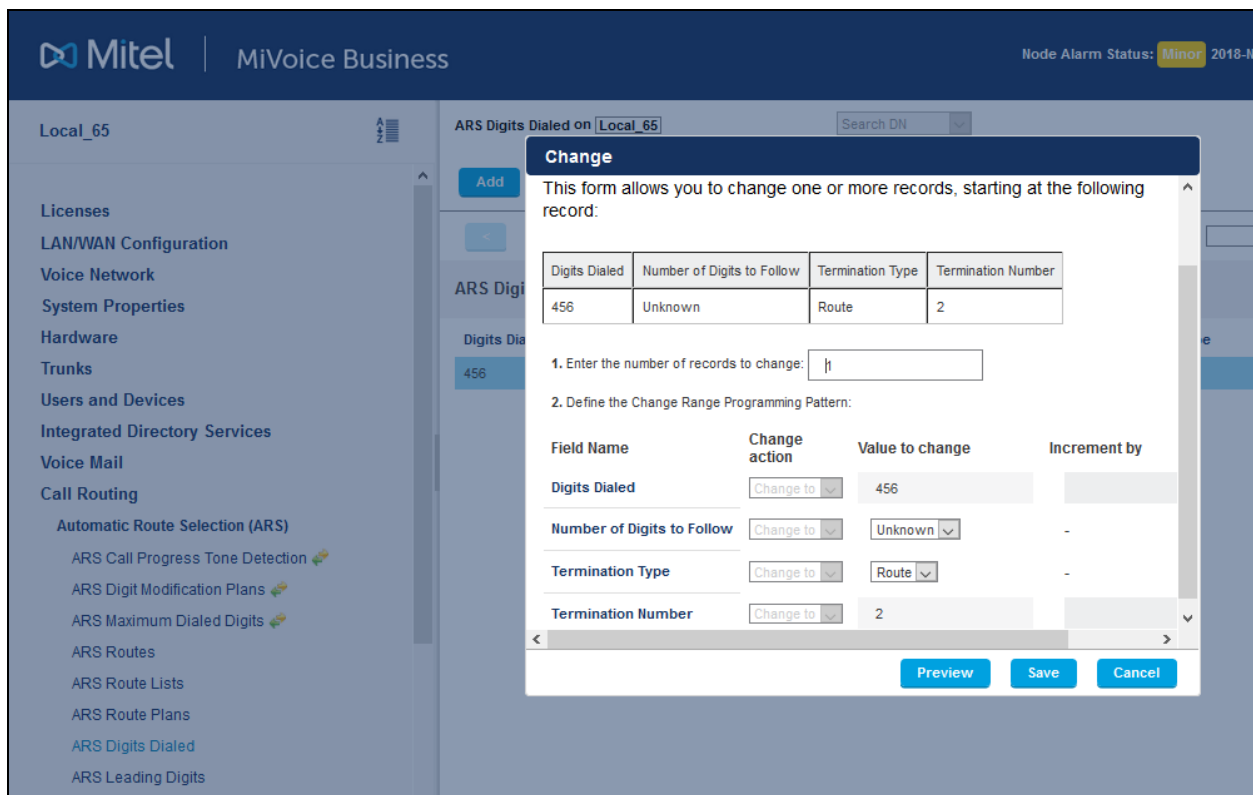


Figure 33 – ARS Digit Dialed Assignment

T.38 Fax Configuration

Service Provider TelNet Worldwide 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).

MiVoice Business

SDS Distribution Error Status: Warning

MN89

Fax Service Profiles on MN89

DN to search

Show form on MN89 (Login Node)

Go

Change

Print...

Import...

Export...

Data Refresh

Inter-Zone Fax Profile

Maximum Fax Rate

High Speed Redundancy

Low Speed Redundancy

Error Correction Mode (ECM)

Override Non-Standard Facilities (NSF)

Label

14400 (V.17, 14400bps)

1

3

Disabled

Disabled

Inter-zone

<

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	-	-	-	-	-	-	-	
5	-	-	-	-	-	-	-	

Licenses

LAN/WAN Configuration

Voice Network

Network Elements

Cluster Elements

Admin Groups

Fax Service Profiles

Fax Advanced Settings

Network Zones

Network Zone Topology

Bandwidth Management

Codec Settings

System Properties

Hardware

Trunks

Users and Devices

Integrated Directory Services

Voice Mail

Call Routing

Music On Hold

Emergency Services Management

Figure 34 - 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”. Service Provider TelNet Worldwide uses the Intra-zone FAX Profile 2 for T.38 FAX.

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	Audio Source	Embedded Music Source	Music-On-Hold Music Source
1	No		1										
2	No		2	T38 Fax Zone									
3	No		1										
4	No		1										

Figure 35 – Zone Assignment

MiVoice Border Gateway Configuration Notes (Optional)

When configuring MiVoice Border Gateway (MBG), you need to identify the working MiVB ICP where to forward SIP messages to and then to configure the SIP trunk.

To do this:

- Login to MBG and click **MiVoice Border Gateway**
- In right pane, click **Service Configuration** tab and then **ICPs** (see Figure 36 for details)

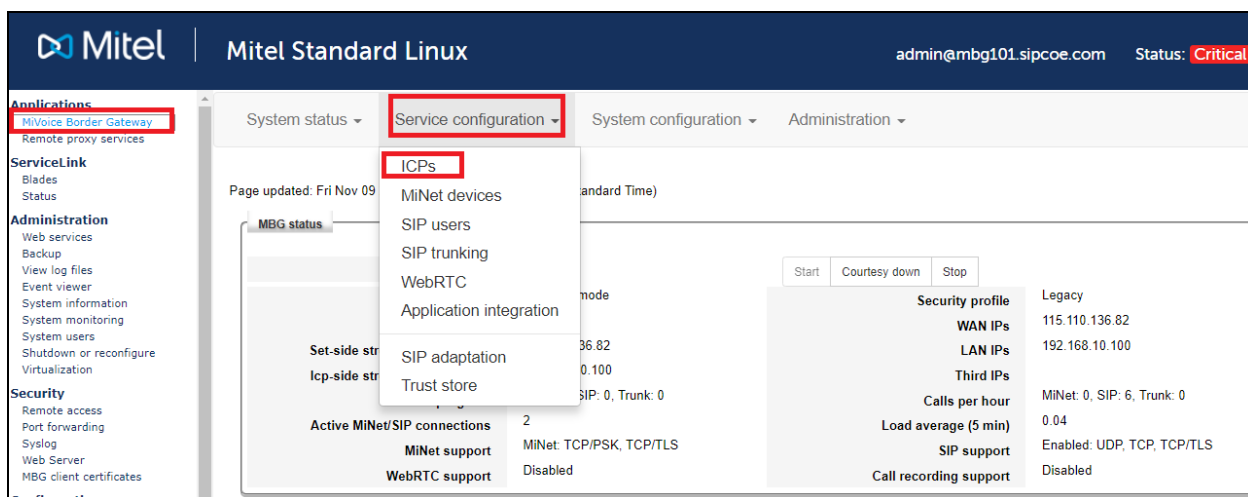


Figure 36 – MBG's Configuration page

- On **ICPs** page, ensure that the “working” MiVB is configured. If needed, click **Add ICP** link and add a new Mitel switch.
- Click **Update** button

Configure MiVoice Business for use with TelNet Worldwide SIP Trunking ³⁰ Using MBG

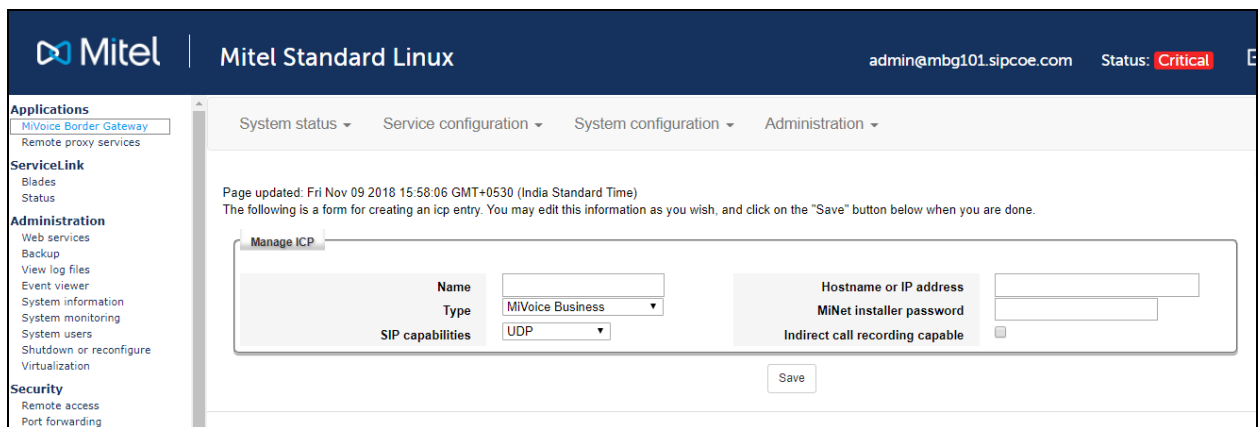


Figure 37 – ICP configuration page

To add a new SIP trunk:

- Click **Service Configuration** tab and then click **SIP trunking** as shown below
- Click **Add a SIP trunk** link (see Figure 38)

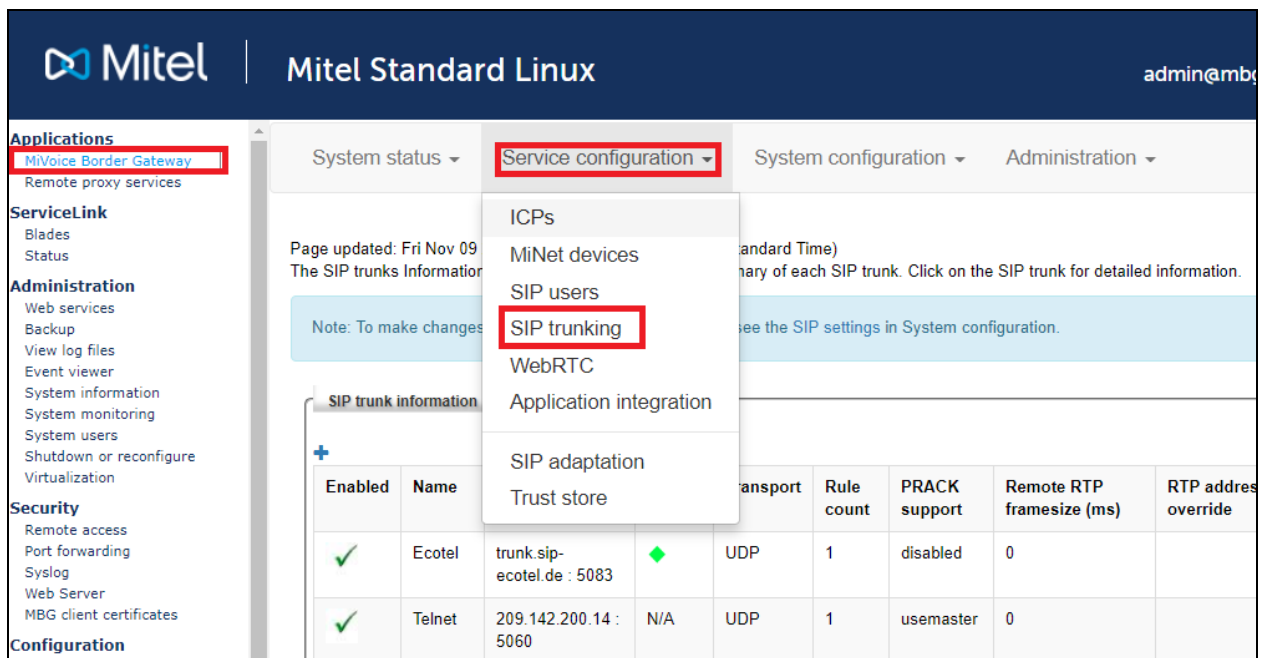


Figure 38 – SIP Trunking configuration page

Enter the SIP trunk's details as shown in Figure 39:

Name – is the name of the trunk

Remote trunk endpoint address – the public IP address of the provider's switch or gateway (this address should be given to you by the provider, e.g. Service Provider TelNet Worldwide).

Configure MiVoice Business for use with TelNet Worldwide SIP Trunking 31 Using MBG

Local/Remote RTP frame size (ms) – is the packetization rate you want to set on this trunk

PRACK – Use master setting.

Routing rule one – it allows routing of any digits to the selected MiVB

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

Click **Save** button

Manage SIP trunk

Enabled ☒

Name

Remote trunk endpoint port

Transport protocol

Accept traffic from all UDP ports ☒

Options keepalives

Rewrite host in PAI ☒

Idle timeout (s)

Local streaming between trunk calls ☐

Log verbosity

Authentication password

Trunk-side RTP security

Inbound

Outbound

Preferred cipher

SIP adaptation receive pipeline

Remote trunk endpoint address

DNS SRV query domain

DNS SRV resiliency timeout

Re-invoke conversion ☐

Options interval

Remote RTP framesize (ms)

RTP address override

PRACK support

Authentication username

Confirm authentication password

ICP-side RTP security

Inbound

Outbound

Preferred cipher

SIP adaptation send pipeline

Search routing rules

Next Previous

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 10

Jump to page 1

First Prev

Match Rule Primary Secondary Description

1 Request URI * MIVB_65 MIVB_65

Next Last

Figure 39 – SIP Trunk configuration settings

Check status: click **System status** tab and then click **SIP Trunks**

Telnet

Status ☒

Reason 0 / 3

Calls in progress / Max 0 / 3

Calls per hour / Max 0 / 603

Reset metrics

Figure 40 – SIP Trunk Status