Mitel Technical Configuration Notes – HO4773

November 3, 2022

MiVB - Configure MiVoice Business 9.4 for use with POLYAI using MBG

Description: This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB and MBG to connect to POLYAI system.

Environment: MiVoice Business 9.4 (9.4.0.25), MiVoice Border Gateway 11.4.0.227, Mitel 69xx MiNET 01.08.00.015, 53XX MiNET 06.05.01.06 and MiCollab 9.6.0.13-01

Version: 3

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Mitel Technical Configuration Notes – Configure MiVoice Business for use with POLYAI using MBG.

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Overview

This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to POLYAI. 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	July, 2022	Interop with Mitel MiVB 9.4 and POLYAI using MBG.

Interop Status

The Interop of POLYAI has been given a Certification status. This Trunking device will be included in the Mitel Interoperability Reference Guide (IRG). The status of POLYAI achieved is:

COMPATIBLE	The most common certification which means POLYAI has been tested and/or validated by the Mitel Third-Party Interop Team. Mitel Product Support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

This was the test setup to generate a basic SIP call between POLYAI and the MiVB using MBG.
Note – Although this testing was performed on the below tested variants, the scope of this testing can be extended to other product variants that work with the same firmware. The list of components for which this testing can be considered applicable is given in the "Additional Applicable Variants" column of the following table –

Manufacturer	Tested Variants	Software Version	Additional Applicable Variants
Mitel	MiVoice Business	Release 9.4 (9.4.0.25)	Legacy Flex, MiCloud Flex GCP
Mitel	MiVoice Border Gateway	11.4.0.227	NA
Mitel	MiCollab Server	9.6.0.13-01	NA
Mitel	69XX MiNET	01.08.00.015	NA
Mitel	53xx MiNET	06.05.01.06	NA

Tested Features

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

Feature	Feature Description	Issues
Basic Call	Making calls from internal/external numbers and POLYAI hangs up the call through IVR.	☑
DTMF	Making calls from internal/external numbers and verified DTMF scenario through IVR (RFC2833).	☑
Hold/Retrieve	Holding/Retrieving the current IVR call.	☑
Call Transfer	Making calls from internal/external numbers and POLYAI transferring the call back to Mitel through IVR internal number/external PSTN number/teleworker user/MiCollab client.	☑
Codec	All the scenarios has been performed over G.711 codec.	☑
TLS/SRTP	Making calls from internal/external numbers and POLYAI hangs up, transfer and DTMF calls through IVR.	☑
Resiliency	Testing resiliency feature between MiVB, MBG and the service provider POLYAI.	☑

☑ - No issues found ✗ - Issues found, cannot recommend using ⚠ - Issues found

Device Limitations and Known Issues

This is a list of problems or unsupported features when POLYAI is connected to the MiVB and MBG.

Feature	Problem Description

Network Topology

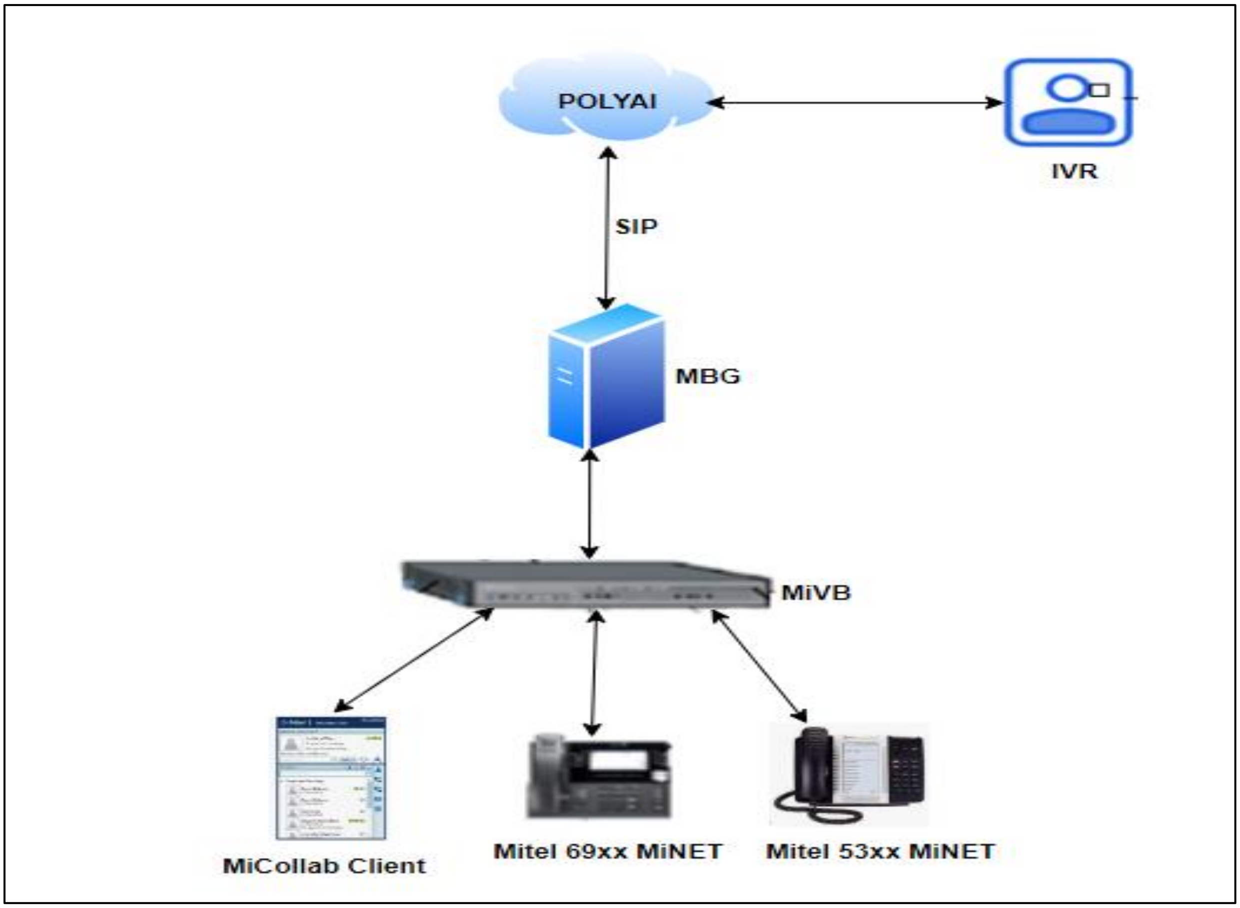


Figure 1 – Network Topology

Call Flow

When Mitel calling ARS (111) followed by any number let's say 1111234 and the call gets through POLYAI and the POLYAI transfer the call back to Mitel internal extension/external PSTN numbers through IVR.

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline on how a device can be configured in a customer environment and how POLYAI MiVB programming was configured in our test environment.

Disclaimer: Although Mitel has attempted to set up 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.

MiVB Configuration Notes

The following steps show how to program a MiVB to interconnect with POLYAI system.

Configuration Template

A configuration template can be found in the same Mitel Knowledge Management System (KMS) 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 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 POLYAI. 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 the applications, and SIP trunking devices.

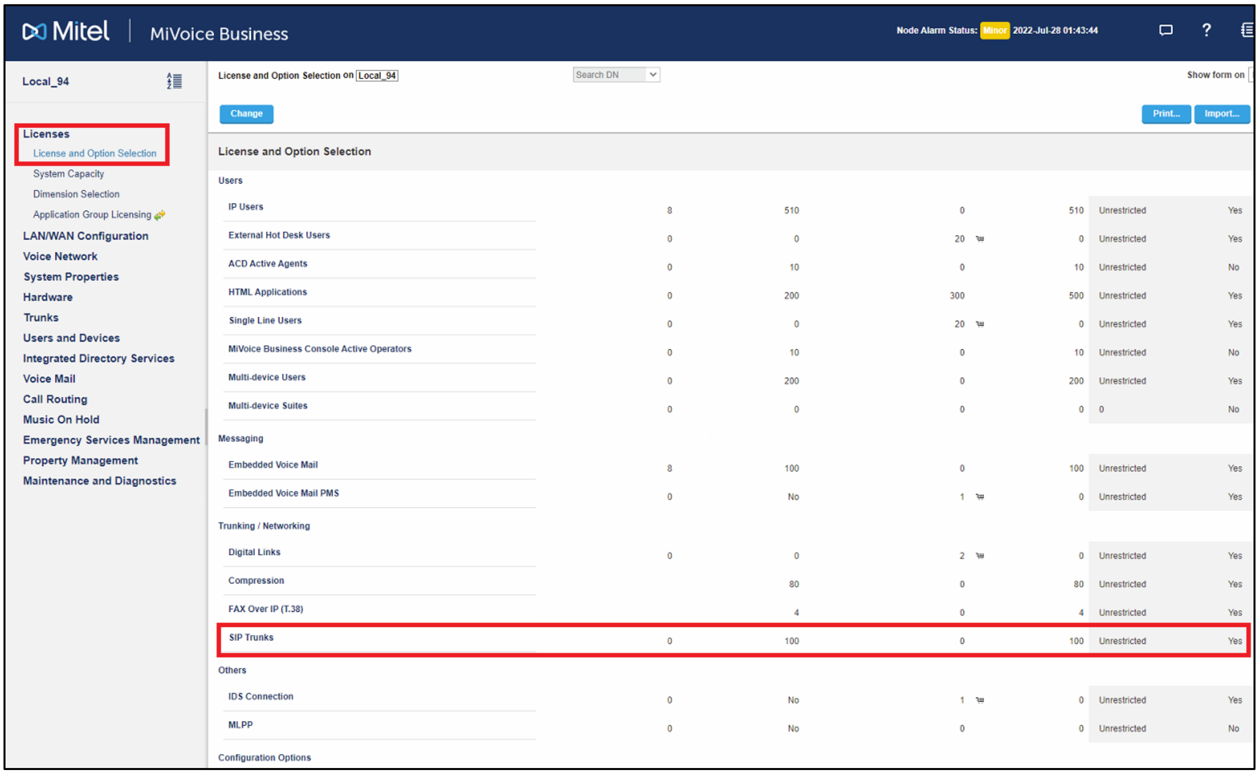


Figure 2 – License and Option Selection

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

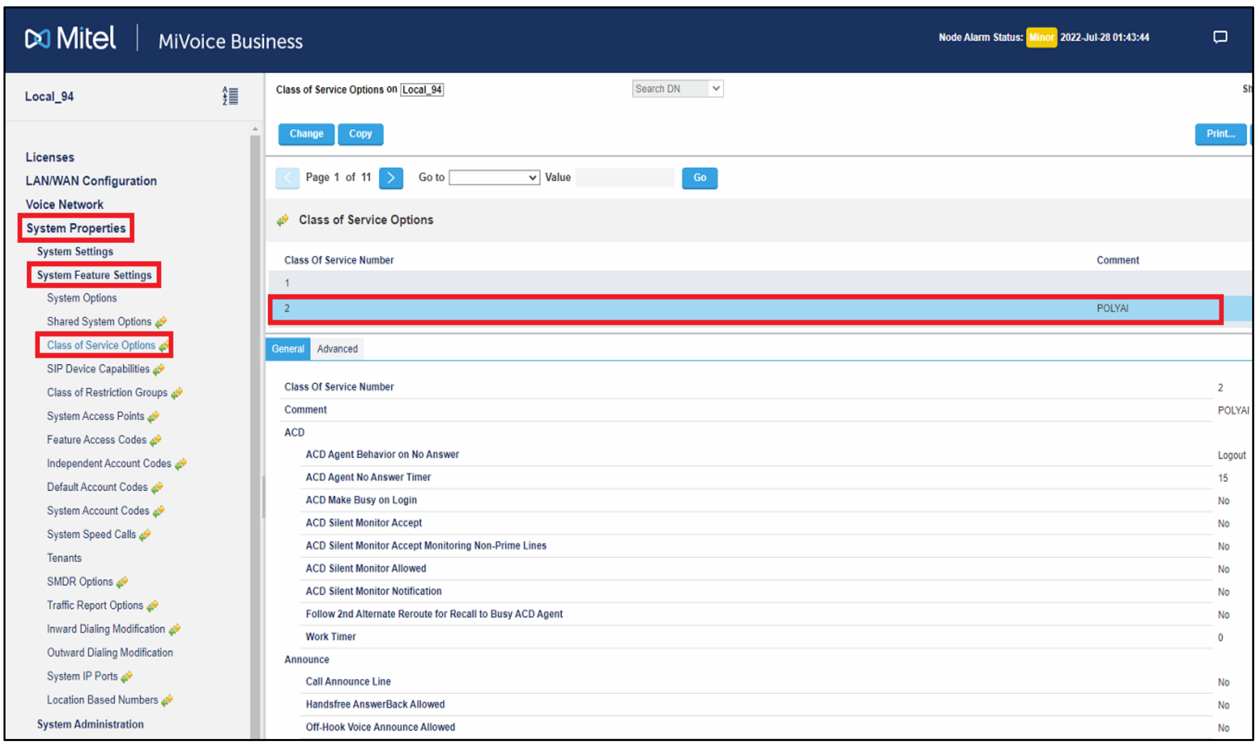


Figure 3 – Class of Service

Network Element Assignment

Create a network element for POLYAI. In this example, the soft switch is reachable by an IP Address and is defined as “POLYAI “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 Provider.**

If your POLYAI 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 Provider. Set the transport to UDP and port to 5060.

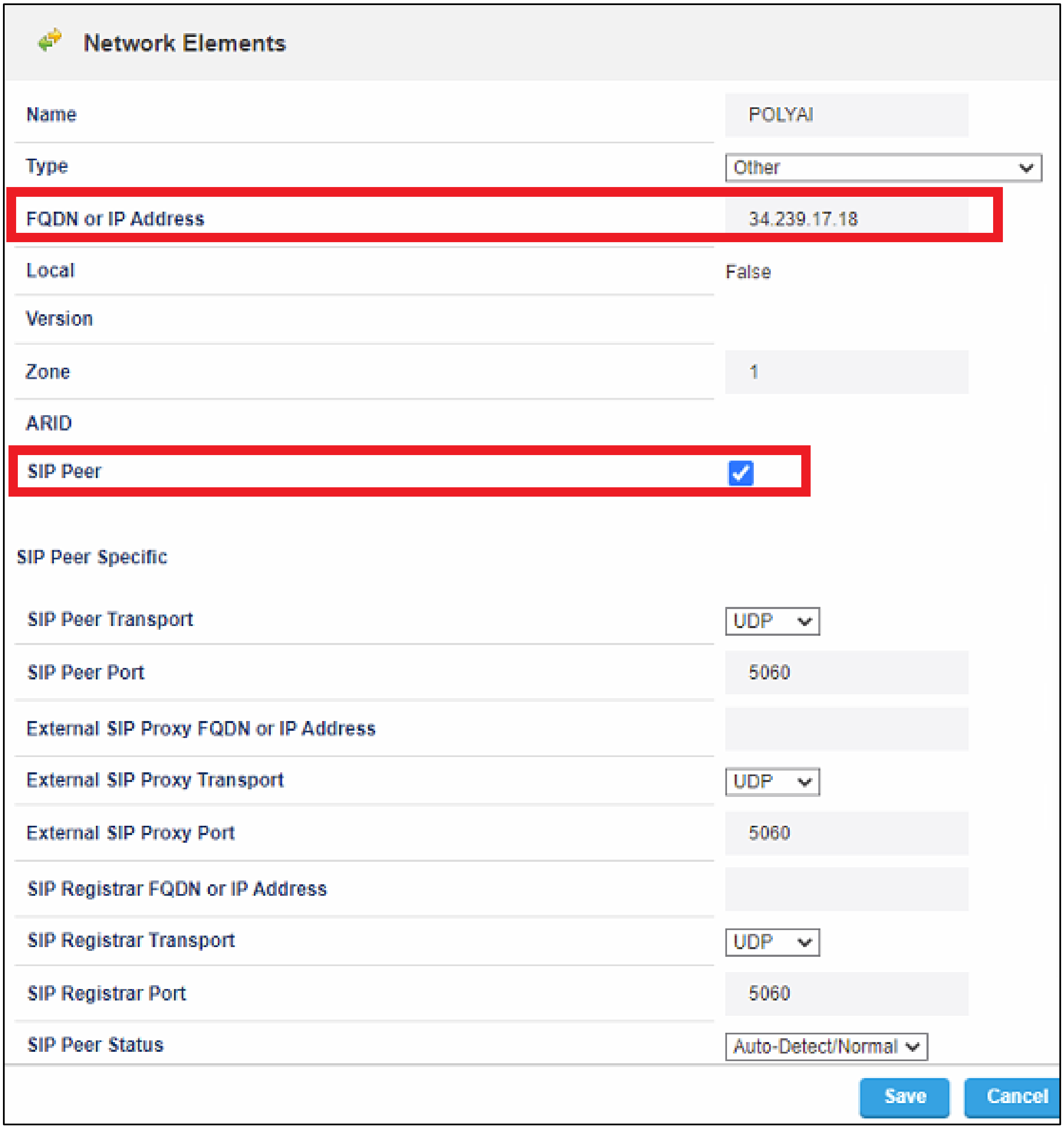


Figure 4 – Network Element Assignment

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 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).

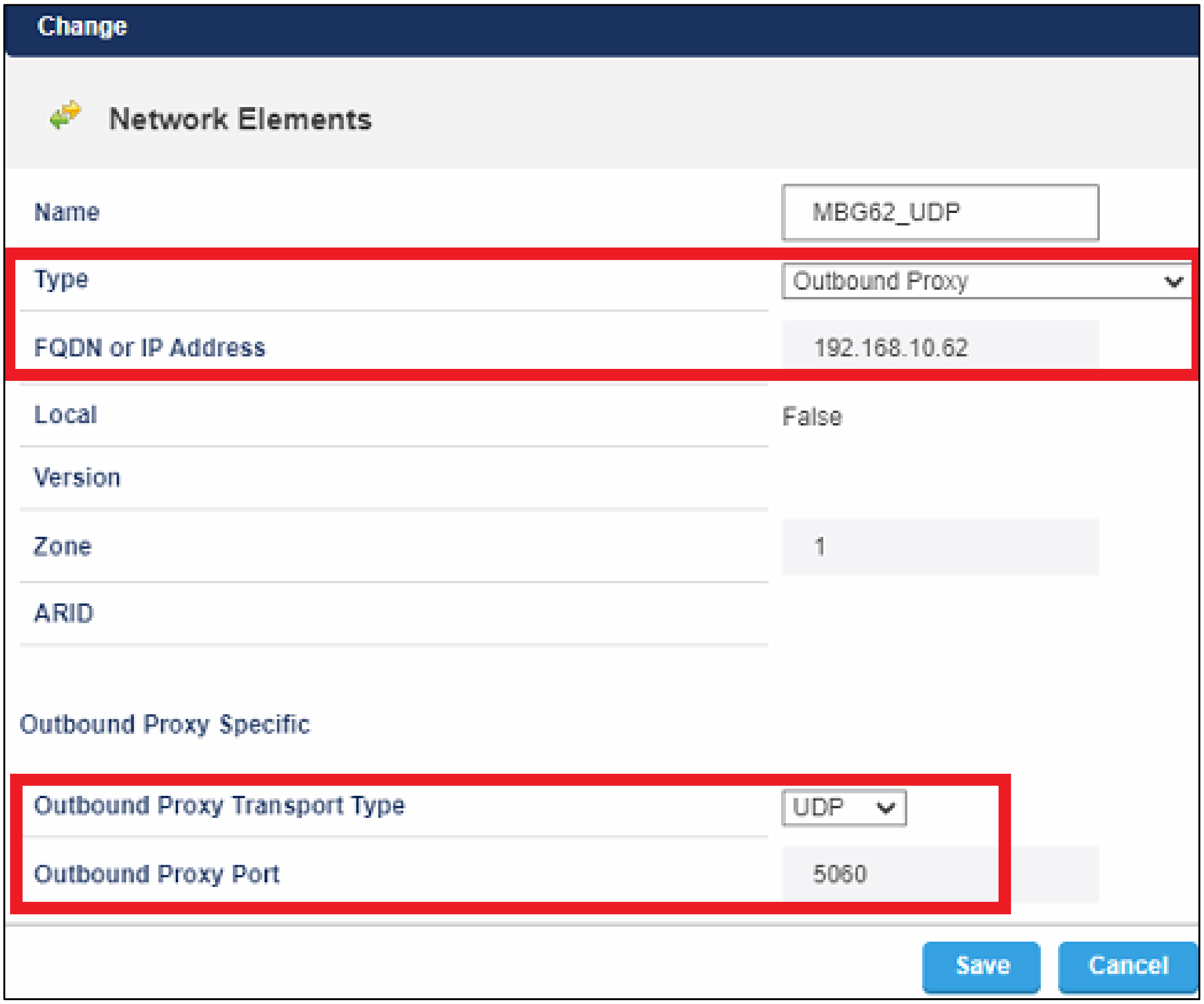


Figure 5 – Network Element Assignment (Proxy)

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 Provider.

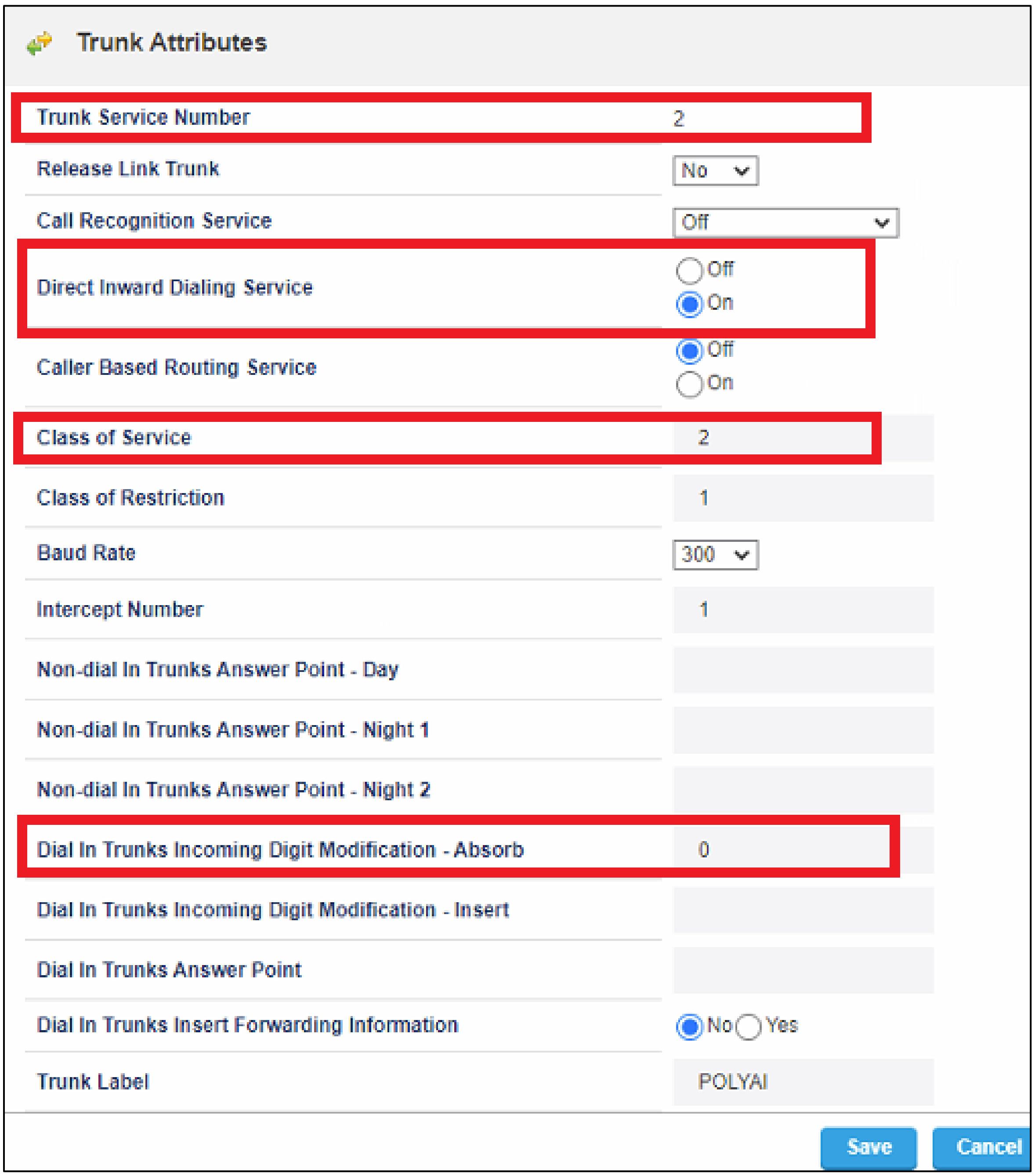


Figure 6 – Trunk Attributes

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 "POLYAI" Network Element.

Registration Username: The Mitel MiVB does not support Bulk Registration; therefore, trunks will have to be registered individually. Enter the Value assigned by POLYAI. Enter one or more numbers. The field has a maximum of 60 characters.

Address Type: Select IP address.

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

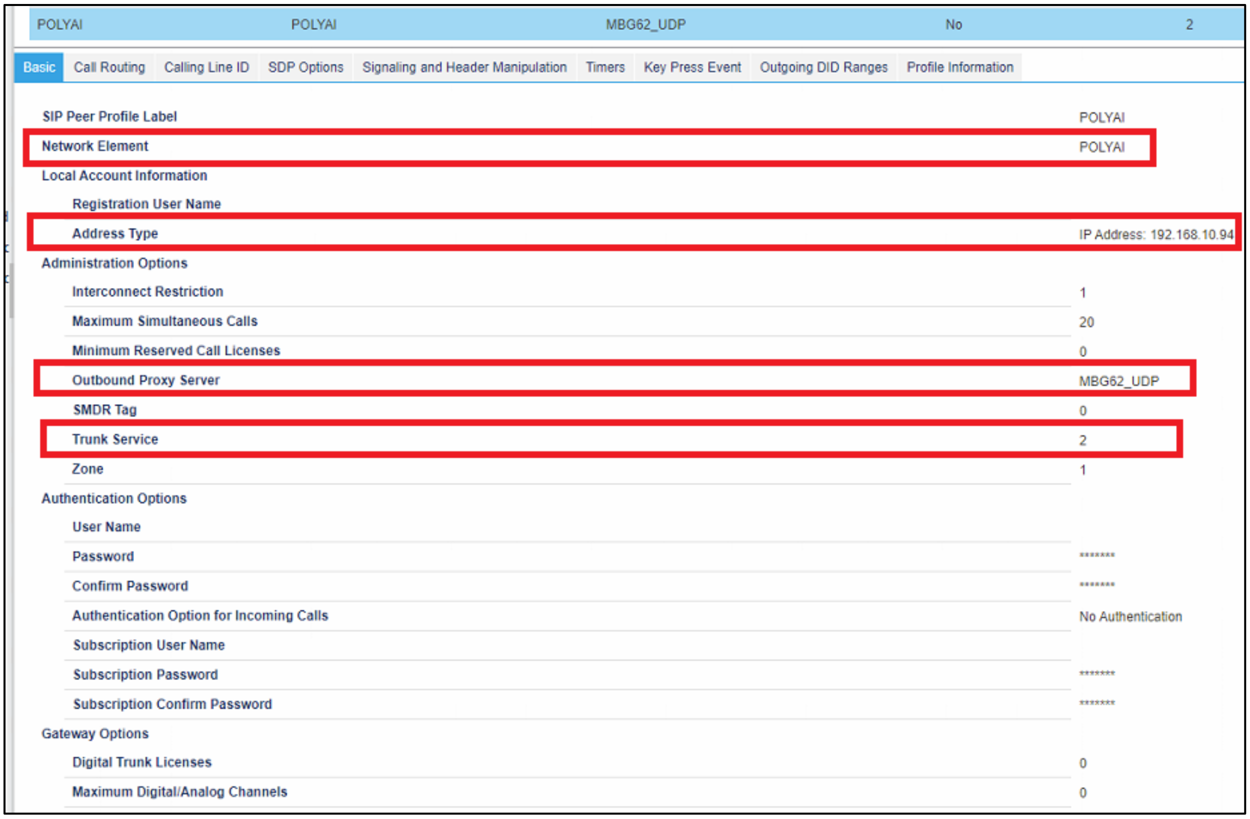
Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided by POLYAI.** Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add private numbers into the 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 POLYAI.

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

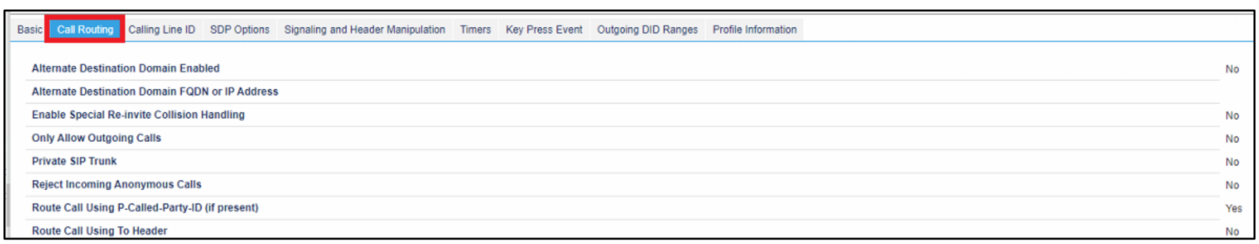


POLYAI	POLYAI	MBG62_UDP	No	2

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	POLYAI
Network Element	POLYAI
Local Account Information	
Registration User Name	
Address Type	IP Address: 192.168.10.94
Administration Options	
Interconnect Restriction	1
Maximum Simultaneous Calls	20
Minimum Reserved Call Licenses	0
Outbound Proxy Server	MBG62_UDP
SMDR Tag	0
Trunk Service	2
Zone	1
Authentication Options	
User Name	
Password	*******
Confirm Password	*******
Authentication Option for Incoming Calls	No Authentication
Subscription User Name	
Subscription Password	*******
Subscription Confirm Password	*******
Gateway Options	
Digital Trunk Licenses	0
Maximum Digital/Analog Channels	0

Figure 7 – 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 8 – 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 9 – 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	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 Codec Selection on Answer	No
Prevent the Use of IP Address 0.0.0.0 in SDP Messages	Yes
Reject Call without telephone-event payload	No
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	Yes

Figure 10 – 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
Discard Received P-Asserted-Identity Headers	No
Domain for Trunk Context	
Emergency Call Headers	CESID in From, [and PAI]
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
Mode for Out-of-Band DTMF	RFC 4733 DTMF
Multilingual Name Display	No
Only use SDP to decide 180 or 183	Yes
Prefer From Header for Caller ID	No
Q.850 Reason Headers	No
Require Reliable Provisional Responses on Outgoing Calls	Yes
Suppress Incoming Name	No
Suppress Redirection Headers	No
Use Fixed Retry Time for 491	No

Figure 11 – 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
User-Defined Header Name	
User-Defined Header Value	

Figure 12 – SIP Peer Profile Assignment- Signaling and Header Manipulation

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 13 – 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	No
Force Out Subscriptions for Remote Digit Monitoring	No
Request Outbound Proxy to Handle Out Subscriptions	No
KPML Transport	default
KPML Port	0

Figure 14 – 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

Update

Index	DID Range	CPN Substitution

Figure 15 – SIP Peer Profile Assignment- Outgoing DID Ranges

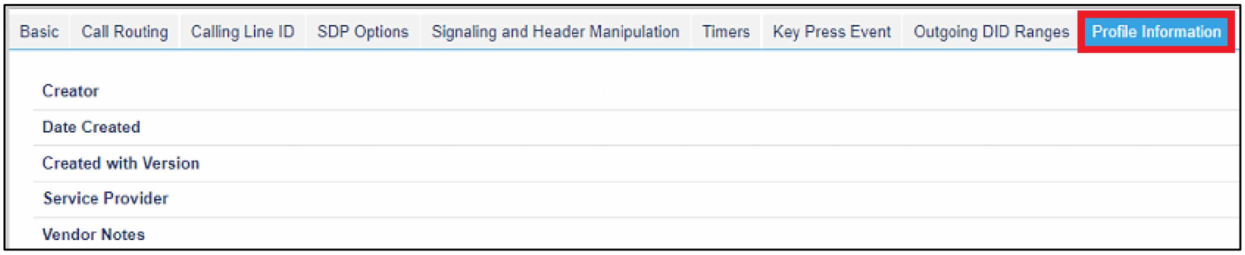


Figure 16 – SIP Peer Profile Assignment- Profile Information

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to POLYAI absorbs or injects additional digits according to your dialling plan. In this example, we will be absorbing 3 digits (in this case will be 111 to dial out).

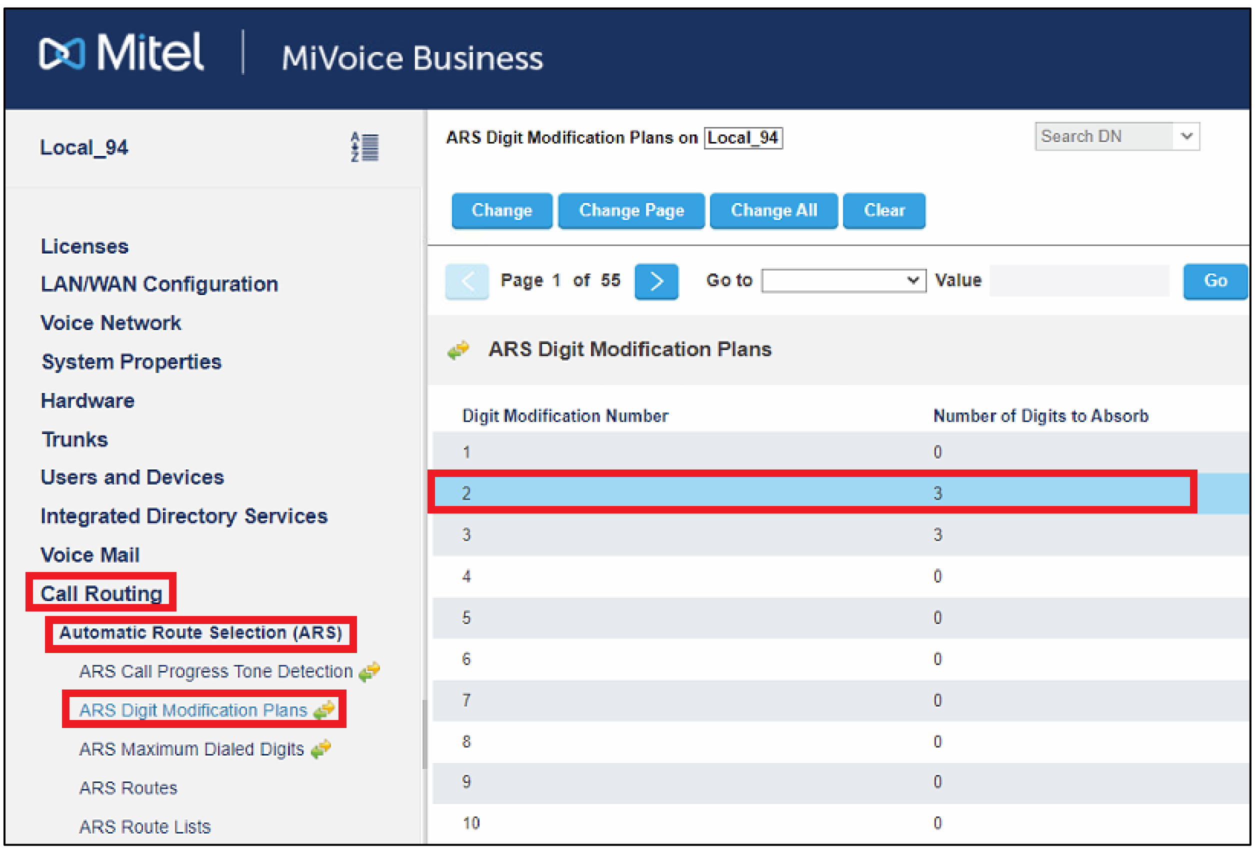


Figure 17 – Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to POLYAI. 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.

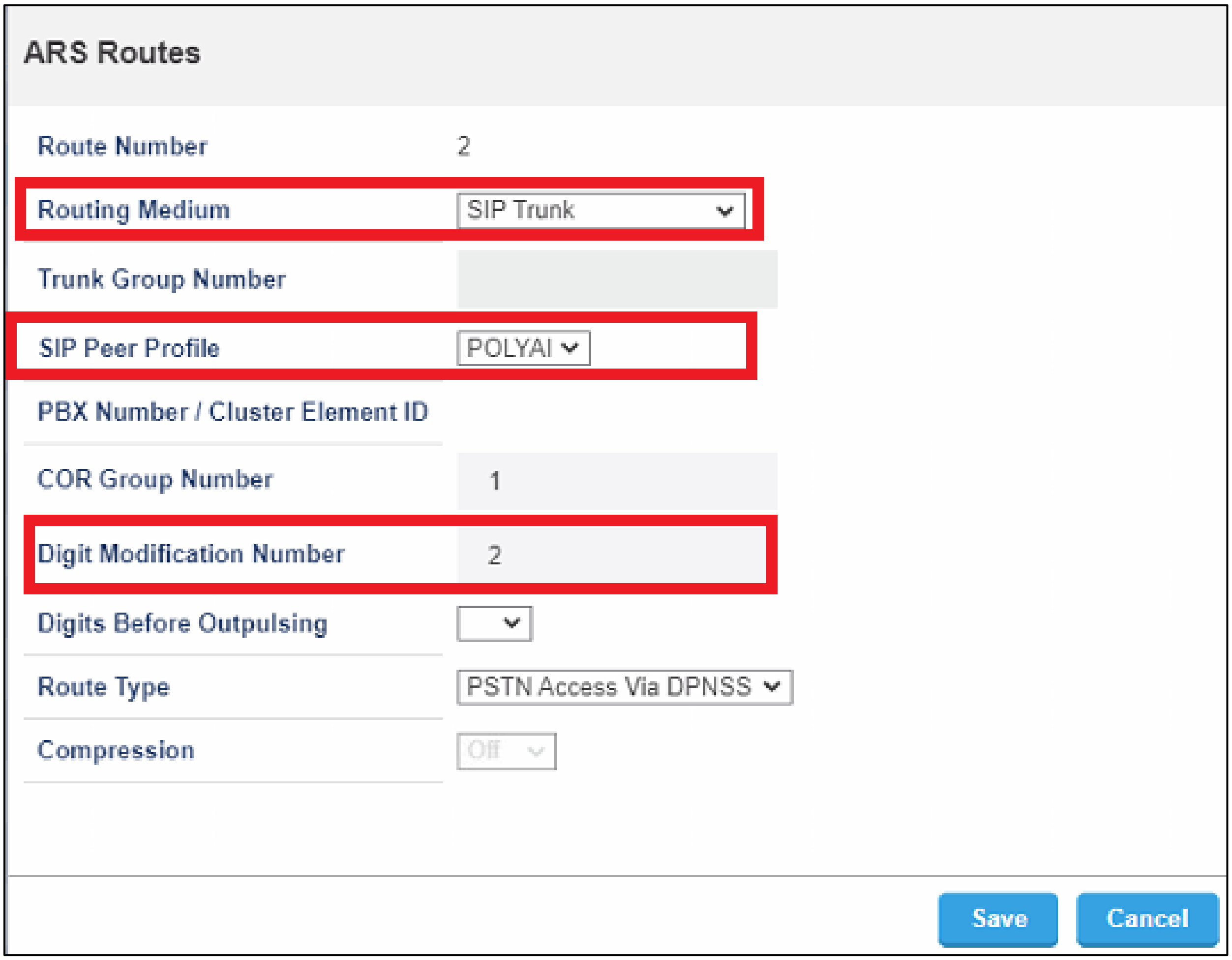


Figure 18 – SIP Trunk Route Assignment

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 111, the call will be routed to POLYAI.

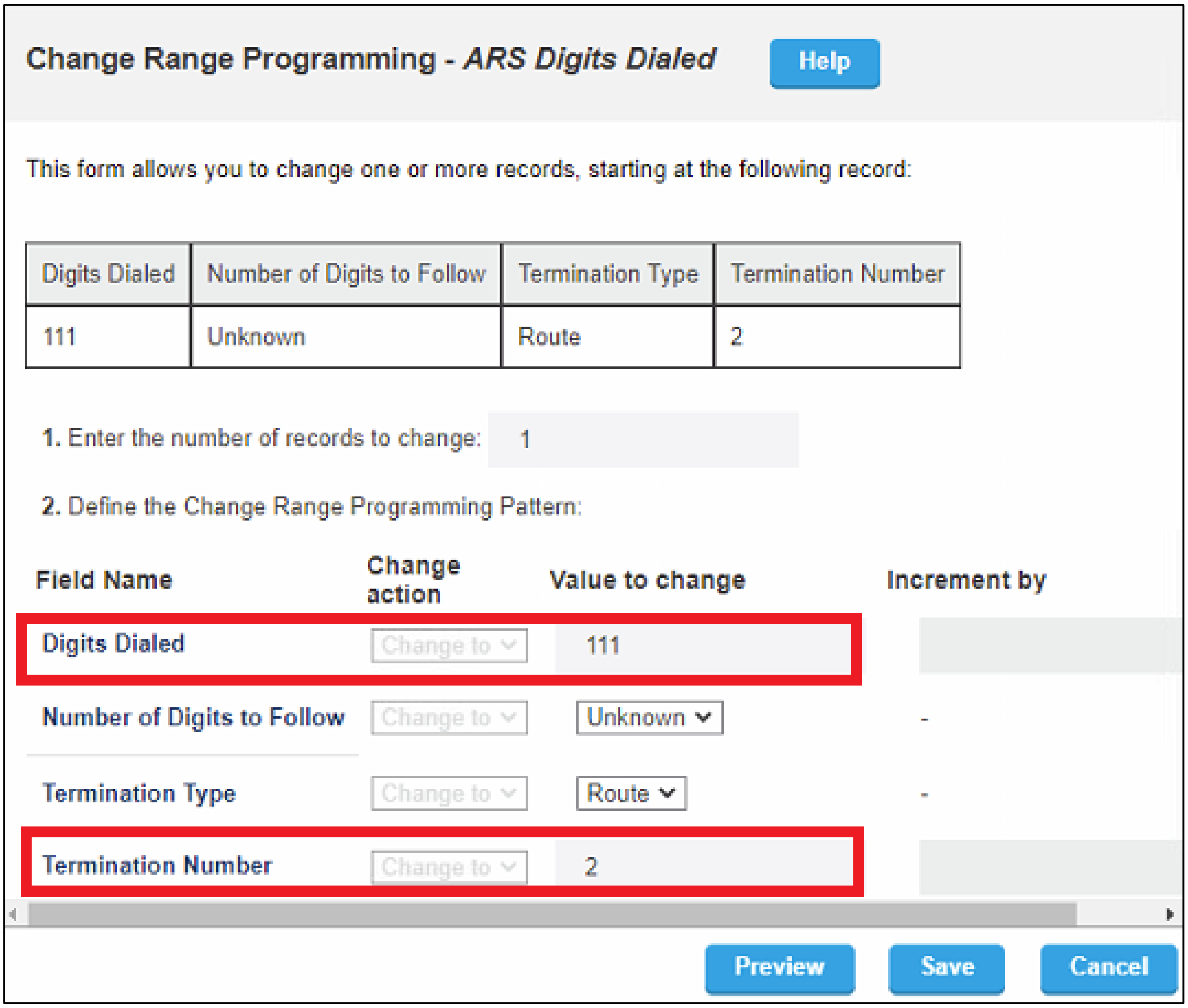


Figure 19 – ARS Digit Dialed Assignment

TLS Configurations

Make sure to configure POLYAI trunk over the TLS with port 5061 and below are the Network Elements configuration details for the TLS on the MiVB.

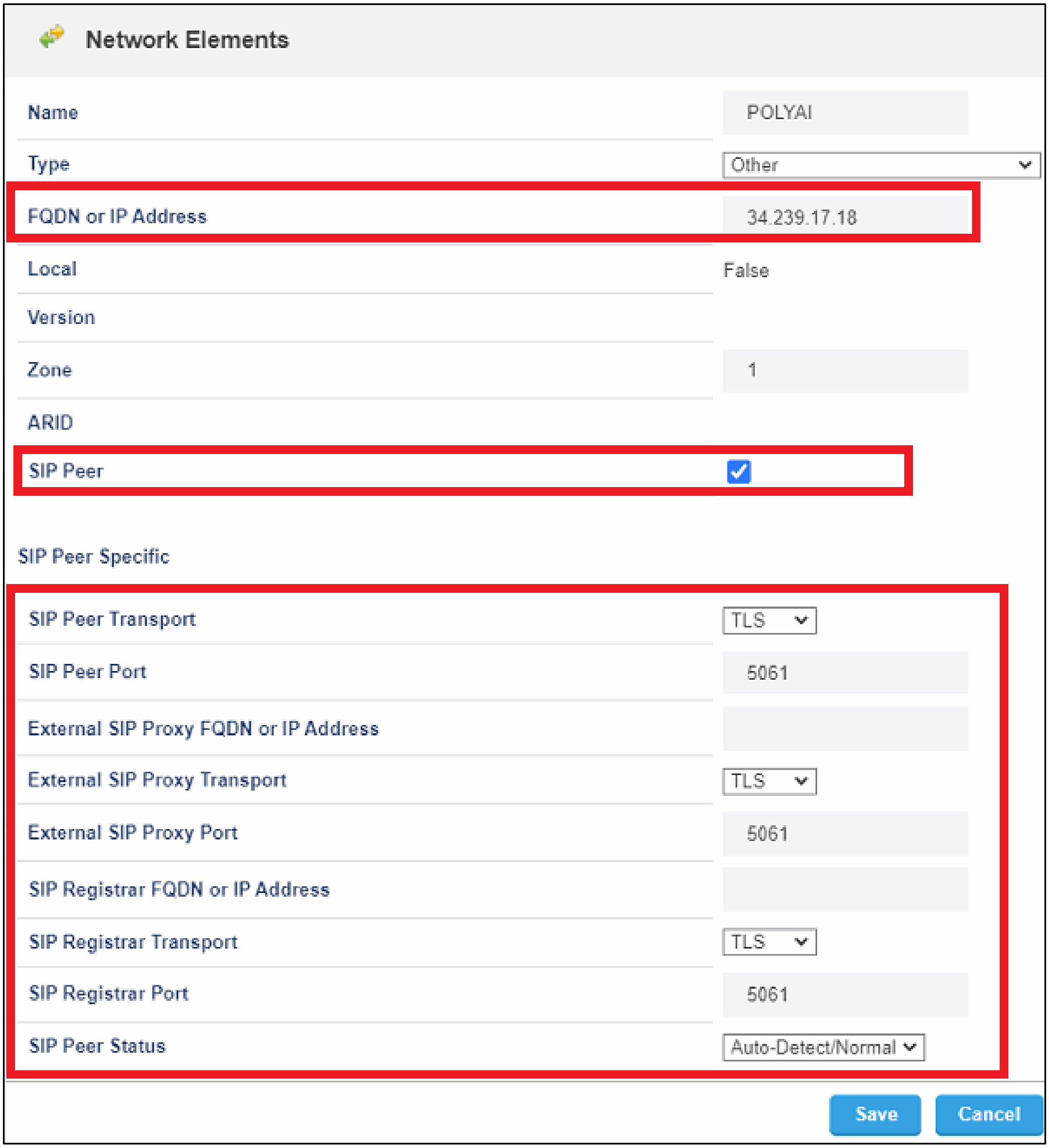


Figure 20 – Network Elements – POLYAI

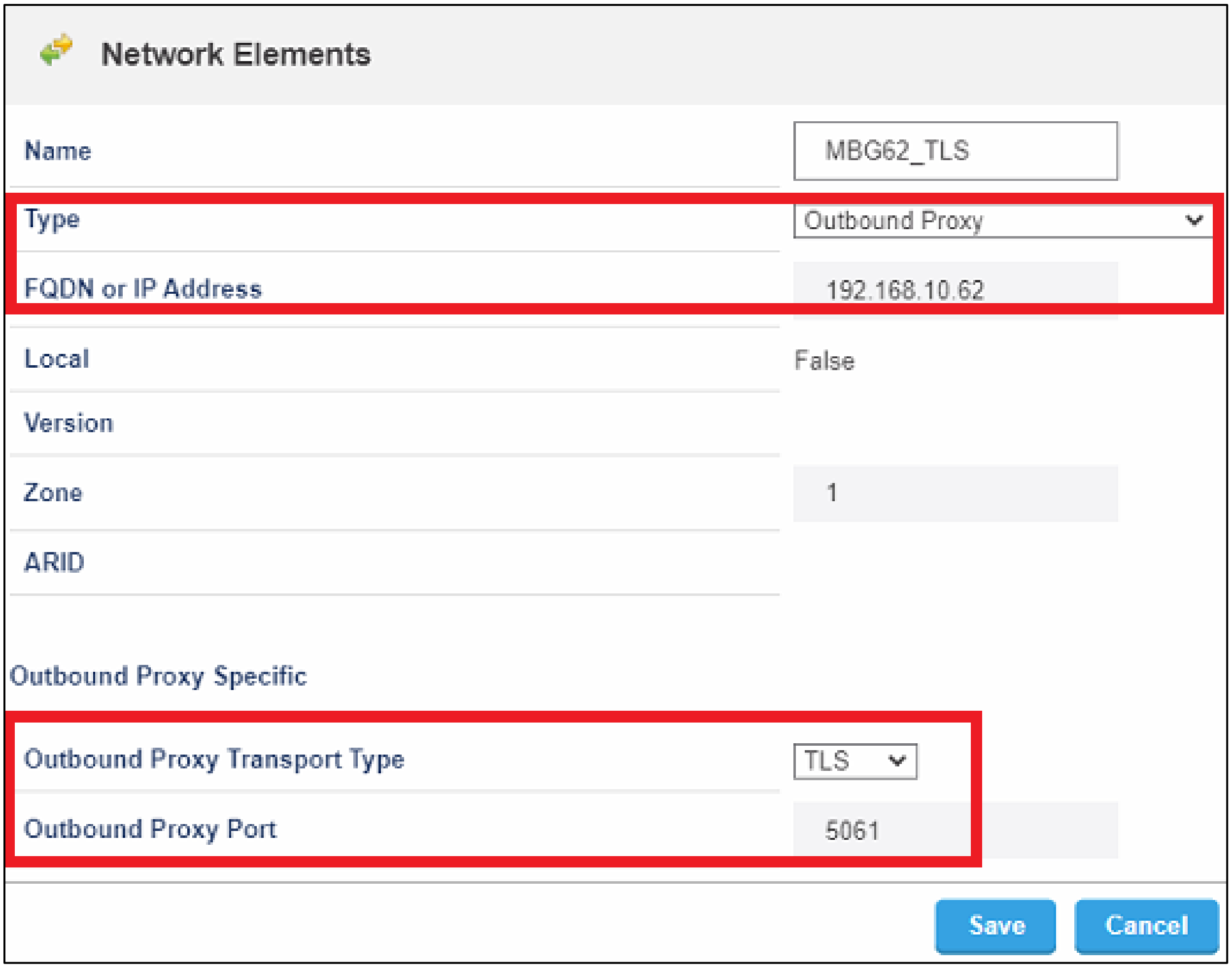


Figure 21 – Network Elements – MBG

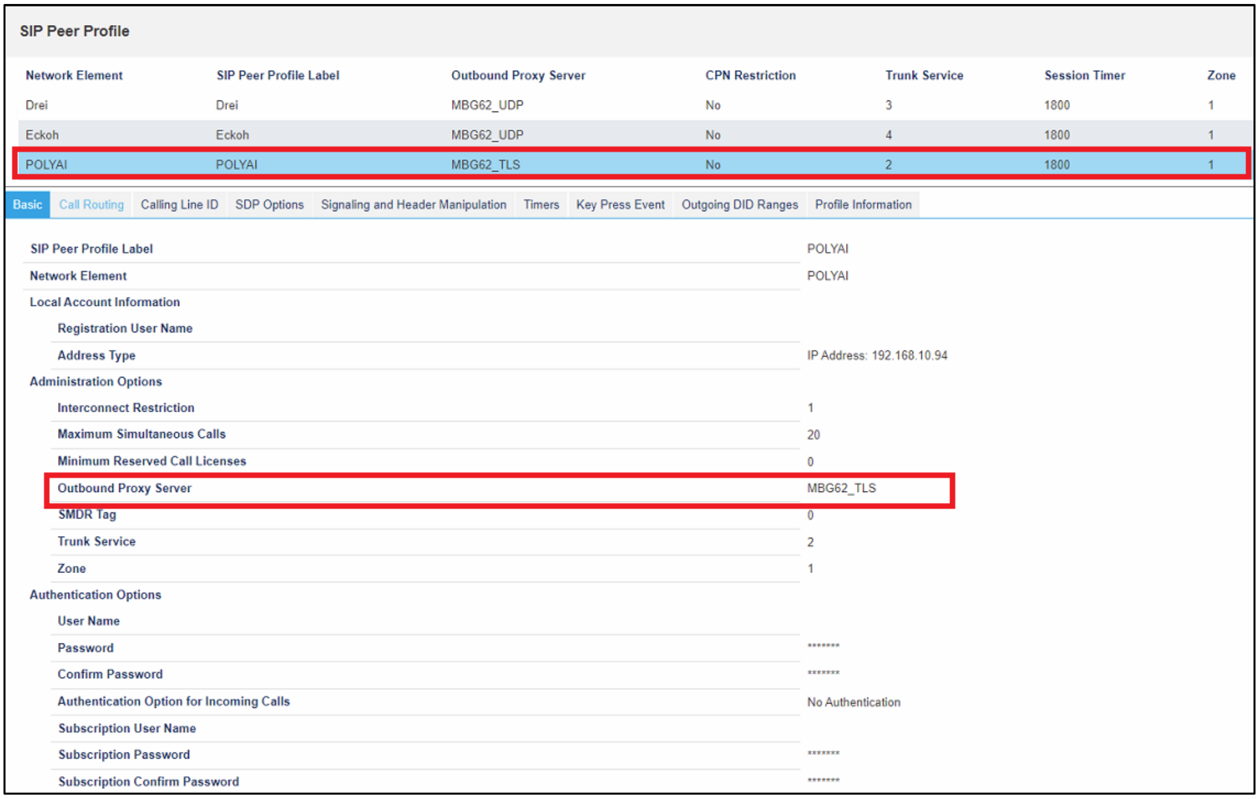


Figure 22 – SIP Peer Profile

MBG TLS trunk Configuration

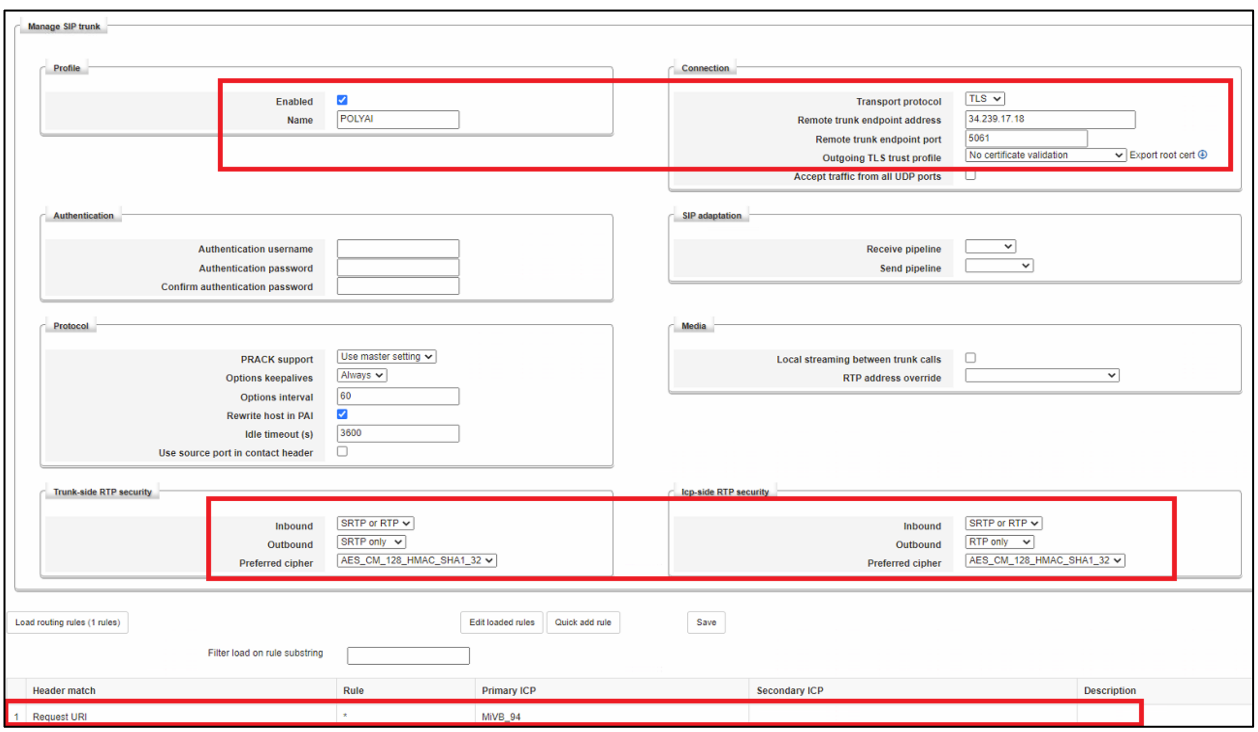


Figure 23 – MBG TLS Trunk Configuration

Note – As a part of this Interop testing for the TLS, we have used RTP/AVP between MiVB, MBG and RTP/SAVP has been used between MBG to POLYAI.

MiVoice Border Gateway Configuration Notes

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 the right pane, click **Network** tab and then **ICPs** (see Figure 24 for details)

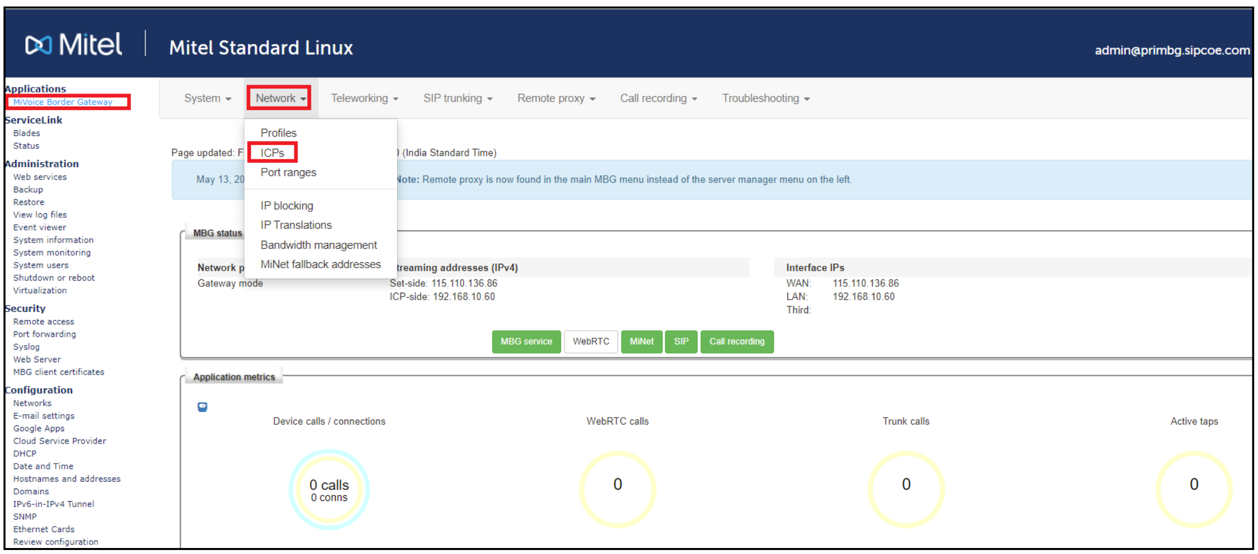


Figure 24 – MBG's Configuration page

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

Figure 25 – ICP configuration page

- Next configure the SIP trunking by click on the 'SIP Trunking' tab and selecting 'Configuration'. See figure 26.
- On the SIP Trunking Configuration page click on the '+' symbol and add POLYAI trunk, see Figure 27.

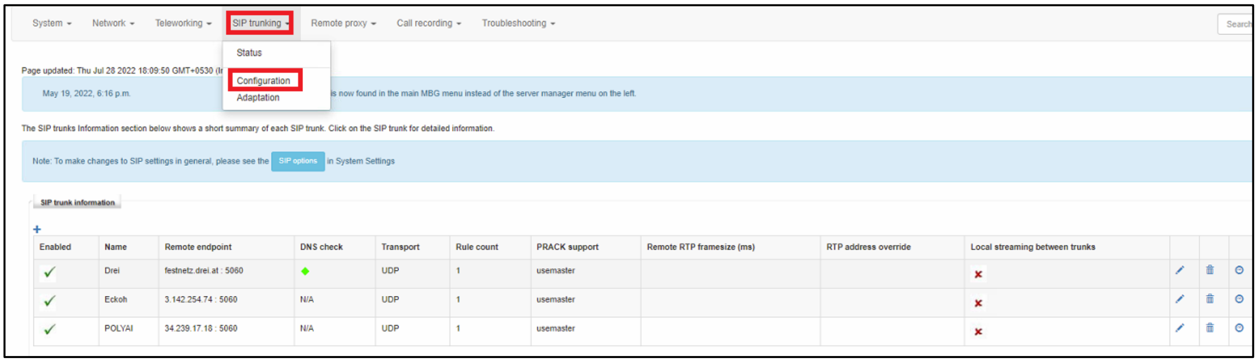


Figure 26 - MBG SIP Trunking Configuration

Enter the SIP Trunking details as shown in Figure 27:

Name: Is the name you want to call the trunk.

Remote trunk endpoint address: Is the public IP address of the provider's switch or gateway. This address should be given to you by the provider, e.g. POLYAI.

Remote trunk endpoint port: 5060.

Options Keepalives: Always.

Options interval: 60

RTP address override: Leave blank.

PRACK support: Use Master Setting.

Routing rule one: The example rule allows routing of any incoming digits to the selected MiVB.

The rest of the settings are optional and could be configured as required. Save the Trunking configuration.

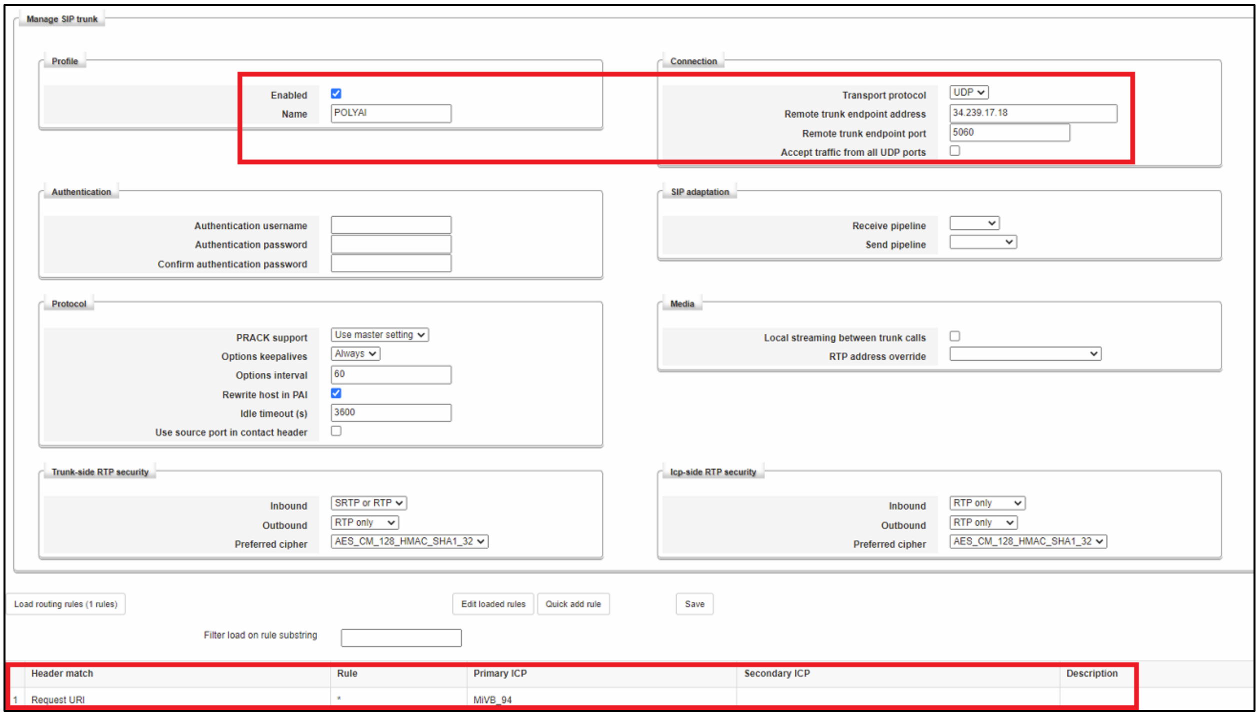


Figure 27 - MBG SIP Trunking Configuration

- Check status: click SIP Trunking and then click Status, see figure 28

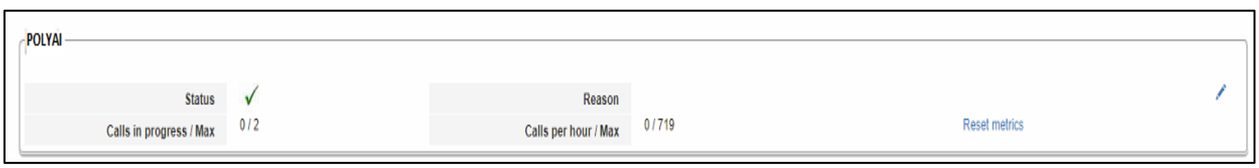


Figure 28 – SIP Trunk Status

Glossary

MiVoice Business	MiVB
MiVoice Border Gateway	MBG
MiNET Interface	MiNET
Mitel Solutions Alliance	MSA
Knowledge Management System	KMS
Class of Service	COS
Automatic Route Selection	ARS