Mitel Technical Configuration Notes – HO5582

October 24, 2025

MiVB - Configure MiVoice Business 10.4 for use with PolyAI using MBG

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

Environment: MiVoice Business 10.4 (10.4.0.21), MiVoice Border Gateway 12.2.0.72, Mitel 69xx/69xxw MiNET 03.00.00.064 and MiCollab Server 10.1.1.7-01

Mitel Technical Configuration Notes – Configure MiVoice Business 10.4 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	Initial interop with Mitel MiVB 9.4 and PolyAI using MBG.
2	October, 2025	Interop with Mitel MiVB 10.4 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	10.4 (10.4.0.21)	NA
Mitel	MiVoice Border Gateway	12.2.0.72	NA
Mitel	69xx/69xxw MiNET	03.00.00.064	NA
Mitel	MiCollab Server	10.1.1.7-01	NA
Mitel	MiCollab Client	10.1.9	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.	☑
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

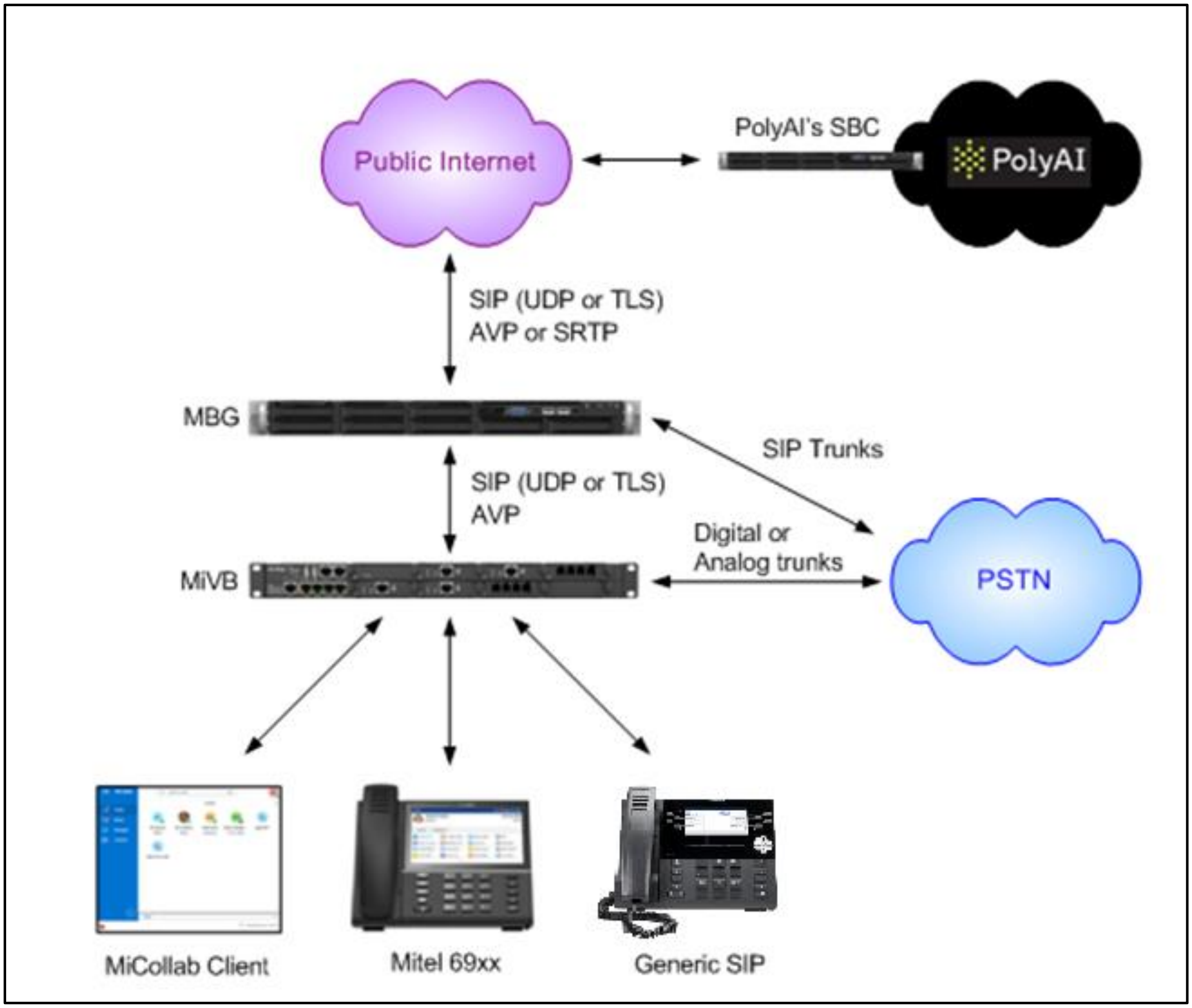


Figure 1 – Network Topology

Call Flow

Calls to PolyAI can be initiated by any device on the MiVB, or external PSTN calls may be transferred to PolyAI via Automatic Route Selection (ARS). PolyAI will interact with the caller and either provide the requested information or transfer the caller to a different destination. If the caller needs to be transferred elsewhere, PolyAI will perform a SIP-REFER to transfer the caller to the specific Directory Number (DN) on the MiVB.

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 or TLS on Port 5061.

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.

For example, if a customer uses SIP trunks (through an MBG) for connecting to PSTN, and those calls are being handled by PolyAI, then each concurrent call from PSTN to PolyAI will consume two (2) MiVB SIP Trunk Licenses and two (2) MBG SIP Proxy license. In contrast, if an internal extension calls PolyAI, then only one (1) MiVB SIP Trunk License and one (1) MBG SIP Proxy license will be consumed. Both MiVB SIP Trunk Licenses and MBG SIP Proxy licenses are concurrent-use licenses, and do not need to be dedicated to PolyAI.

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.

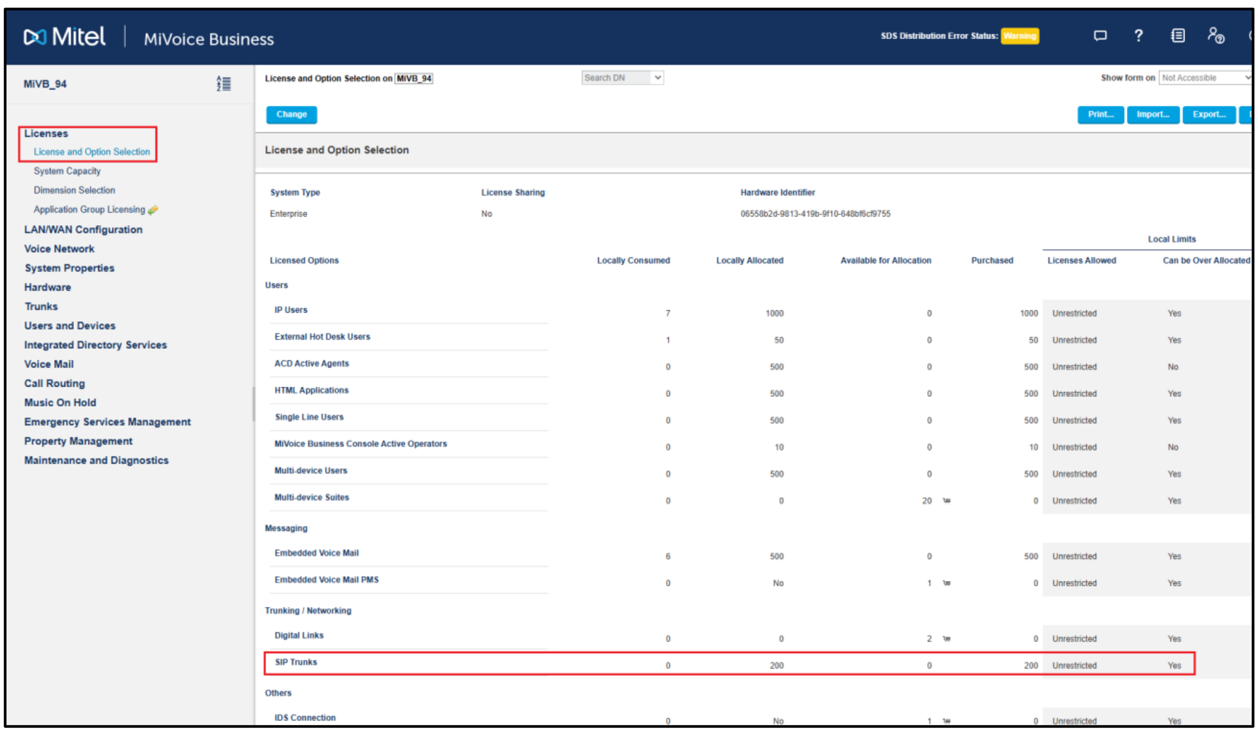


Figure 2 – License and Option Selection

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 at a minimum, the settings below are changed from the defaults:

- Add a descriptive name for the COS in the Comment Field
- Public Network Access via DPNSS set to Yes
- Campon Tone Security/FAX Machine set to Yes
- Busy Override Security set to Yes

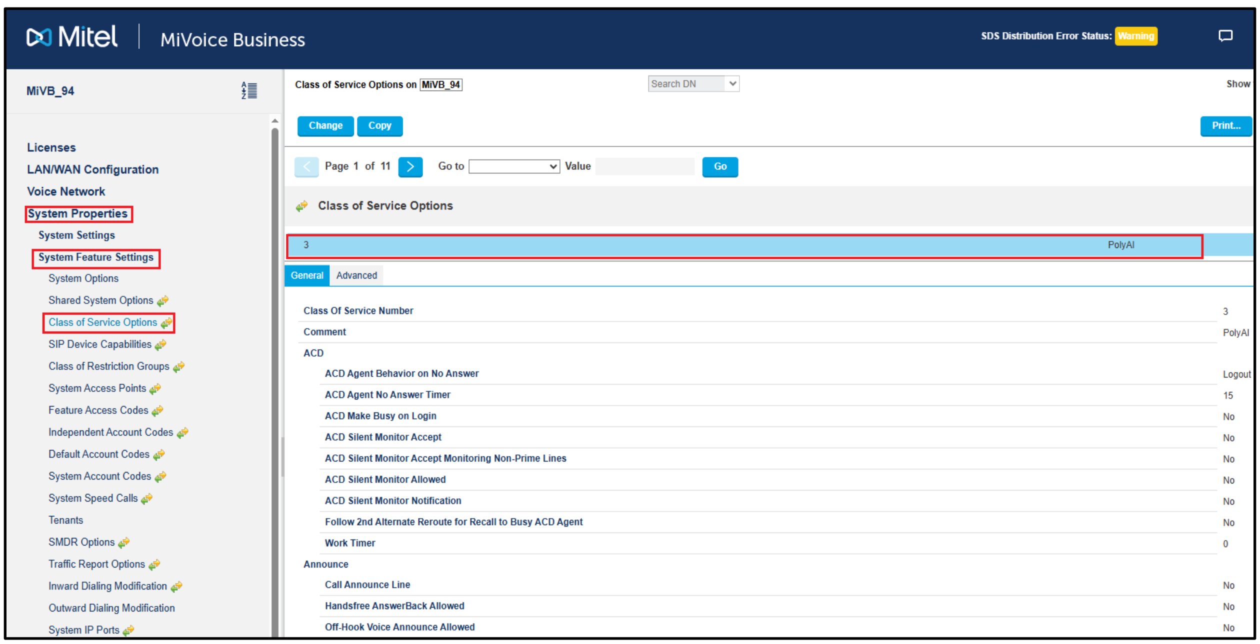


Figure 3 – Class of Service

Class of Restriction (COR)

You should verify that any device that will place calls to PolyAI (including stations and PSTN trunks) is assigned a Class of Restriction (COR) which will not block calls to PolyAI. In the example below, COR 2 is assigned to the phones and trunks as well as PolyAI. Since COR 2 is not specifically included in the Class of Restriction for Group list as shown below, any device or trunk using COR 2 will be able to access PolyAI trunks:

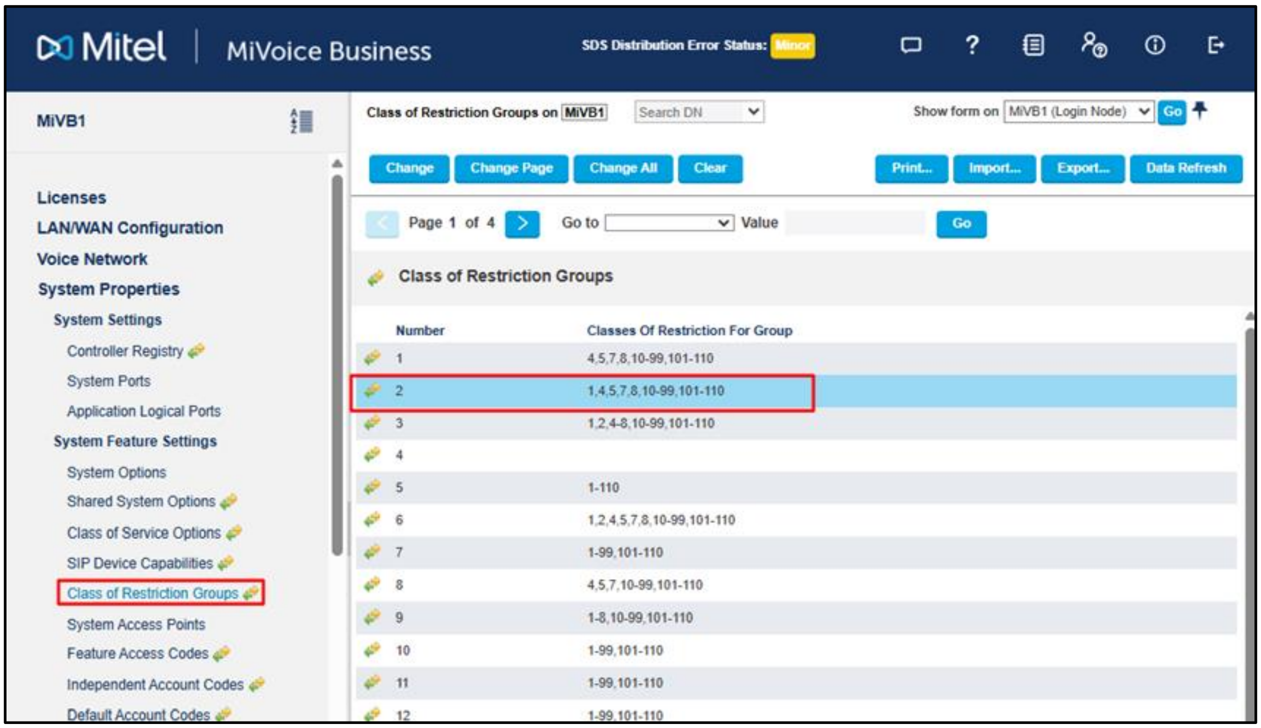


Figure 4 – Class of Restriction

Network Element Assignment (PolyAI Trunks)

Create two network elements for PolyAI. PolyAI will provide their Primary and Secondary SBCs IP addresses prior to deployment. PolyAI supports both UDP or TLS, however Mitel recommends using TLS for connectivity to PolyAI over the public internet.

- Set the **Name** appropriately, set the **Type** to “Other” and enter the **FQDN or IP address** of PolyAI’s SBCs.
- Ensure that **SIP Peer** is checked.
- If you are using UDP, set the **Transport/Proxy/Registrar** values to UDP and port 5060 as shown below. If using TLS, set these values to TLS and the port to 5061.

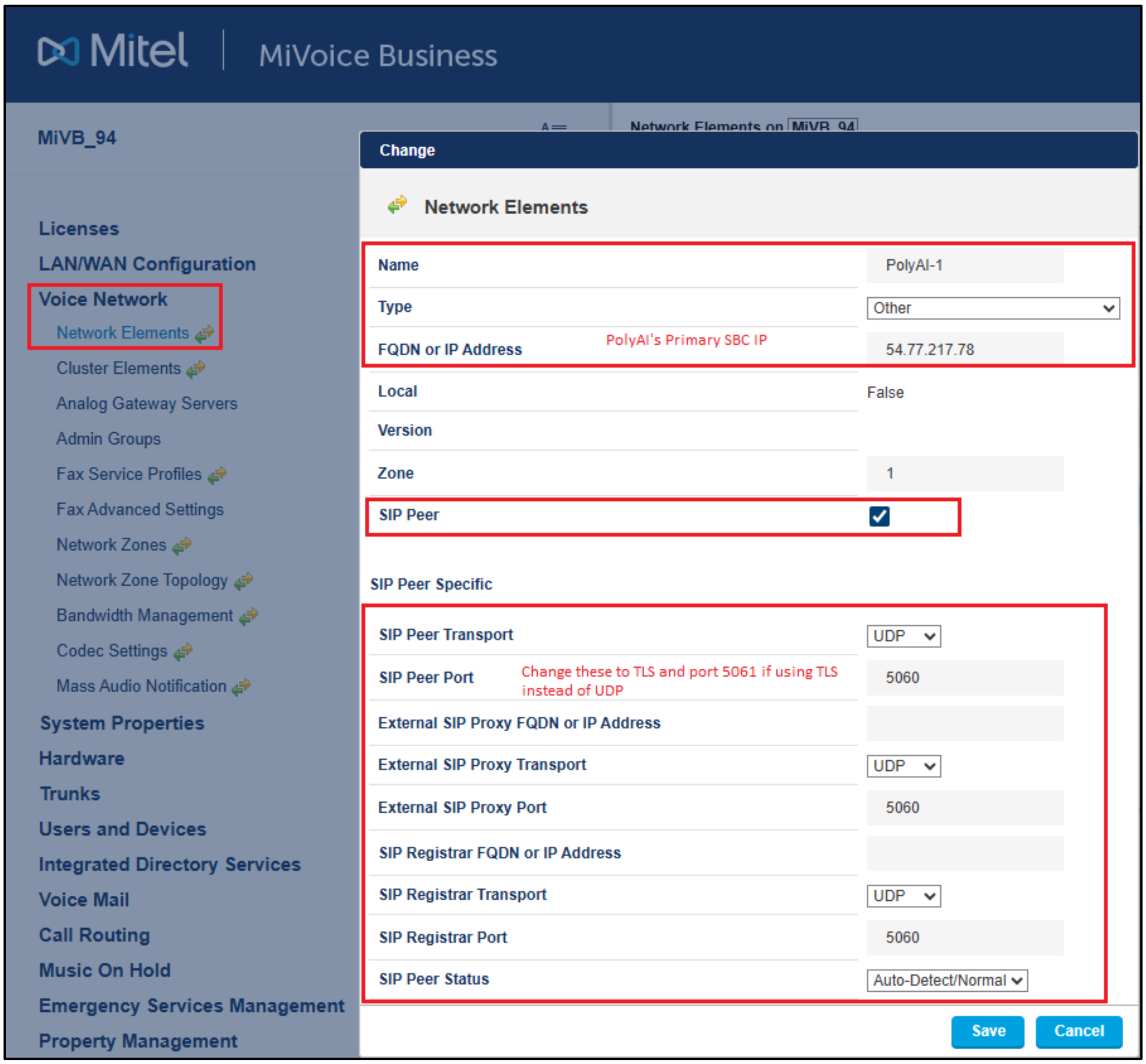


Figure 5a – Network Element Assignment (Primary PolyAI SBC)

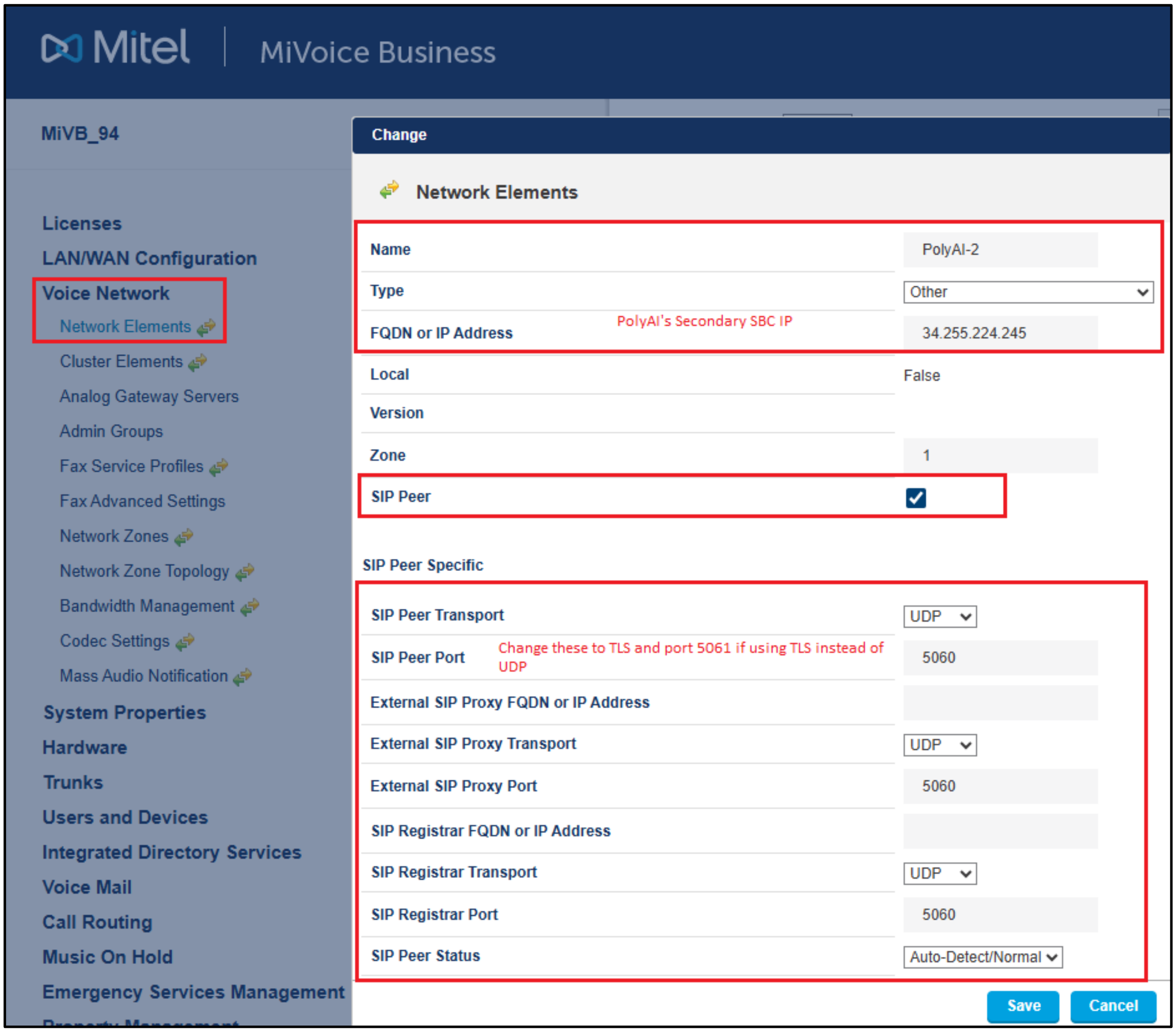


Figure 5b – Network Element Assignment (Secondary PolyAI SBC)

Network Element Assignment (MBG Proxy)

A MiVoice Border Gateway (MBG) will be needed to proxy all traffic between PolyAI and the MiVB. As such, there needs to be a Network Element configured for the MBG Proxy. For existing installations using SIP trunking, it is likely there is already a Network Element for the MBG Proxy, but it will be configured to work with the existing PSTN carrier's Transport Type (TCP/UDP or TLS) and ports (5060 or 5061). If the carrier's settings are the same as what you plan to use for PolyAI, then no change needs to be made. If the existing MBG Proxy Network Element that is used for PSTN SIP trunks does not match, you will need to create a new Network Element for the MBG Proxy and assign the proper Transport Type and ports:

- Set the **Name** appropriately, set the **Type** to "Outbound Proxy", and enter the **FQDN or IP address** of the MBG.
- Set the desired **Transport Type** and **Ports** (TLS / 5061 or UDP / 5060).

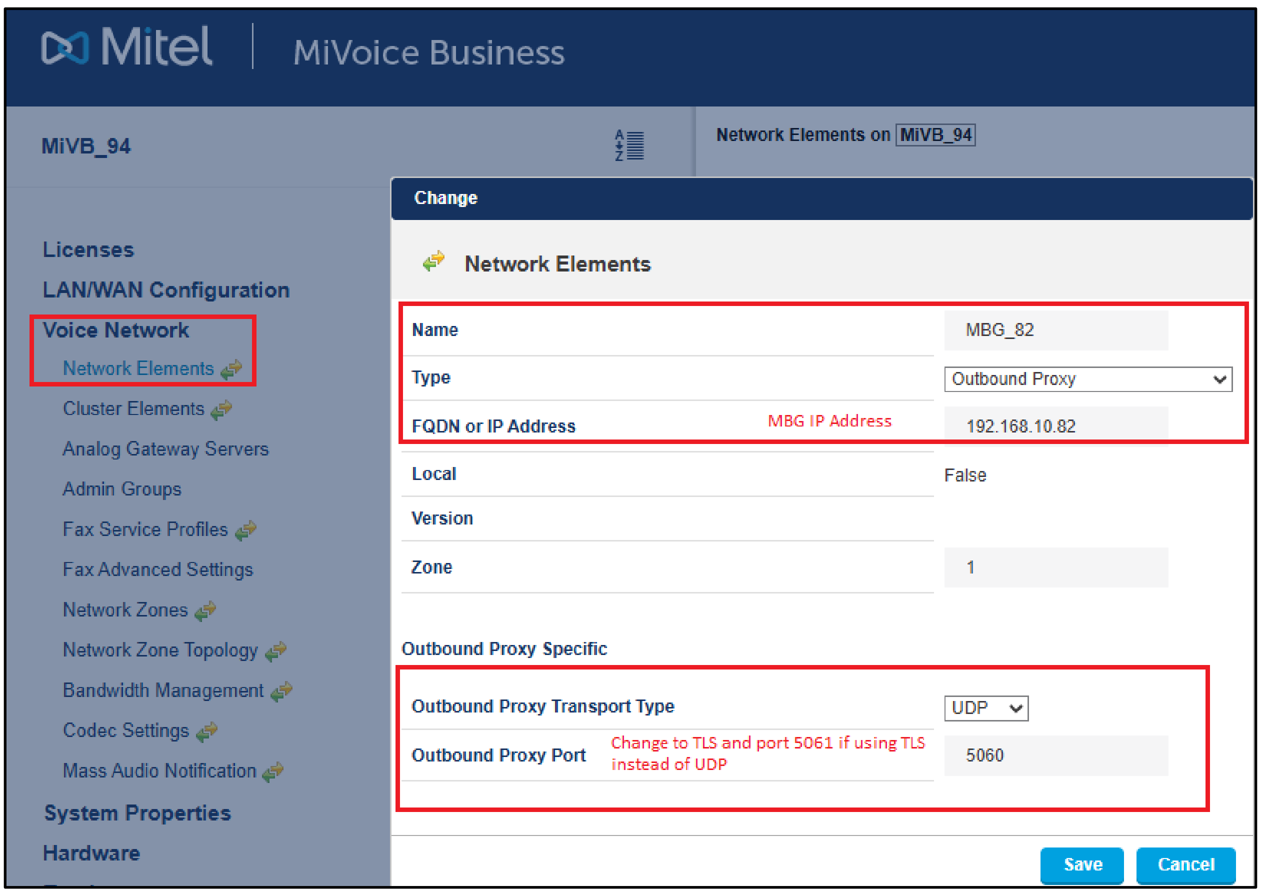


Figure 6 – Network Element Assignment (MBG Proxy)

Trunk Attributes

The Trunk Service Number is configured on the Trunk Attributes form.

- Select an unused **Trunk Service Number**, assign appropriate **COS and COR** from the previous steps.
- Set **Dial in Trunks Incoming Digit Modification** – **Absorb to 0**. Note: Leaving this field blank will cause calls to fail.
- Set the **Trunk Label** to a descriptive name.

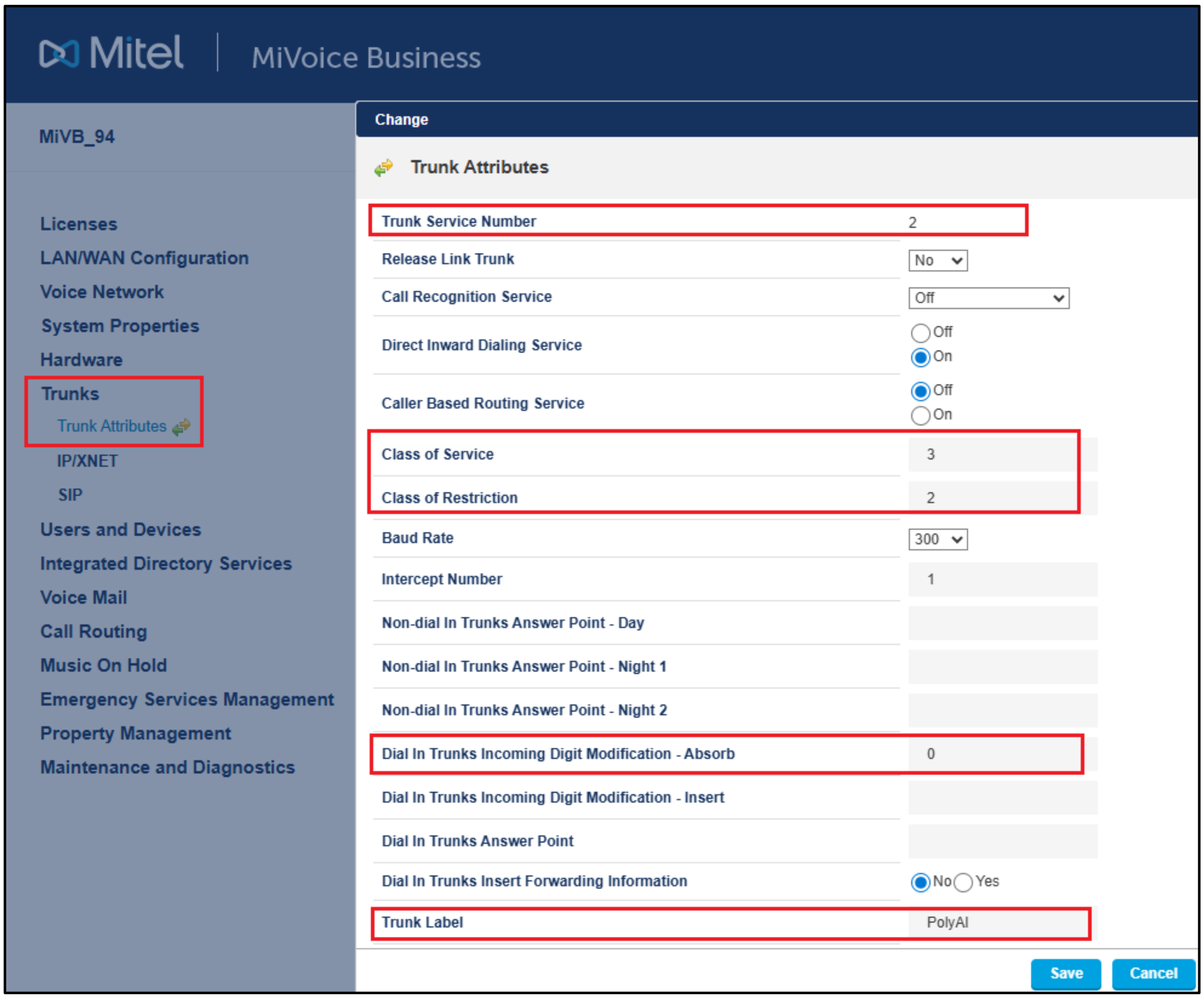


Figure 7 – Trunk Attributes

SIP Peer Profile

Two new SIP Peer Profiles will need to be created, one for each PolyAI SBCs (Primary and Secondary). The SIP Peer Profiles should be configured with the following options:

- **SIP Peer Profile Label**: Add a descriptive name for the Primary and Secondary PolyAI SIP Peer Profiles.
- **Network Element**: Each SIP Peer Profile needs to be associated with the associated previously created Network Element. For example, SIP Peer Profile "PolyAI1" should be associated with Network Element "PolyAI-1" that was previously created.
- **Registration Username**: Leave blank.
- **Address Type**: Enter FQDN or IP address of the MiVB.
- **Maximum Simultaneous Calls**: If desired, you can restrict the maximum number of concurrent calls to PolyAI that will be allowed. Since SIP trunk licenses are consumed when in use and are shared across the entire system, this value can be particularly important if the quantity of SIP trunk licenses on the system is constrained. Note that this number must be less than the total number of SIP trunk licenses on the system.
- **Minimum Reserved Call Licenses**: Enter a non-zero value here if you wish to reserve a specific quantity of SIP Trunk licenses for accessing PolyAI.
- **Outbound Proxy Server**: Select the MBG Proxy Network Element that was previously configured for both SIP Peer Profiles.
- **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 defaults to 0).
- **Authentication**: No usernames or passwords are required for PolyAI and should be left blank. PolyAI uses whitelisting to authorize SIP connections. You will be required to provide the public FQDN or IP address of your MBG during the initial coordination with PolyAI.

NOTE: Please configure both SIP Peer Profiles as per the snippets below. The secondary PolyAI SBC's profile is identical to the primary, so no changes are required except SIP Peer Profile "PolyAI2" should be associated with Network Element "PolyAI-2" that was previously created.

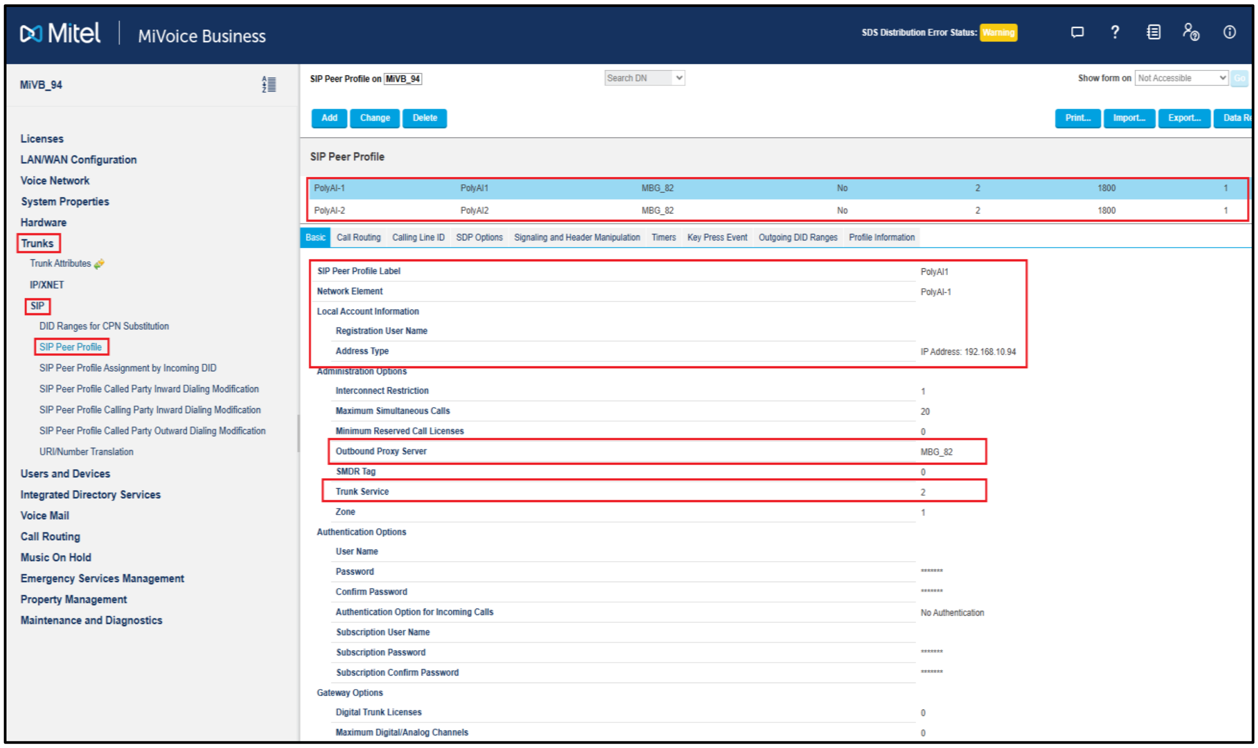


Figure 8a – SIP Peer Profile Assignment - Basic (Primary PolyAI SBC)

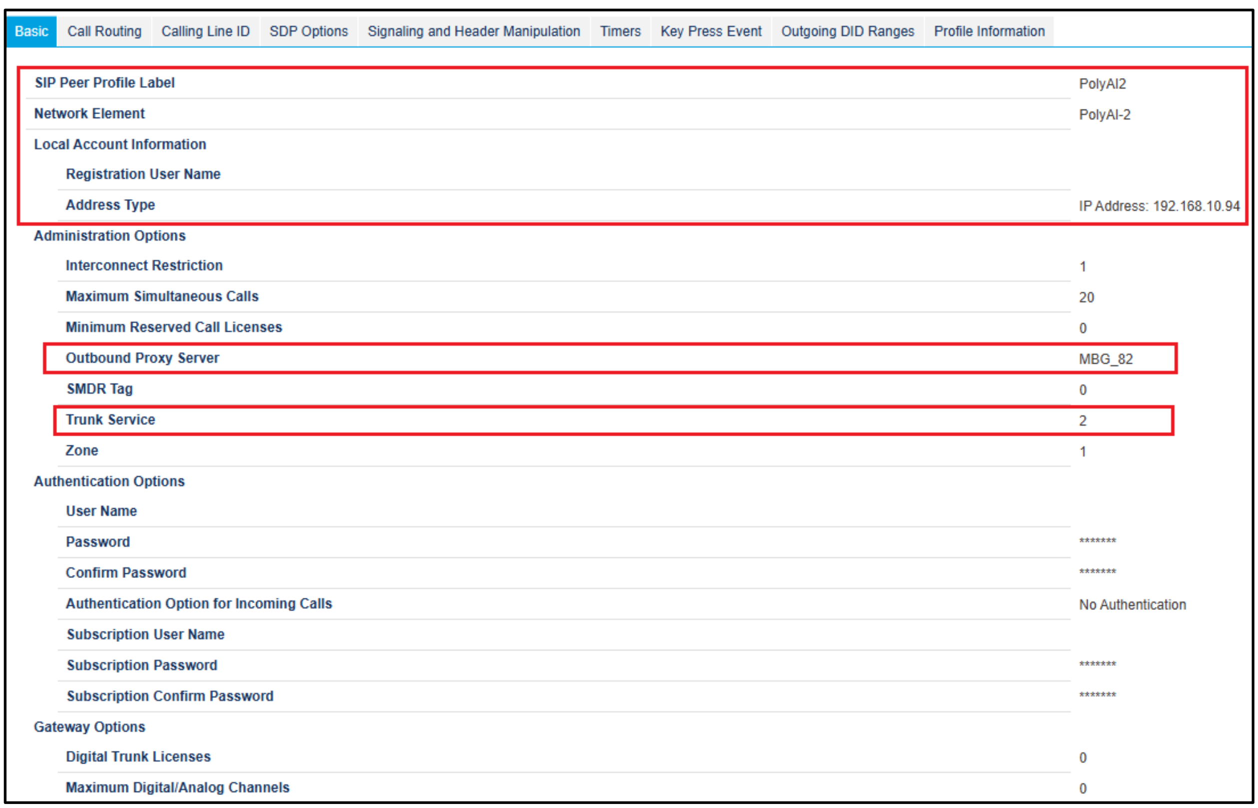
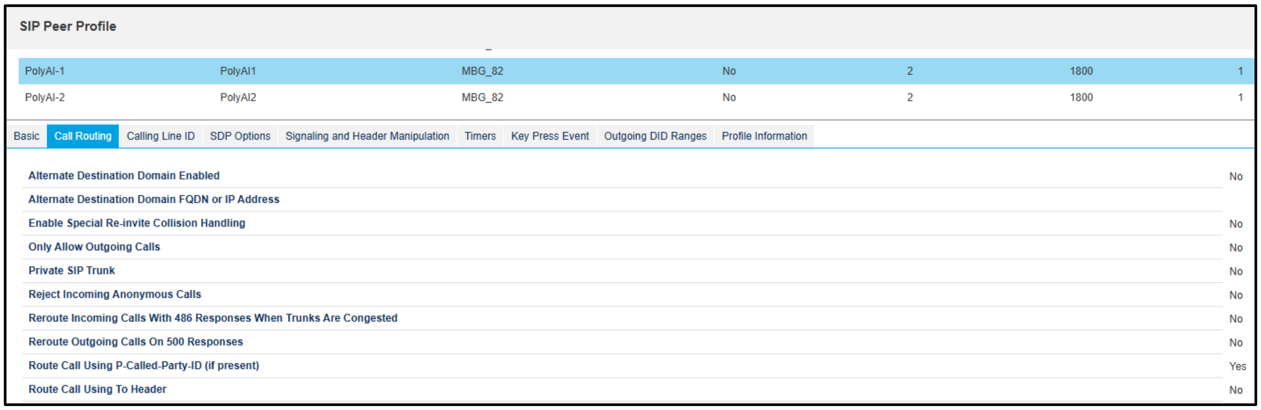


Figure 8b – SIP Peer Profile Assignment - Basic (Secondary PolyAI SBC)



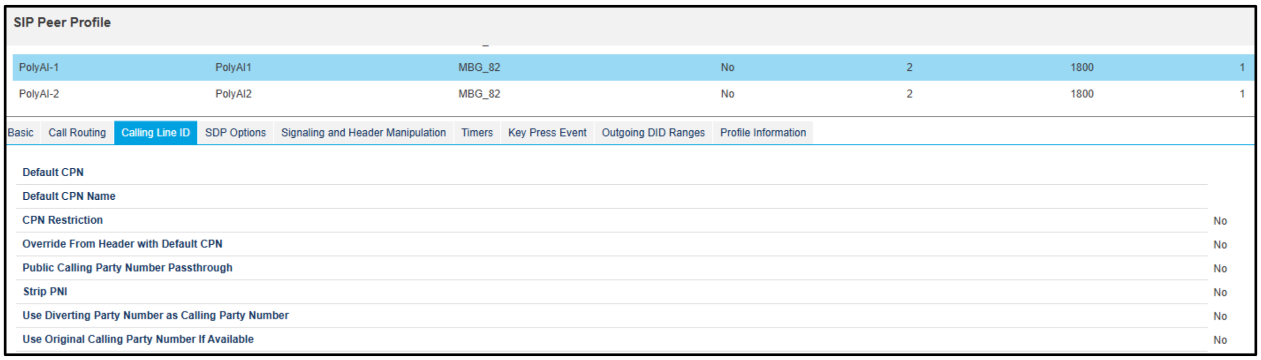
SIP Peer Profile

PolyAI-1	PolyAI1	MBG_82	No	2	1800	1
PolyAI-2	PolyAI2	MBG_82	No	2	1800	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
Reroute Incoming Calls With 486 Responses When Trunks Are Congested	No
Reroute Outgoing Calls On 500 Responses	No
Route Call Using P-Called-Party-ID (if present)	Yes
Route Call Using To Header	No

Figure 9 – SIP Peer Profile Assignment - Call Routing



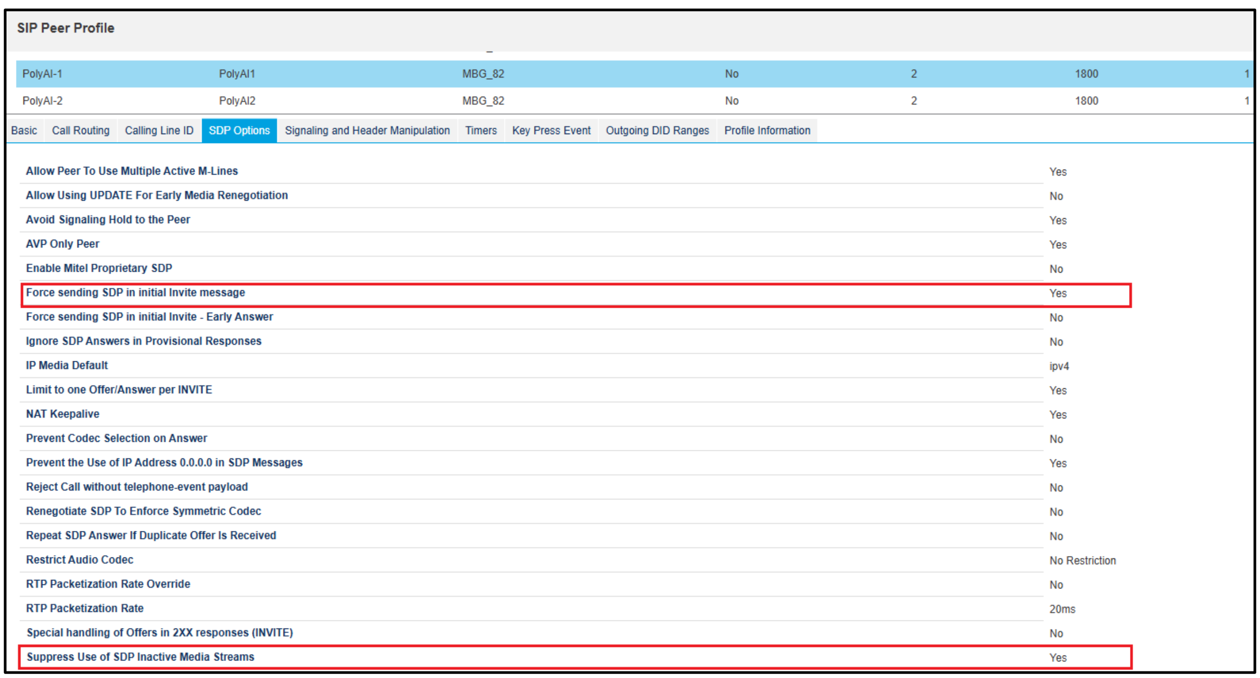
SIP Peer Profile

PolyAI-1	PolyAI1	MBG_82	No	2	1800	1
PolyAI-2	PolyAI2	MBG_82	No	2	1800	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 10 – SIP Peer Profile Assignment - Calling Line ID



SIP Peer Profile

PolyAI-1	PolyAI1	MBG_82	No	2	1800	1
PolyAI-2	PolyAI2	MBG_82	No	2	1800	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 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 11 – SIP Peer Profile Assignment - SDP Options

SIP Peer Profile

PolyAI-1	PolyAI1	MBG_82	No	2	1800	1
PolyAI-2	PolyAI2	MBG_82	No	2	1800	1

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
Use Privacy: none	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
User-Defined Header Name	
User-Defined Header Value	
Retry Registration On 404 Response From Peer	No

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

SIP Peer Profile

PolyAI-1	PolyAI1	MBG_82	No	2	1800
PolyAI-2	PolyAI2	MBG_82	No	2	1800

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	60
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

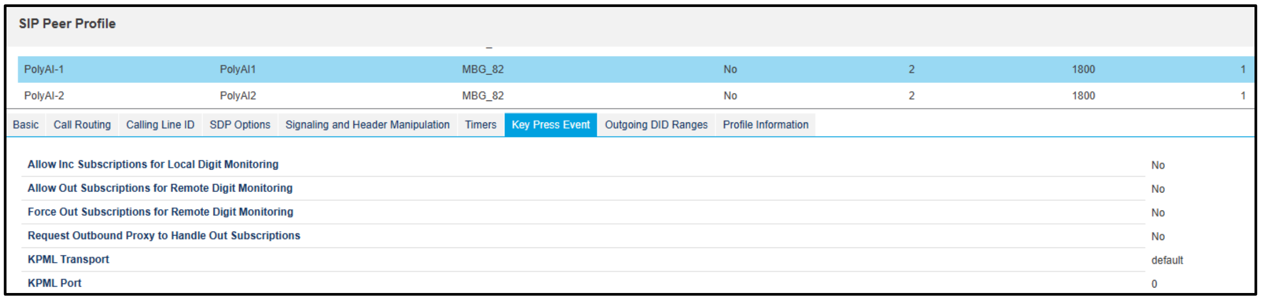


Figure 14 – SIP Peer Profile Assignment - Key Press Event

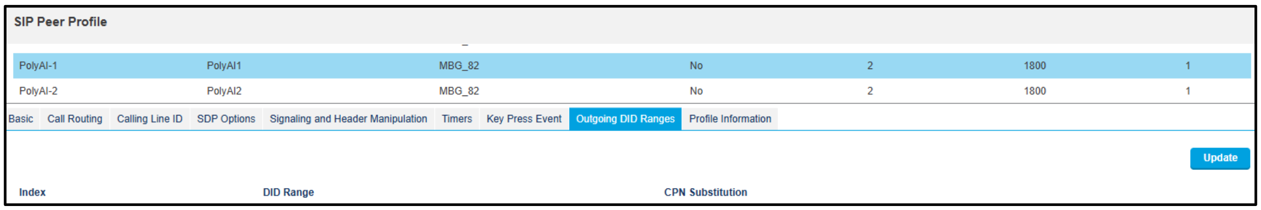


Figure 15 – SIP Peer Profile Assignment - Outgoing DID Ranges

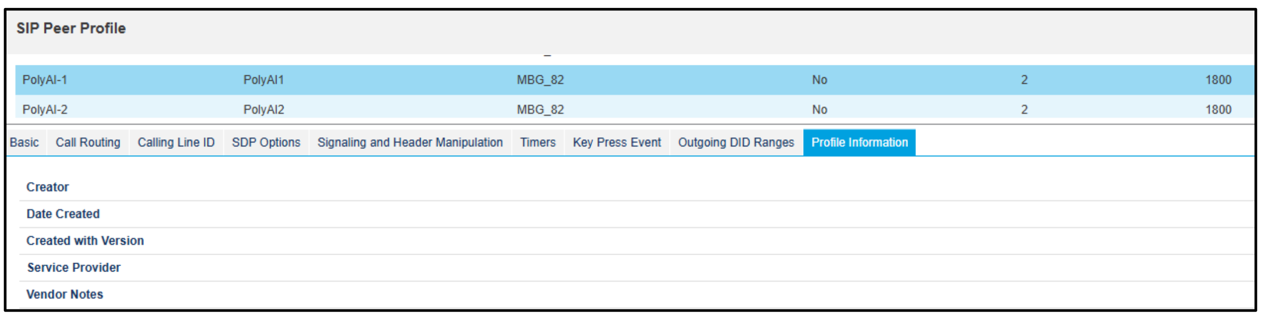


Figure 16 – SIP Peer Profile Assignment- Profile Information

ARS Digit Modification Plans

PolyAI does not require any additional digits to be added or absorbed. Ensure that you have a Digit Modification Plan that neither absorbs nor inserts any digits.

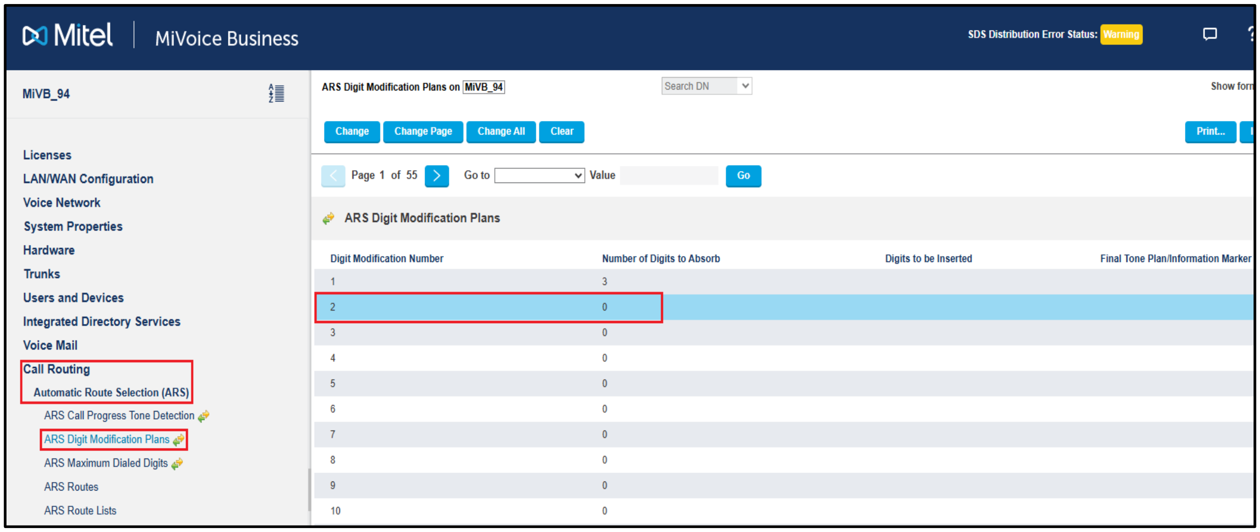


Figure 17 – Digit Modification Assignment

ARS Routes

Two ARS Routes are required for PolyAI, one for each SIP Peer Profile. In the step after this one, they will be placed into the same ARS Route List.

Create two new ARS Routes as shown below:

- Select an unused **Route Number** for each.
- **Routing Medium**: Select SIP Trunk.
- **SIP Peer Profile**: Select the associated PolyAI SIP Peer Profile (PolyAI Primary for the first ARS Route, Secondary for the second ARS Route).
- **COR Group Number**: Use the previously created COR.

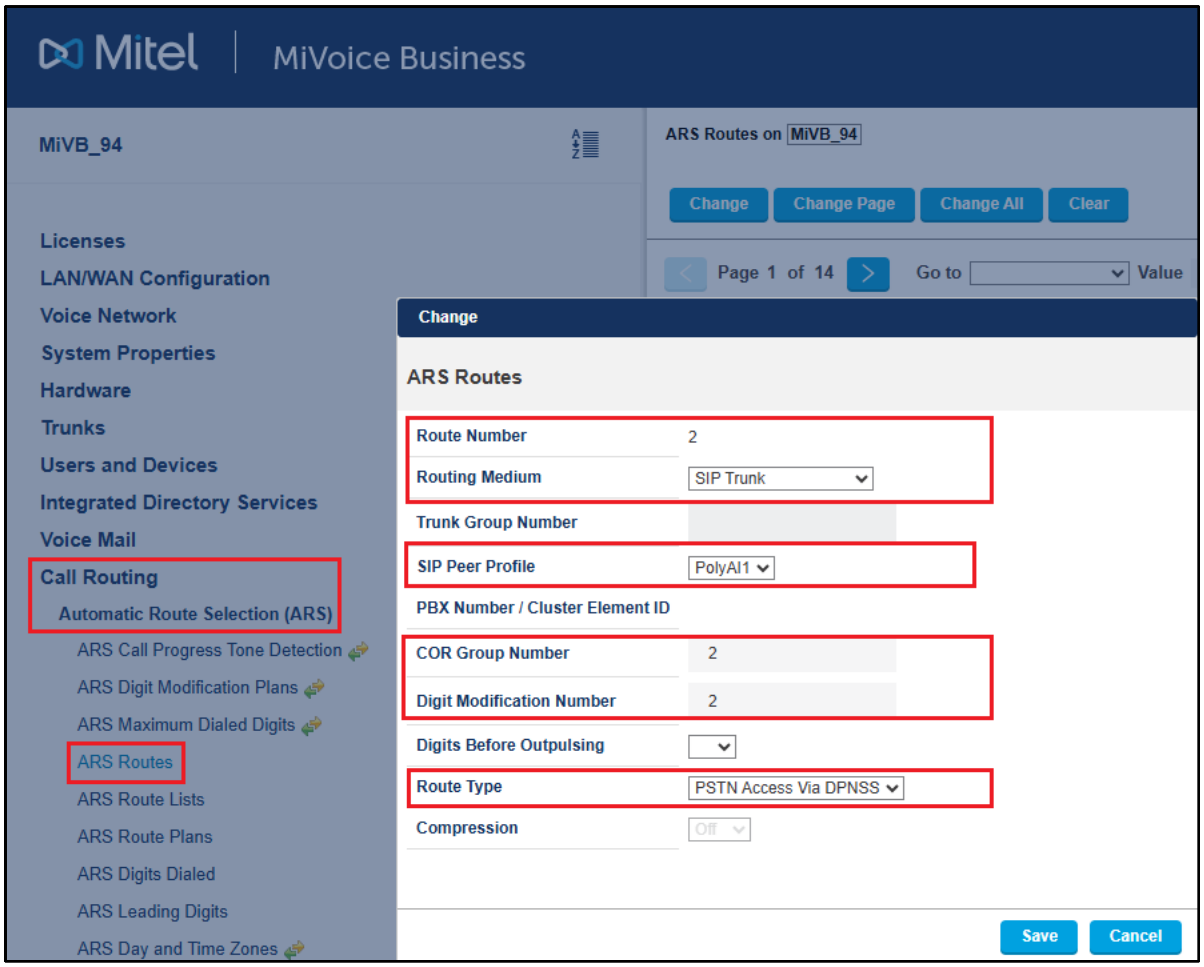


Figure 18a – SIP Trunk Route Assignment (Primary PolyAI SBC)

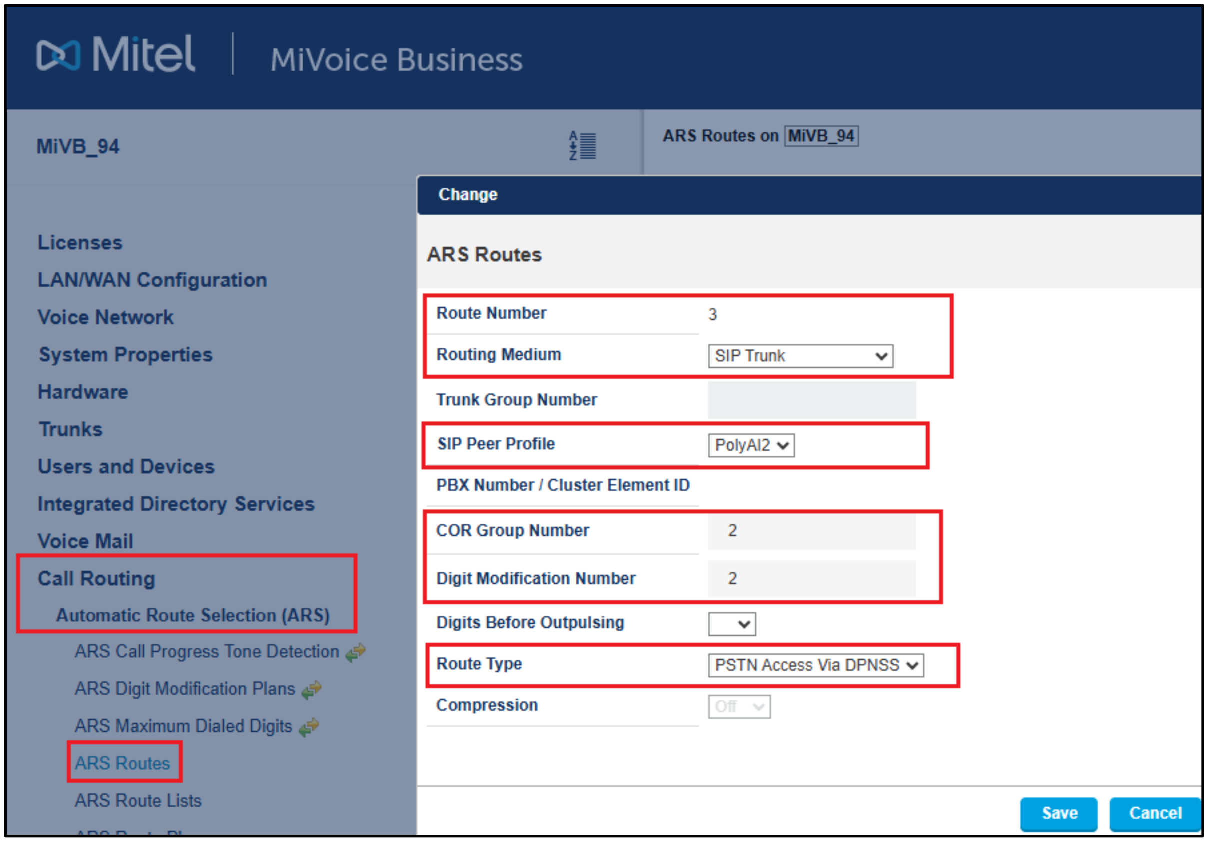


Figure 18b – SIP Trunk Route Assignment (Secondary PolyAI SBC)

ARS Route List

A Route List will need to be created that contains both PolyAI ARS Routes. All calls to PolyAI will be sent to this Route List. If the Primary Route is unavailable, the call will be routed to the Secondary Route.

- Select an unused **Route List** Number.
- **1st Choice route**: Choose the first ARS Route created in the previous step (PolyAI Primary).
- **2nd Choice route**: Choose the second ARS Route created in the previous step (PolyAI Secondary).

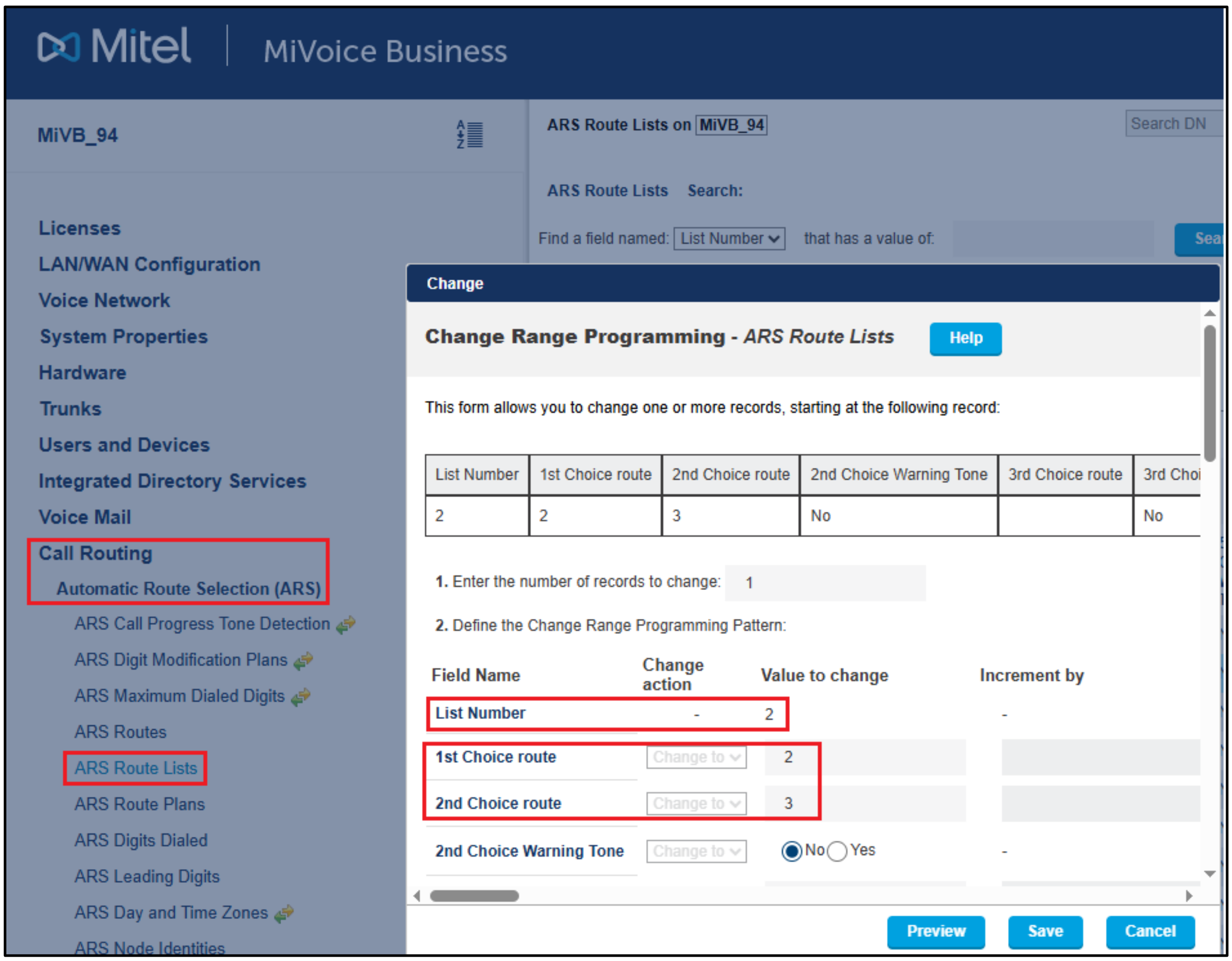


Figure 19 – ARS Route List Assignment

ARS Digits Dialed

ARS will be used to route calls to PolyAI when a specific string of digits is dialed, whether directly from an internal extension or as a PSTN Direct Inward Dial (DID) destination. For most customers, a single entry is all that is required. However, if there are multiple PolyAI Agent workflows, then multiple digit strings can be defined to access each of those separate workflows. The programming will be the same for each digit string. You will be required to provide the digit strings(s) and the desired associated Agent workflow to PolyAI during the initial coordination and setup.

- Choose the **Digits Dialed** string(s).
- **Number of Digits to Follow**: Set to 0.
- **Termination Type**: Set to List.
- **Termination Number**: Select the ARS Route List number created in the previous step.

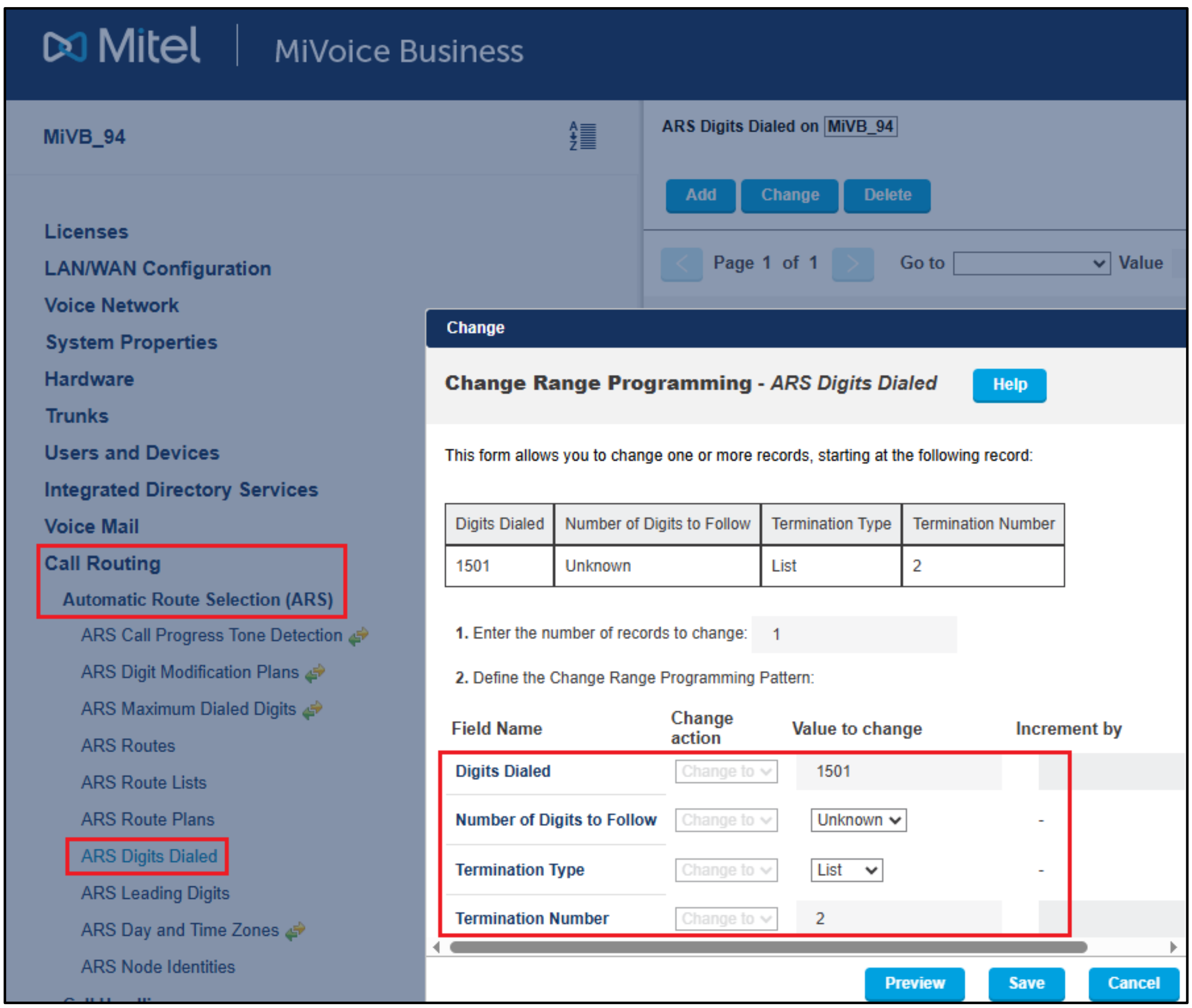


Figure 20 – ARS Digit Dialed Assignment

Direct Inward Dial (DID) Routing

External PSTN calls can be routed directly to PolyAI as DID numbers:

- Click "**Add**"
- **DID Number**: Enter the external PSTN DID number to be routed to PolyAI
- **Destination Number**: Enter the same digit string that was programmed in ARS for internal extensions to reach PolyAI

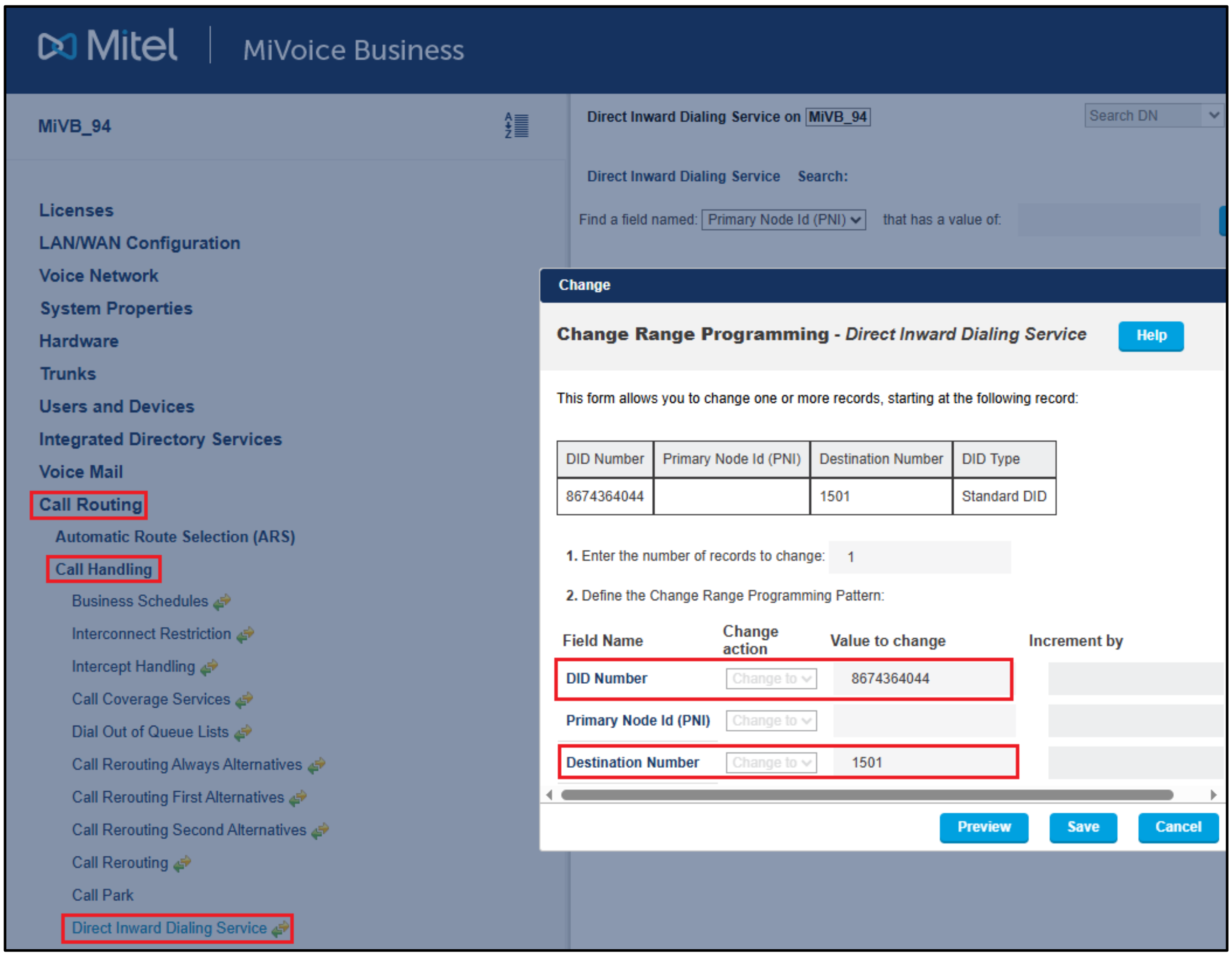


Figure 21 – Direct Inward Dial (DID)

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 22 for details)

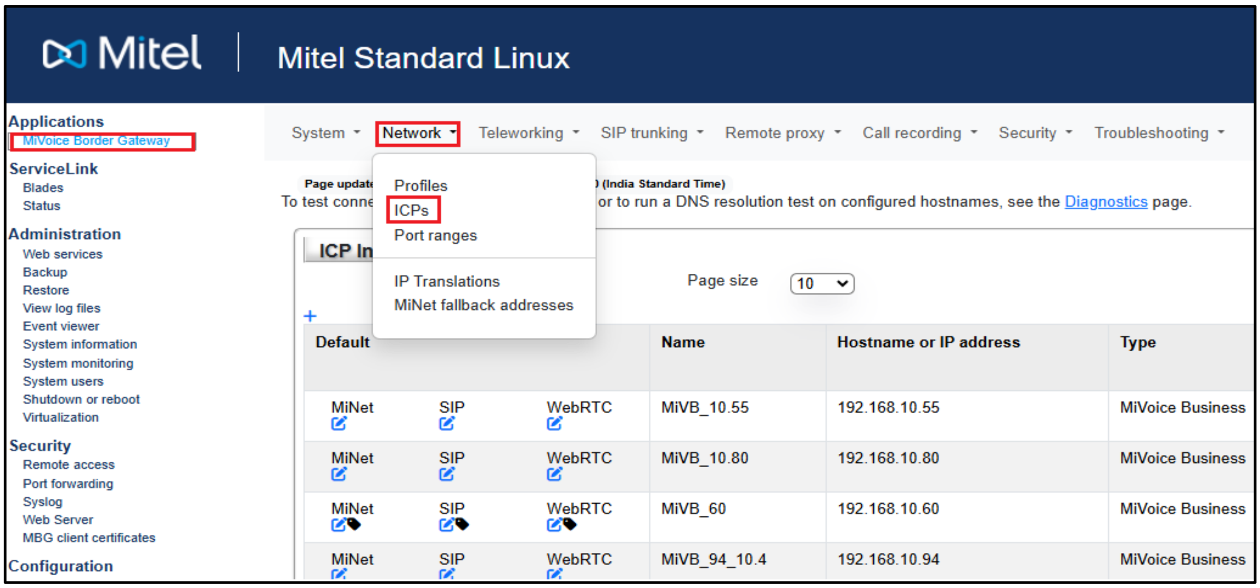


Figure 22 – MBG's Configuration page

- 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

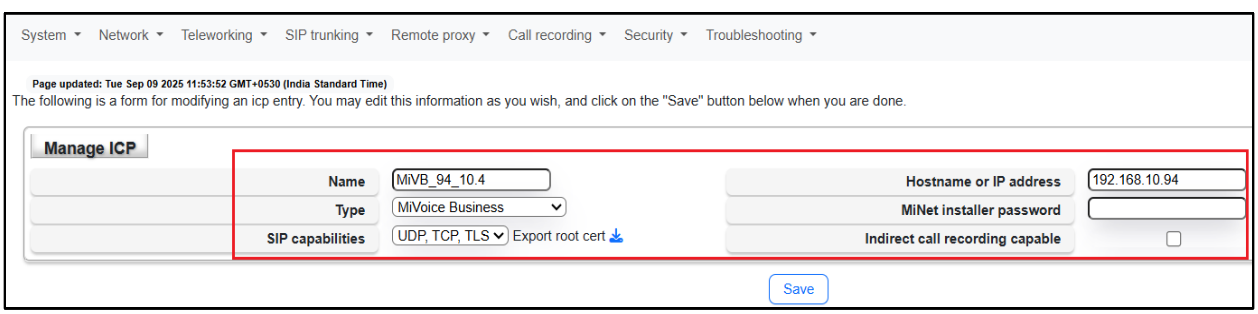


Figure 23 – ICP configuration page

- Next configure the SIP trunking by clicking on the "**SIP Trunking**" tab and selecting "**SIP Trunks**". See figure 24.

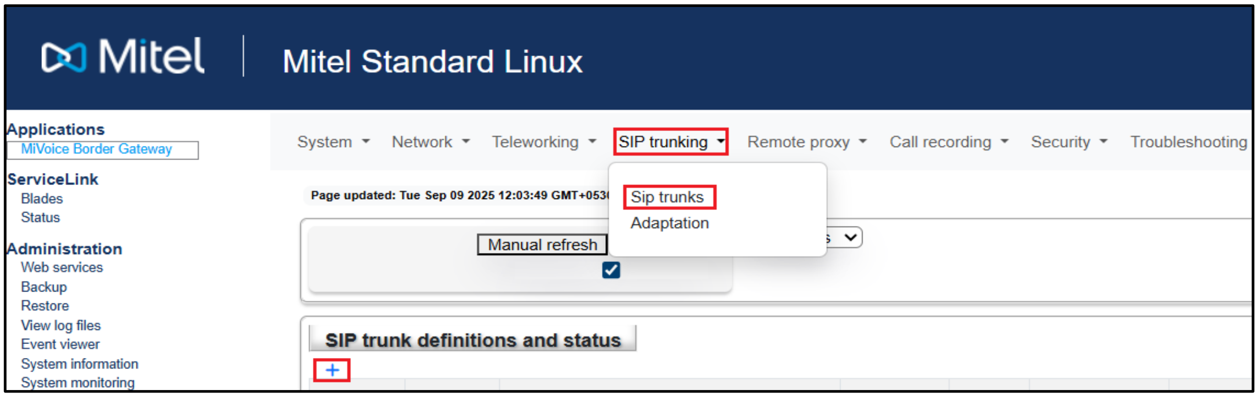


Figure 24 - MBG SIP Trunking Configuration

- The MBG will need to be programmed with both PolyAI SBCs (Primary and Secondary), as shown below. See figure 25 and 26.
- On the SIP Trunking Configuration page click on the "**+**" symbol and create SIP trunks for each PolyAI SBCs (Primary and Secondary).

Enter the SIP Trunking details as shown in figure 25 and 26:

- **Profile** > **Name**: Provide a descriptive name for the trunk.
- **Connection** > **Transport protocol**: Select UDP or TLS.
- **Connection** > **Remote trunk endpoint address**: Enter the respective IP address of the PolyAI Primary or Secondary SBCs.
- **Connection** > **Remote trunk endpoint port**: For UDP, use 5060. For TLS, use 5061.
- **Connection** > **Outgoing TLS trust profile (if using TLS):** Select "Outgoing trust profile for "PolyAI" after uploading PolyAI certificate on MBG under **Security** --> **Trust Store** as per figure 28.
- **Trunk-side RTP security** – If using TLS, Both Inbound and Outbound should be set to "**SRTP only**".
- **Icp-side RTP security** – For both UDP and TLS, set inbound and outbound to "**RTP only**." During certification testing for TLS, RTP/AVP was used between MiVB and MBG, and RTP/SAVP between MBG and PolyAI.
- You must click "**Save**" prior to performing the next step. Once the initial trunk configuration has been Saved, click "**Quick add rule**".
- Click the blue "**+**" next to "**Add rule**".
- **Header Match**: Select Request URI.
- **Rule**: Enter "*****" into this field.
- **Primary ICP**: Select the primary MiVB where the SIP trunks have been built during the previous steps.
- **Secondary ICP**: if there is a resilient MiVB in use, and PolyAI SIP trunk programming has been replicated on that MiVB as well, select the resilient MiVB. Otherwise, this field can remain blank.

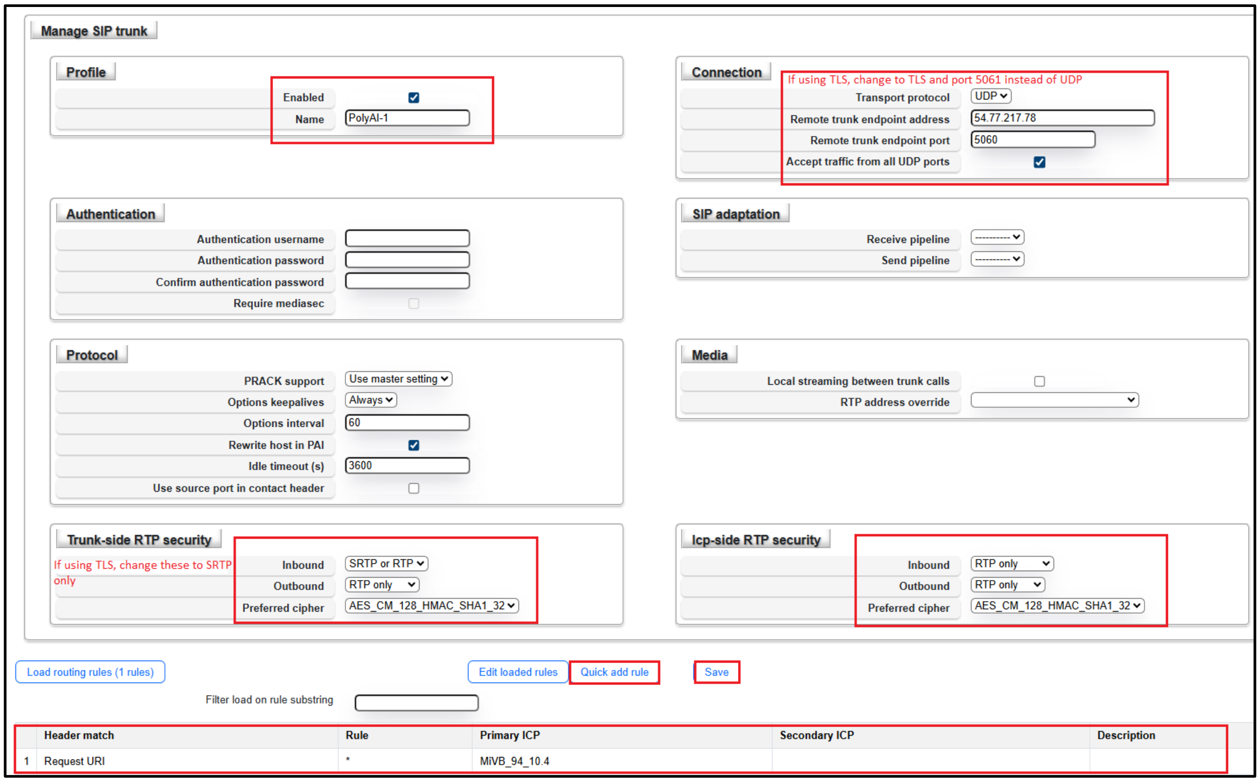


Figure 25 - MBG SIP Trunking Configuration for Primary PolyAI SBC

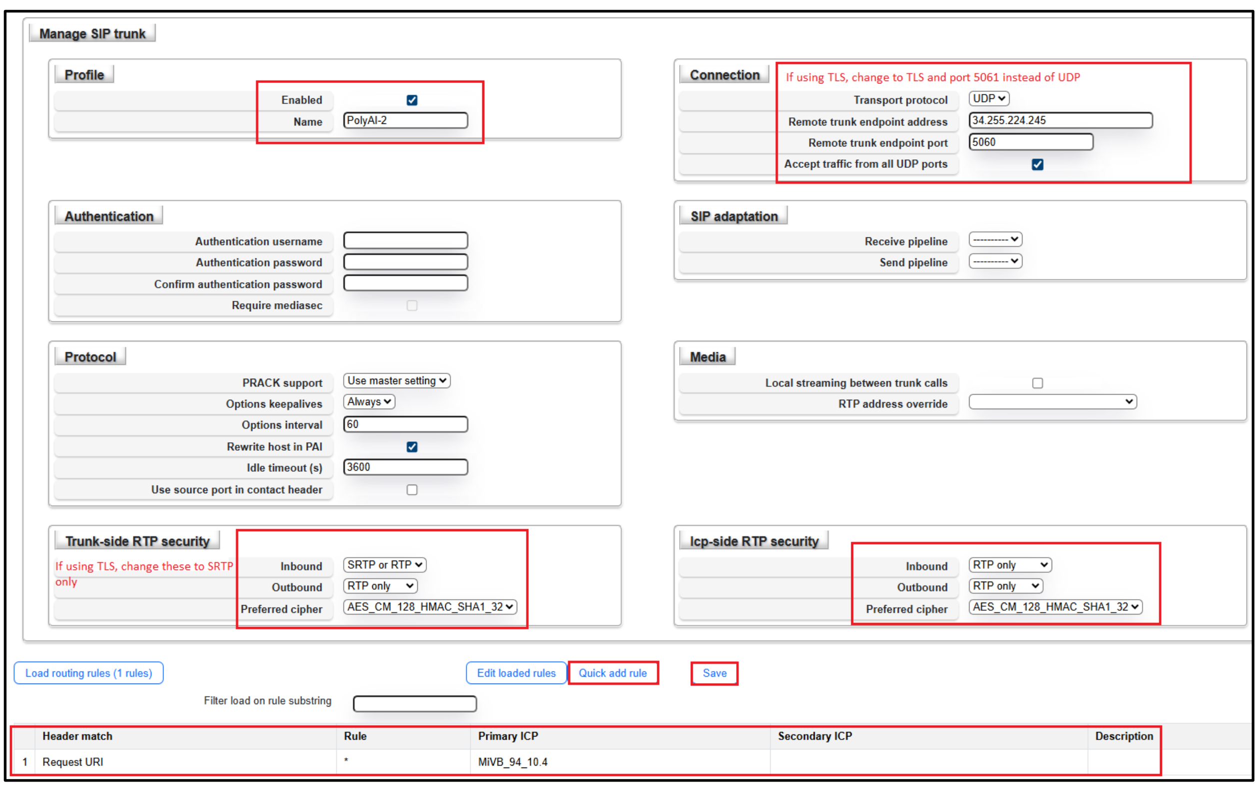


Figure 26 - MBG SIP Trunking Configuration for Secondary PolyAI SBC

- **Check status**: Click on "**SIP Trunking"** and then click on "**SIP Trunks**", see the status for both PolyAI SBCs trunks as per the figure 27.

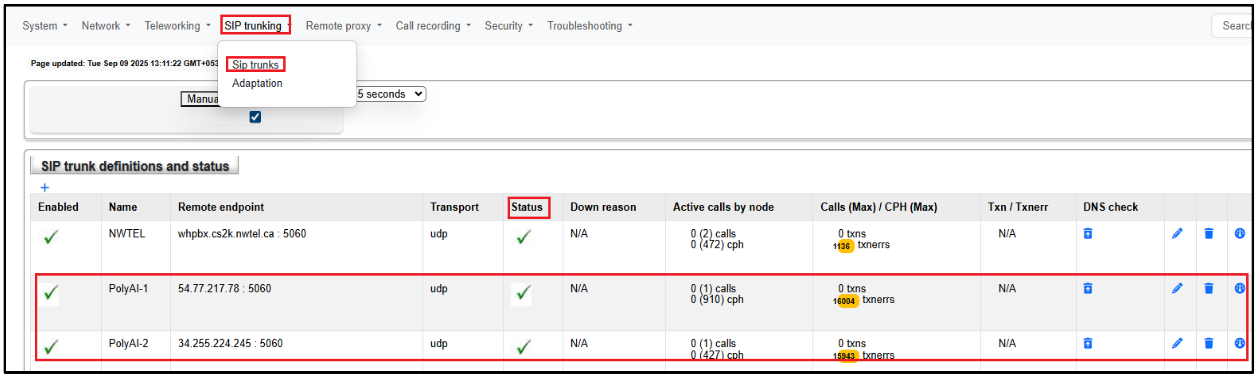


Figure 27 – SIP Trunk Status for Both PolyAI SBCs

Please upload the PolyAI certificate onto the MBG under **Security → Trust Store** for the TLS/SRTP connection.

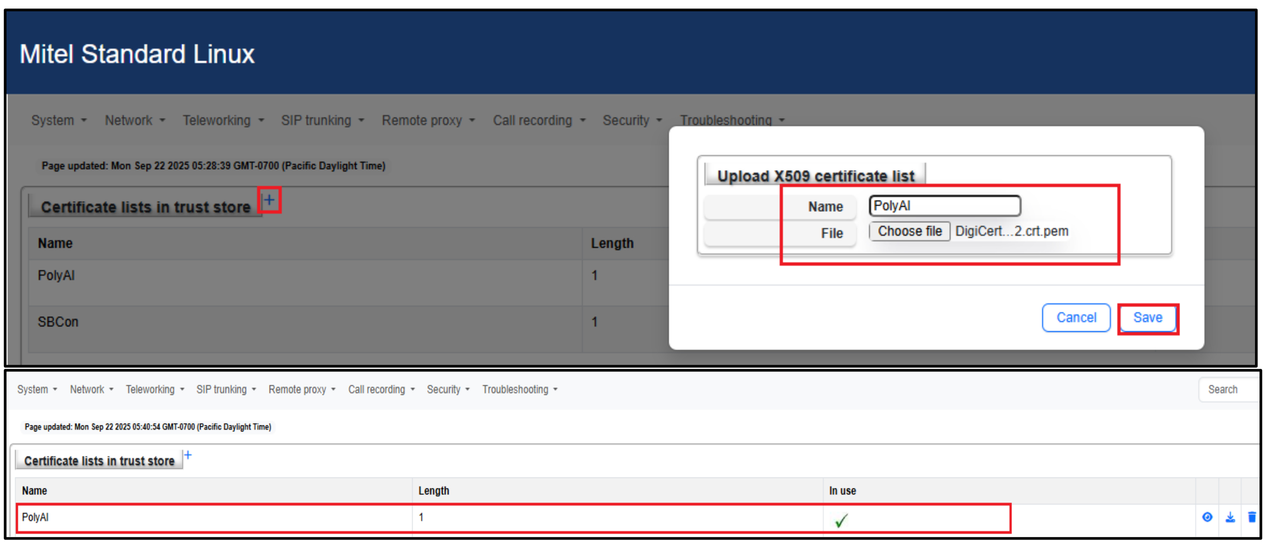


Figure 28 – Certificate Upload

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