



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

NOTICE

The information contained in this document is believed to be accurate in all respects but is not warranted by Mitel Networks™ Corporation (MITEL®). The information is subject to change without notice and should not be construed in any way as a commitment by Mitel or any of its affiliates or subsidiaries. Mitel and its affiliates and subsidiaries assume no responsibility for any errors or omissions in this document. Revisions of this document or new editions of it may be issued to incorporate such changes.

No part of this document can be reproduced or transmitted in any form or by any means - electronic or mechanical - for any purpose without written permission from Mitel Networks Corporation.

TRADEMARKS

Mitel is a trademark of Mitel Networks Corporation.

Windows and Microsoft are trademarks of Microsoft Corporation.

Other product names mentioned in this document may be trademarks of their respective companies and are hereby acknowledged.

Mitel Technical Configuration Notes – Configure MiVoice Business 10.4 for use with PolyAI using MBG.

October 2025 – HO5582

®, ™ Trademark of Mitel Networks Corporation

© Copyright 2025, Mitel Networks Corporation

All rights reserved

Table of Contents

Overview.....	1
Interop History	1
Interop Status.....	1
Software & Hardware Setup	1
Tested Features.....	2
Device Limitations and Known Issues	2
Network Topology.....	3
Configuration Notes.....	4
MiVB Configuration Notes	4
MiVoice Border Gateway Configuration Notes.....	23
Glossary.....	27

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.	<input checked="" type="checkbox"/>
DTMF	Making calls from internal/external numbers and verified DTMF scenario through IVR (RFC2833).	<input checked="" type="checkbox"/>
Hold/Retrieve	Holding/Retrieving the current IVR call.	<input checked="" type="checkbox"/>
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.	<input checked="" type="checkbox"/>
Codec	All the scenarios has been performed over G.711 codec.	<input checked="" type="checkbox"/>
TLS/SRTP	Making calls from internal/external numbers and PolyAI hangs up, transfer and DTMF calls through IVR.	<input checked="" type="checkbox"/>
Resiliency	Testing resiliency feature between MiVB, MBG and the service provider PolyAI.	<input checked="" type="checkbox"/>

- 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

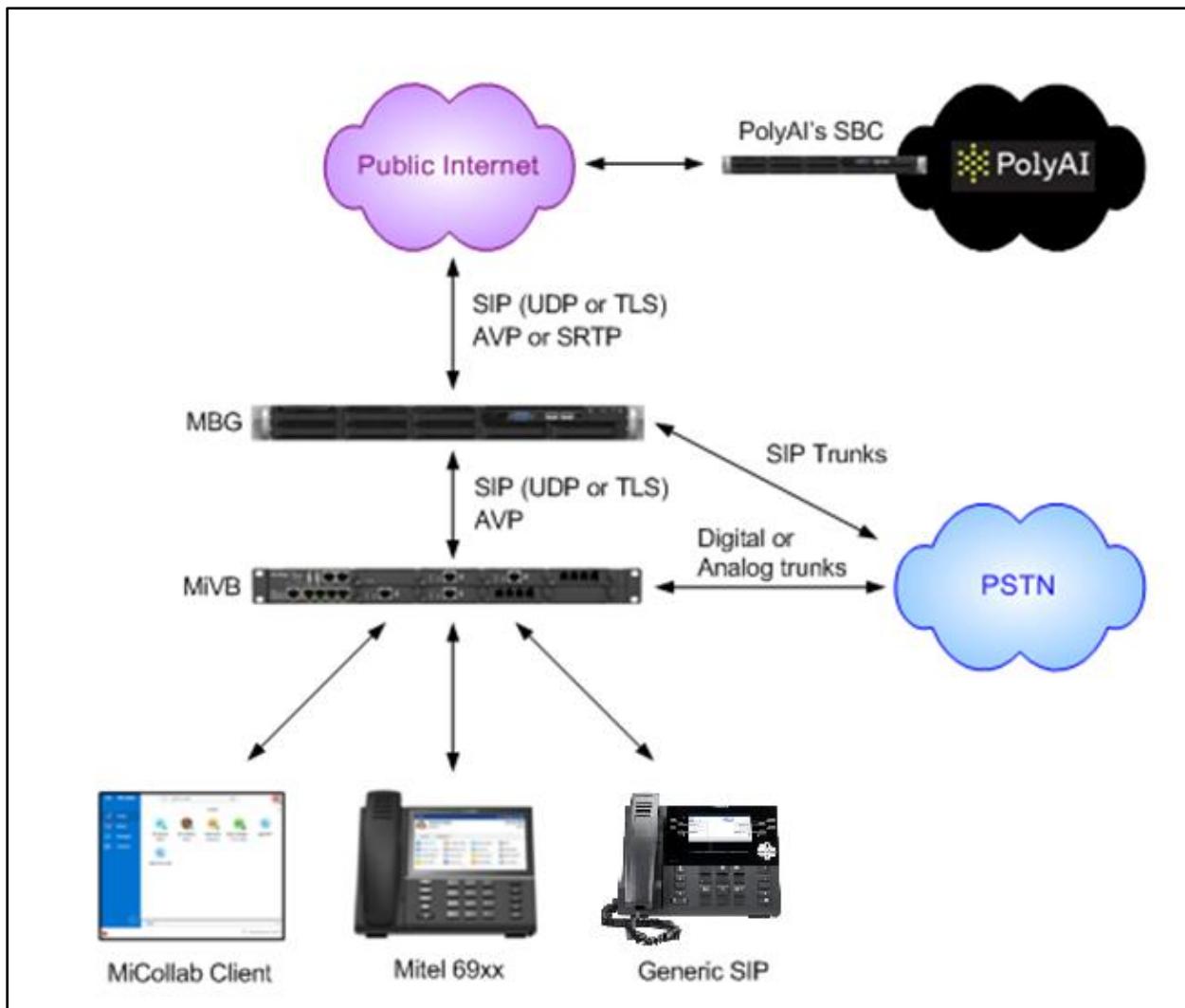


Figure 1 – Network Topology

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

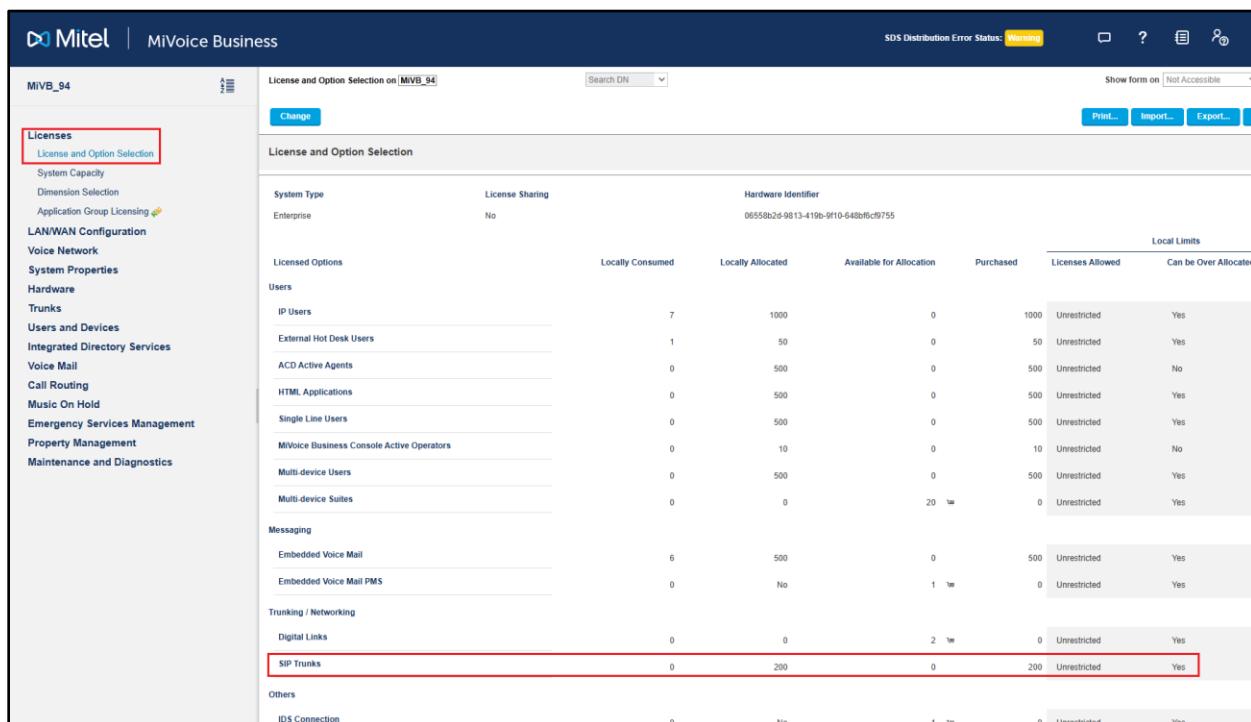
The SIP signaling connection uses UDP on Port 5060 or TLS on Port 5061.

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.



The screenshot shows the 'License and Option Selection' page for a system named 'MiVB_94'. The left sidebar lists various system configurations. The main table displays license usage across different categories. The 'SIP Trunks' row is highlighted with a red box.

System Type	License Sharing	Hardware Identifier	Local Limits			
Licensed Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Licenses Allowed	Can be Over Allocated
IP Users	7	1000	0	1000	Unrestricted	Yes
External Hot Desk Users	1	50	0	50	Unrestricted	Yes
ACD Active Agents	0	500	0	500	Unrestricted	No
HTML Applications	0	500	0	500	Unrestricted	Yes
Single Line Users	0	500	0	500	Unrestricted	Yes
MiVoice Business Console Active Operators	0	10	0	10	Unrestricted	No
Multi-device Users	0	500	0	500	Unrestricted	Yes
Multi-device Suites	0	0	20	0	Unrestricted	Yes
Embedded Voice Mail	6	500	0	500	Unrestricted	Yes
Embedded Voice Mail PMS	0	No	1	0	Unrestricted	Yes
Digital Links	0	0	2	0	Unrestricted	Yes
SIP Trunks	0	200	0	200	Unrestricted	Yes
Others	0	No	1	0	Unrestricted	Yes

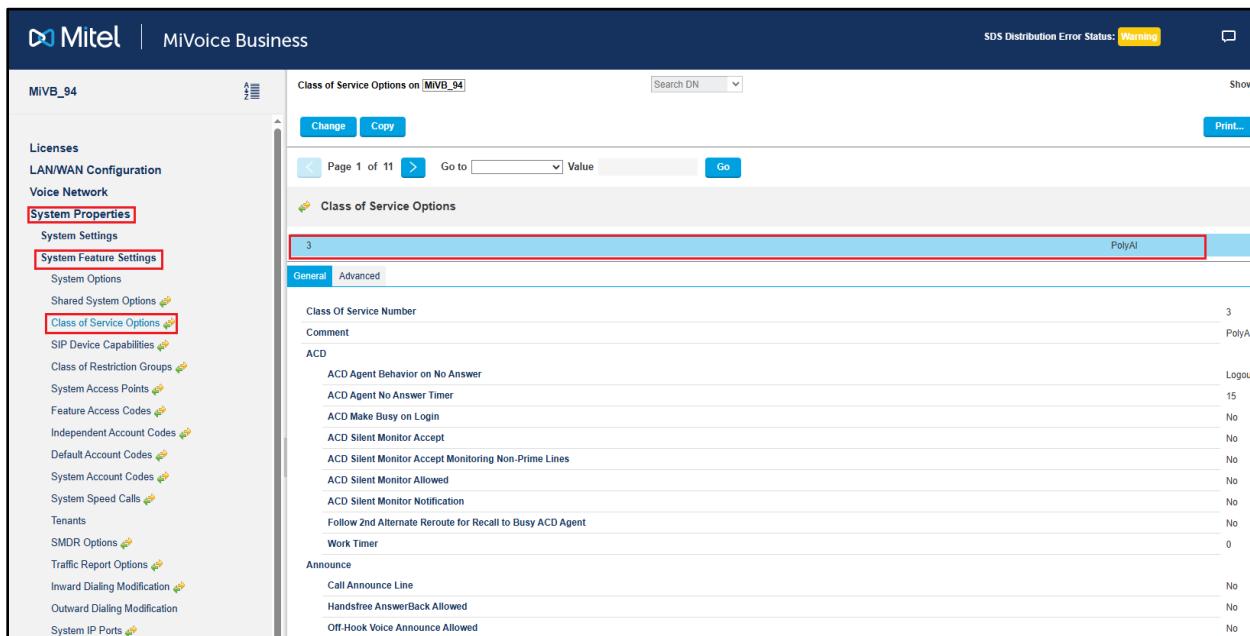
Figure 2 – License and Option Selection

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

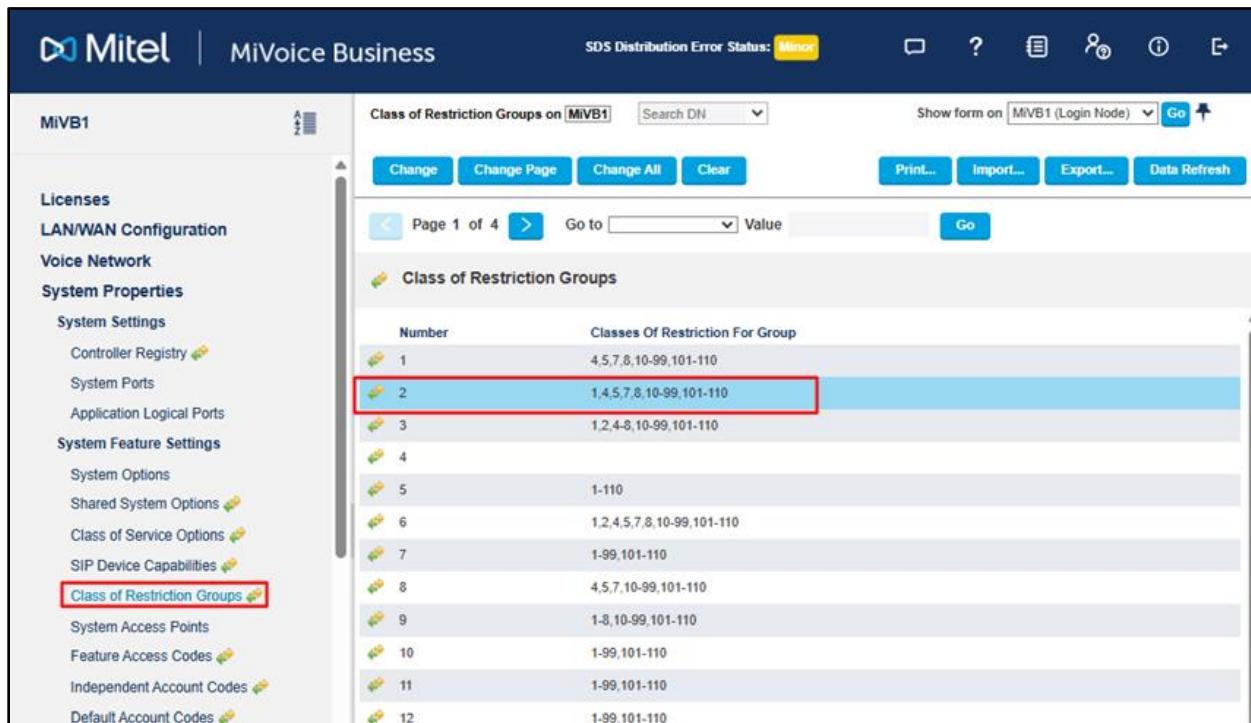


Class Of Service Number	Comment
3	PolyAI
ACD	
ACD Agent Behavior on No Answer	3
ACD Agent No Answer Timer	PolyAI
ACD Make Busy on Login	15
ACD Silent Monitor Accept	No
ACD Silent Monitor Accept Monitoring Non-Prime Lines	No
ACD Silent Monitor Allowed	No
ACD Silent Monitor Notification	No
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent	No
Work Timer	0
Announce	
Call Announce Line	No
Handsfree AnswerBack Allowed	No
Off-Hook Voice Announce Allowed	No

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:



Number	Classes Of Restriction For Group
1	4,5,7,8,10-99,101-110
2	1,4,5,7,8,10-99,101-110
3	1,2,4-8,10-99,101-110
4	
5	1-110
6	1,2,4,5,7,8,10-99,101-110
7	1-99,101-110
8	4,5,7,10-99,101-110
9	1-8,10-99,101-110
10	1-99,101-110
11	1-99,101-110
12	1-99,101-110

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.

Network Elements

Name	PolyAI-1
Type	Other
FQDN or IP Address	PolyAI's Primary SBC IP 54.77.217.78
Local	False
Version	
Zone	1
SIP Peer	<input checked="" type="checkbox"/>

SIP Peer Specific

SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	UDP
External SIP Proxy Port	5060
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	UDP
SIP Registrar Port	5060
SIP Peer Status	Auto-Detect/Normal

Save **Cancel**

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

Mitel | MiVoice Business

MiVB_94

Change

Network Elements

Name	PolyAI-2
Type	Other
FQDN or IP Address	PolyAI's Secondary SBC IP 34.255.224.245
Local	False
Version	
Zone	1
SIP Peer	<input checked="" type="checkbox"/>

SIP Peer Specific

SIP Peer Transport	UDP
SIP Peer Port	5060
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	UDP
External SIP Proxy Port	5060
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	UDP
SIP Registrar Port	5060
SIP Peer Status	Auto-Detect/Normal

Save Cancel

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).

The screenshot shows the MiVoice Business software interface. The left sidebar menu includes: Licenses, LAN/WAN Configuration, Voice Network (highlighted with a red box), Network Elements (highlighted with a red box), Cluster Elements, Analog Gateway Servers, Admin Groups, Fax Service Profiles, Fax Advanced Settings, Network Zones, Network Zone Topology, Bandwidth Management, Codec Settings, Mass Audio Notification, System Properties, and Hardware. The main content area has a title 'Change' and a sub-section 'Network Elements'. It displays a table with the following data:

Name	MBG_82
Type	Outbound Proxy
FQDN or IP Address	MBG IP Address 192.168.10.82
Local	False
Version	
Zone	1

Below this is a section titled 'Outbound Proxy Specific' with the following data:

Outbound Proxy Transport Type	UDP
Outbound Proxy Port	Change to TLS and port 5061 if using TLS instead of UDP 5060

At the bottom right are 'Save' and 'Cancel' buttons.

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.

The screenshot shows the MiVoice Business software interface. The left sidebar lists various system configurations. The 'Trunks' and 'Trunk Attributes' items are highlighted with a red box. The main 'Trunk Attributes' screen shows several configuration parameters. Some of these parameters are highlighted with a red box: 'Trunk Service Number' (set to 2), 'Class of Service' (set to 3), 'Class of Restriction' (set to 2), 'Dial In Trunks Incoming Digit Modification - Absorb' (set to 0), and 'Trunk Label' (set to PolyAI). The 'Save' and 'Cancel' buttons are visible at the bottom right of the form.

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.

Mitel | MiVoice Business

SDS Distribution Error Status: Warning

MIVB_94

Licenses
LANWAN Configuration
Voice Network
System Properties
Hardware
Trunks
Trunk Attributes
IPXNET
SIP
DID Ranges for CPN Substitution
SIP Peer Profile
SIP Peer Profile Assignment by Incoming DID
SIP Peer Profile Called Party Inward Dialing Modification
SIP Peer Profile Calling Party Inward Dialing Modification
SIP Peer Profile Called Party Outward Dialing Modification
URI/Number Translation
Users and Devices
Integrated Directory Services
Voice Mail
Call Routing
Music On Hold
Emergency Services Management
Property Management
Maintenance and Diagnostics

SIP Peer Profile on **MIVB_94** Search DN:

SIP Peer Profile

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

Basic

SIP Peer Profile Label PolyAI1
Network Element PolyAI-1
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 MBG_82
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 8a – SIP Peer Profile Assignment - Basic (Primary PolyAI SBC)

Basic

SIP Peer Profile Label PolyAI2
Network Element PolyAI-2
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 MBG_82
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 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
Alternate Destination Domain Enabled							No
Alternate Destination Domain FQDN or IP Address							No
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
Default CPN							
Default CPN Name							
CPN Restriction							No
Override From Header with Default CPN							No
Public Calling Party Number Passthrough							No
Strip PN							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
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	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
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

SIP Peer Profile							
PolyAI-1		PolyAI1		MBG_82		-	
PolyAI-2		PolyAI2		MBG_82		-	
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges
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
KPMI Transport							default
KPMI Port							0

Figure 14 – SIP Peer Profile Assignment - Key Press Event

SIP Peer Profile							
PolyAI-1		PolyAI1		MBG_82		-	
PolyAI-2		PolyAI2		MBG_82		-	
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges
Profile Information							
Index		DID Range			CPN Substitution		
							Update

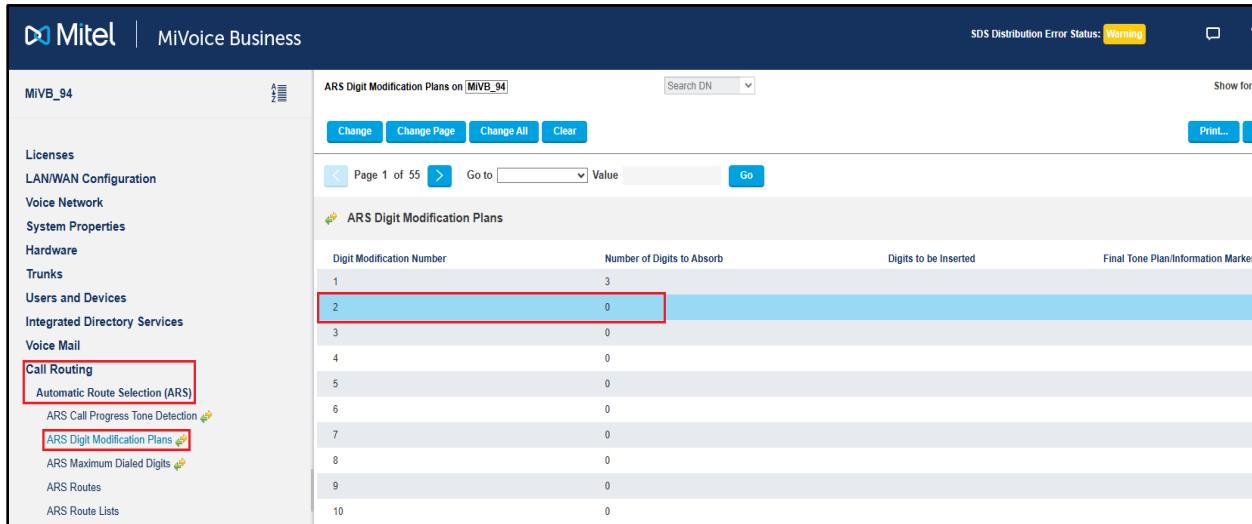
Figure 15 – SIP Peer Profile Assignment - Outgoing DID Ranges

SIP Peer Profile							
PolyAI-1		PolyAI1		MBG_82		-	
PolyAI-2		PolyAI2		MBG_82		-	
Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges
Profile Information							
Creator							
Date Created							
Created with Version							
Service Provider							
Vendor Notes							

Figure 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.



Digit Modification Number	Number of Digits to Absorb	Digits to be Inserted	Final Tone Plan/Information Marker
1	3	0	
2	0	0	
3	0	0	
4	0	0	
5	0	0	
6	0	0	
7	0	0	
8	0	0	
9	0	0	
10	0	0	

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.

MiVB_94 | MiVoice Business

Licenses
LAN/WAN Configuration
Voice Network
System Properties
Hardware
Trunks
Users and Devices
Integrated Directory Services
Voice Mail
Call Routing
Automatic Route Selection (ARS)
ARS Call Progress Tone Detection
ARS Digit Modification Plans
ARS Maximum Dialed Digits
ARS Routes
ARS Route Lists
ARS Route Plans
ARS Digits Dialed
ARS Leading Digits
ARS Day and Time Zones

ARS Routes on MiVB_94

Change Page Change All Clear

Page 1 of 14 Go to Value

ARS Routes

Route Number	2
Routing Medium	SIP Trunk
Trunk Group Number	
SIP Peer Profile	PolyAI1
PBX Number / Cluster Element ID	
COR Group Number	2
Digit Modification Number	2
Digits Before Outpulsing	
Route Type	PSTN Access Via DPNSS
Compression	Off

Save Cancel

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

Mitel | MiVoice Business

MiVB_94 | ARS Routes on MiVB_94

Licenses
LAN/WAN Configuration
Voice Network
System Properties
Hardware
Trunks
Users and Devices
Integrated Directory Services
Voice Mail
Call Routing
Automatic Route Selection (ARS)
ARS Call Progress Tone Detection
ARS Digit Modification Plans
ARS Maximum Dialed Digits
ARS Routes
ARS Route Lists
ARS Rule Lists

Change

ARS Routes

Route Number: 3
Routing Medium: SIP Trunk
Trunk Group Number:
SIP Peer Profile: PolyAI2
PBX Number / Cluster Element ID

COR Group Number: 2
Digit Modification Number: 2
Digits Before Outpulsing:
Route Type: PSTN Access Via DPNSS
Compression: Off

Save Cancel

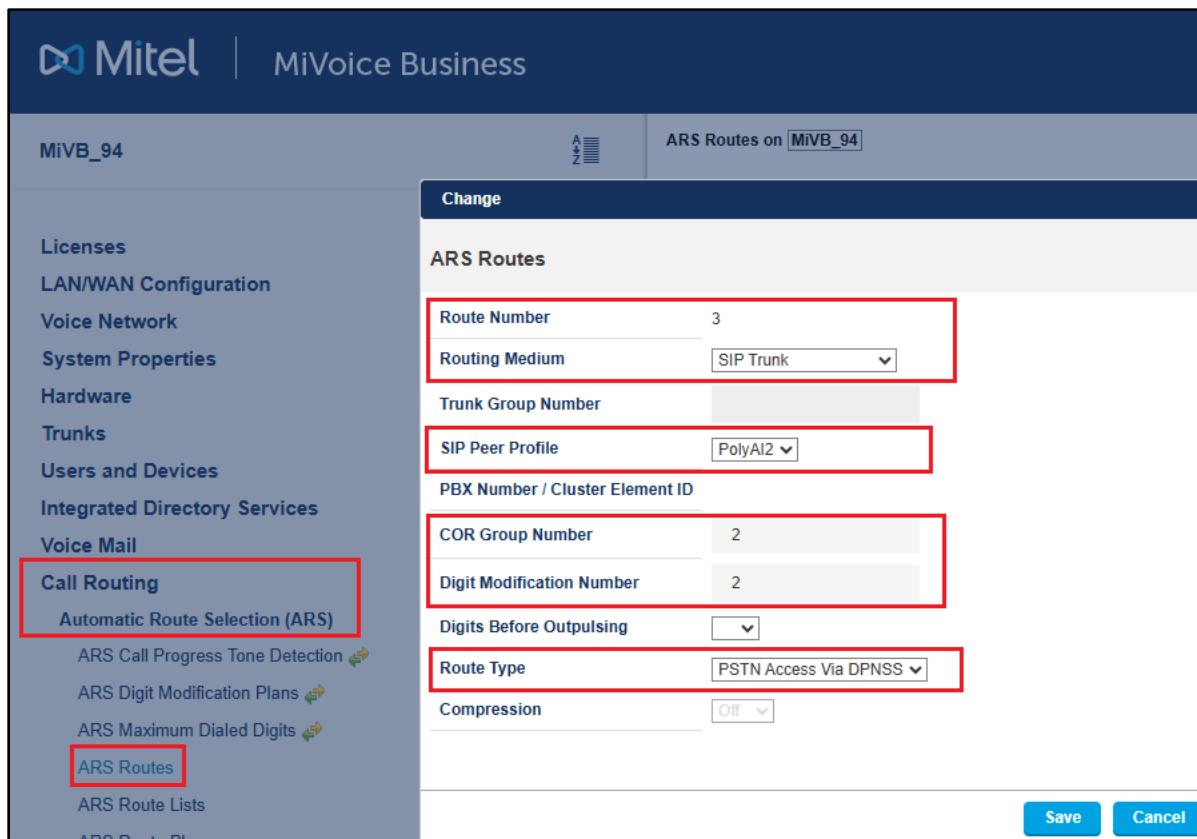


Figure 18b – SIP Trunk Route Assignment (Secondary PolyAI SBC)

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).

Mitel | MiVoice Business

MiVB_94

ARS Route Lists on MiVB_94

Search DN

Licenses

LAN/WAN Configuration

Voice Network

System Properties

Hardware

Trunks

Users and Devices

Integrated Directory Services

Voice Mail

Call Routing

Automatic Route Selection (ARS)

ARS Call Progress Tone Detection

ARS Digit Modification Plans

ARS Maximum Dialed Digits

ARS Routes

ARS Route Lists

ARS Route Plans

ARS Digits Dialed

ARS Leading Digits

ARS Day and Time Zones

ARS Node Identities

ARS Route Lists on MiVB_94

ARS Route Lists Search:

Find a field named: List Number that has a value of:

Change

Change Range Programming - ARS Route Lists Help

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

List Number	1st Choice route	2nd Choice route	2nd Choice Warning Tone	3rd Choice route	3rd Cho
2	2	3	No		No

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
List Number	-	2	-
1st Choice route	Change to	2	
2nd Choice route	Change to	3	
2nd Choice Warning Tone	Change to	No	Yes

Preview Save Cancel

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.

Mitel | MiVoice Business

MiVB_94

Licenses

LAN/WAN Configuration

Voice Network

System Properties

Hardware

Trunks

Users and Devices

Integrated Directory Services

Voice Mail

Call Routing

Automatic Route Selection (ARS)

ARS Call Progress Tone Detection

ARS Digit Modification Plans

ARS Maximum Dialed Digits

ARS Routes

ARS Route Lists

ARS Route Plans

ARS Digits Dialed

ARS Leading Digits

ARS Day and Time Zones

ARS Node Identities

ARS Digits Dialed on MiVB_94

Add Change Delete

Page 1 of 1 Go to Value

Change

Change Range Programming - ARS Digits Dialed

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

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
1501	Unknown	List	2

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
Digits Dialed	Change to	1501	
Number of Digits to Follow	Change to	Unknown	-
Termination Type	Change to	List	-
Termination Number	Change to	2	

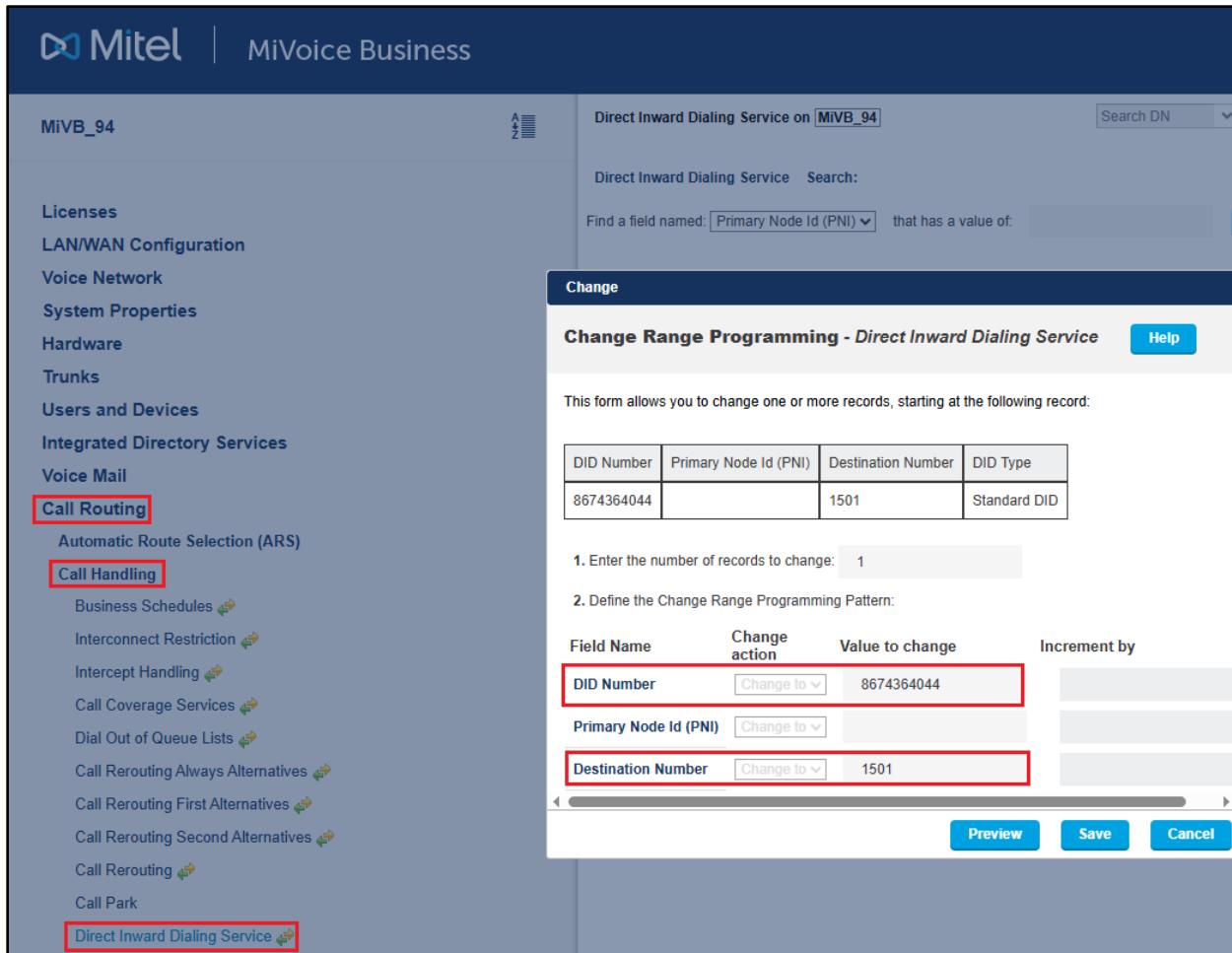
Preview Save Cancel

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



Mitel | MiVoice Business

MiVB_94

Licenses

LAN/WAN Configuration

Voice Network

System Properties

Hardware

Trunks

Users and Devices

Integrated Directory Services

Voice Mail

Call Routing

Automatic Route Selection (ARS)

Call Handling

Business Schedules

Interconnect Restriction

Intercept Handling

Call Coverage Services

Dial Out of Queue Lists

Call Rerouting Always Alternatives

Call Rerouting First Alternatives

Call Rerouting Second Alternatives

Call Rerouting

Call Park

Direct Inward Dialing Service

Direct Inward Dialing Service on MiVB_94

Search DN

Find a field named: Primary Node Id (PNI) that has a value of:

Change

Change Range Programming - Direct Inward Dialing Service

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

DID Number	Primary Node Id (PNI)	Destination Number	DID Type
8674364044		1501	Standard DID

1. Enter the number of records to change: 1

2. Define the Change Range Programming Pattern:

Field Name	Change action	Value to change	Increment by
DID Number	Change to	8674364044	
Primary Node Id (PNI)	Change to		
Destination Number	Change to	1501	

Preview Save Cancel

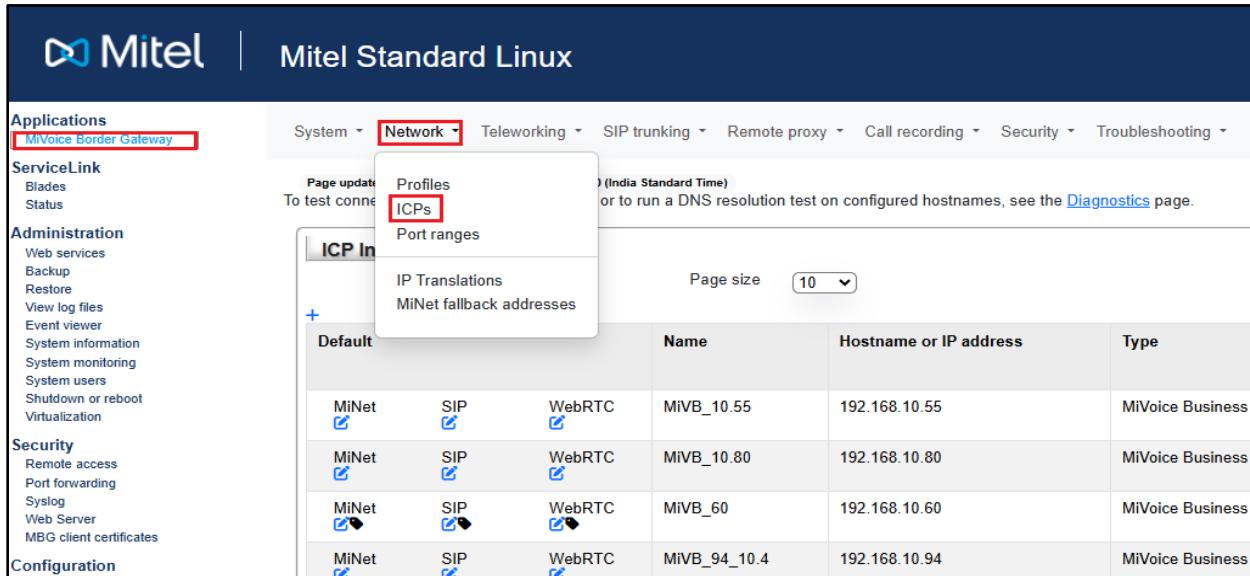
Figure 21 – Direct Inward Dial (DID)

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)

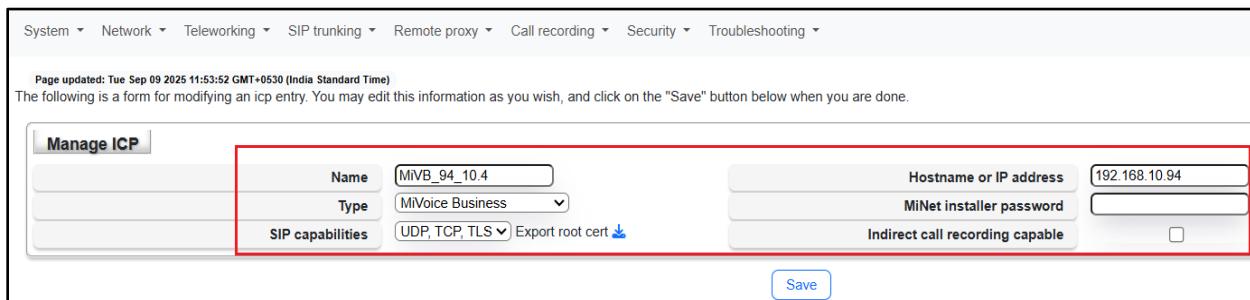


The screenshot shows the MiVoice Border Gateway configuration interface. The left sidebar has a 'Applications' section with 'MiVoice Border Gateway' highlighted. The main content area has a 'Network' tab selected. A sub-menu for 'ICPs' is open, showing options like 'Profiles', 'ICPs', and 'Port ranges'. The 'ICPs' option is also highlighted. The main table lists four ICP entries:

	Name	Hostname or IP address	Type		
MiNet	SIP	WebRTC	MiVB_10.55	192.168.10.55	MiVoice Business
MiNet	SIP	WebRTC	MiVB_10.80	192.168.10.80	MiVoice Business
MiNet	SIP	WebRTC	MiVB_60	192.168.10.60	MiVoice Business
MiNet	SIP	WebRTC	MiVB_94_10.4	192.168.10.94	MiVoice Business

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



The screenshot shows the 'Manage ICP' configuration form. The 'Name' field is set to 'MiVB_94_10.4', 'Type' is 'MiVoice Business', 'Hostname or IP address' is '192.168.10.94', and 'MiNet installer password' is empty. Under 'SIP capabilities', 'UDP, TCP, TLS' is selected and 'Export root cert' is checked. The 'Indirect call recording capable' checkbox is unchecked. A red box highlights the 'Name', 'Type', 'Hostname or IP address', and 'MiNet installer password' fields.

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**”.
- **Ice-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.

Manage SIP trunk

Profile	Connection										
Enabled <input checked="" type="checkbox"/> Name <input type="text" value="PolyAI-1"/>	If using TLS, change to TLS and port 5061 instead of UDP Transport protocol <input type="button" value="UDP"/> Remote trunk endpoint address <input type="text" value="64.77.217.78"/> Remote trunk endpoint port <input type="text" value="5060"/> Accept traffic from all UDP ports <input checked="" type="checkbox"/>										
Authentication	SIP adaptation										
Authentication username <input type="text"/> Authentication password <input type="text"/> Confirm authentication password <input type="text"/> Require mediasec <input type="checkbox"/>	Receive pipeline <input type="button"/> Send pipeline <input type="button"/>										
Protocol	Media										
PRACK support <input type="button" value="Use master setting"/> Options keepalives <input type="button" value="Always"/> Options interval <input type="text" value="60"/> Rewrite host in PAI <input checked="" type="checkbox"/> Idle timeout (s) <input type="text" value="3600"/> Use source port in contact header <input type="checkbox"/>	Local streaming between trunk calls <input type="checkbox"/> RTP address override <input type="button"/>										
Trunk-side RTP security	lcp-side RTP security										
If using TLS, change these to SRTP only Inbound <input type="button" value="SRTP or RTP"/> Outbound <input type="button" value="RTP only"/> Preferred cipher <input type="button" value="AES_CM_128_HMAC_SHA1_32"/>	Inbound <input type="button" value="RTP only"/> Outbound <input type="button" value="RTP only"/> Preferred cipher <input type="button" value="AES_CM_128_HMAC_SHA1_32"/>										
<input type="button" value="Load routing rules (1 rules)"/> <input type="button" value="Edit loaded rules"/> <input type="button" value="Quick add rule"/> <input type="button" value="Save"/> Filter load on rule substring <input type="text"/>											
<table border="1"> <thead> <tr> <th>Header match</th> <th>Rule</th> <th>Primary ICP</th> <th>Secondary ICP</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>1 Request URI</td> <td>*</td> <td>MIVB_94_10.4</td> <td></td> <td></td> </tr> </tbody> </table>		Header match	Rule	Primary ICP	Secondary ICP	Description	1 Request URI	*	MIVB_94_10.4		
Header match	Rule	Primary ICP	Secondary ICP	Description							
1 Request URI	*	MIVB_94_10.4									

Figure 25 - MBG SIP Trunking Configuration for Primary PolyAI SBC

Manage SIP trunk

Profile	Connection										
Enabled <input checked="" type="checkbox"/> Name <input type="text" value="PolyAI-2"/>	If using TLS, change to TLS and port 5061 instead of UDP Transport protocol <input type="button" value="UDP"/> Remote trunk endpoint address <input type="text" value="34.255.224.245"/> Remote trunk endpoint port <input type="text" value="5060"/> Accept traffic from all UDP ports <input checked="" type="checkbox"/>										
Authentication	SIP adaptation										
Authentication username <input type="text"/> Authentication password <input type="text"/> Confirm authentication password <input type="text"/> Require mediasec <input type="checkbox"/>	Receive pipeline <input type="button"/> Send pipeline <input type="button"/>										
Protocol	Media										
PRACK support <input type="button" value="Use master setting"/> Options keepalives <input type="button" value="Always"/> Options interval <input type="text" value="60"/> Rewrite host in PAI <input checked="" type="checkbox"/> Idle timeout (s) <input type="text" value="3600"/> Use source port in contact header <input type="checkbox"/>	Local streaming between trunk calls <input type="checkbox"/> RTP address override <input type="button"/>										
Trunk-side RTP security	lcp-side RTP security										
If using TLS, change these to SRTP only Inbound <input type="button" value="SRTP or RTP"/> Outbound <input type="button" value="RTP only"/> Preferred cipher <input type="button" value="AES_CM_128_HMAC_SHA1_32"/>	Inbound <input type="button" value="RTP only"/> Outbound <input type="button" value="RTP only"/> Preferred cipher <input type="button" value="AES_CM_128_HMAC_SHA1_32"/>										
<input type="button" value="Load routing rules (1 rules)"/> <input type="button" value="Edit loaded rules"/> <input type="button" value="Quick add rule"/> <input type="button" value="Save"/> Filter load on rule substring <input type="text"/>											
<table border="1"> <thead> <tr> <th>Header match</th> <th>Rule</th> <th>Primary ICP</th> <th>Secondary ICP</th> <th>Description</th> </tr> </thead> <tbody> <tr> <td>1 Request URI</td> <td>*</td> <td>MIVB_94_10.4</td> <td></td> <td></td> </tr> </tbody> </table>		Header match	Rule	Primary ICP	Secondary ICP	Description	1 Request URI	*	MIVB_94_10.4		
Header match	Rule	Primary ICP	Secondary ICP	Description							
1 Request URI	*	MIVB_94_10.4									

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.

SIP trunk definitions and status

Enabled	Name	Remote endpoint	Transport	Status	Down reason	Active calls by node	Calls (Max) / CPH (Max)	Txn / Txnerr	DNS check
✓	NWTEL	whpbx.cs2k.nwtel.ca : 5060	udp	✓	N/A	0 (2) calls 0 (472) cph	0 txns 1136 txnerrs	N/A	edit, edit, edit, edit
✓	PolyAI-1	54.77.217.78 : 5060	udp	✓	N/A	0 (1) calls 0 (910) cph	0 txns 1600 txnerrs	N/A	edit, edit, edit, edit
✓	PolyAI-2	34.255.224.245 : 5060	udp	✓	N/A	0 (1) calls 0 (427) cph	0 txns 1595 txnerrs	N/A	edit, edit, edit, edit

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.

Mitel Standard Linux

Certificate lists in trust store

Name	Length	In use
PolyAI	1	✓
SBCon	1	

Upload X509 certificate list

Name	PolyAI
File	Choose file DigiCert...2.crt.pem

Cancel Save

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