

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.
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### 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

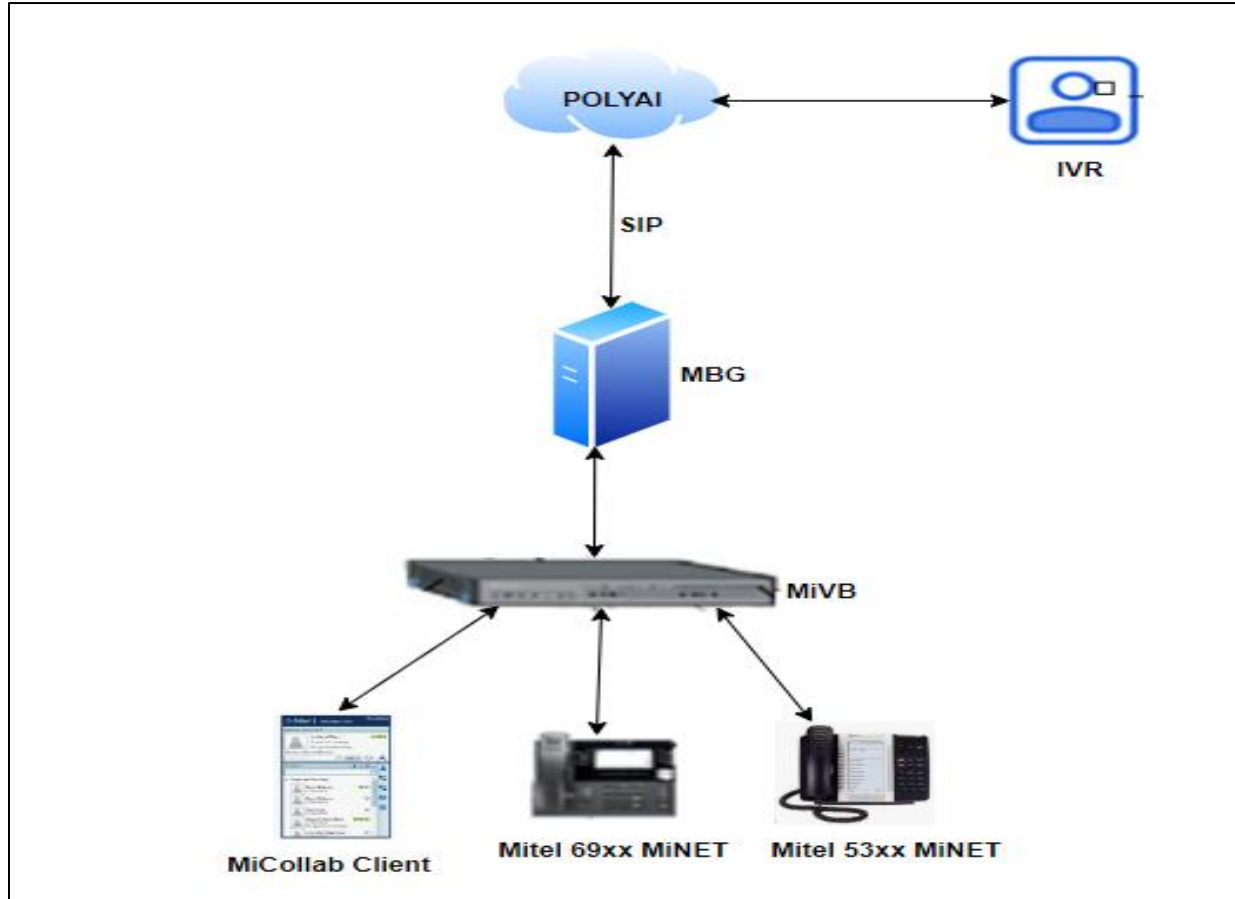


Figure 1 – Network Topology

### 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).

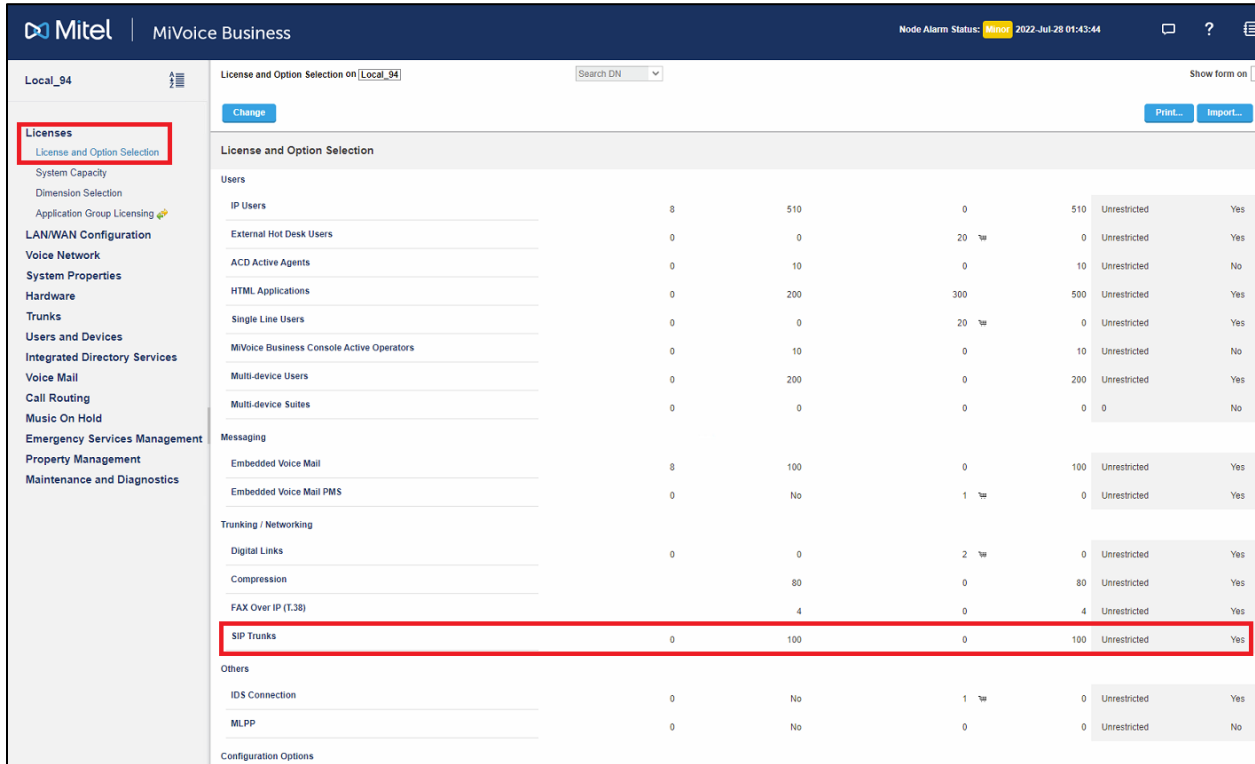
#### *Assumptions for MiVB Programming*

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.



The screenshot shows the Mitel MiVoice Business configuration interface. The left sidebar contains a menu with 'Licenses' highlighted. The main area is titled 'License and Option Selection on Local\_94'. It contains a table with various license categories and their associated values.

Category	Item	Current Value	Maximum Value	Unit	License Type	Option		
Users	IP Users	8	510	0	510	Unrestricted	Yes	
	External Hot Desk Users	0	0	20	Yes	0	Unrestricted	Yes
	ACD Active Agents	0	10	0	10	Unrestricted	No	
	HTML Applications	0	200	300	500	Unrestricted	Yes	
	Single Line Users	0	0	20	Yes	0	Unrestricted	Yes
	MiVoice Business Console Active Operators	0	10	0	10	Unrestricted	No	
	Multi-device Users	0	200	0	200	Unrestricted	Yes	
	Multi-device Suites	0	0	0	0	0	0	No
	Messaging	Embedded Voice Mail	8	100	0	100	Unrestricted	Yes
		Embedded Voice Mail PMS	0	No	1	Yes	0	Unrestricted
Trunking / Networking		Digital Links	0	0	2	Yes	0	Unrestricted
	Compression		80	0	80	Unrestricted	Yes	
	FAX Over IP (T.38)		4	0	4	Unrestricted	Yes	
	SIP Trunks	0	100	0	100	Unrestricted	Yes	
Others	IDS Connection	0	No	1	Yes	0	Unrestricted	Yes
	MLPP	0	No	0	0	Unrestricted	No	

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

Local\_94 | MiVoice Business | Node Alarm Status: Minor 2022-Jul-28 01:43:44

Local\_94 | Class of Service Options on Local\_94 | Search DN | Change | Copy | Print...

Page 1 of 11 | Go to | Value | Go

Class of Service Options

Class Of Service Number	Comment
1	
2	POLYAI

General | Advanced

Class Of Service Number: 2

Comment: POLYAI

ACD

Option	Value
ACD Agent Behavior on No Answer <td>Logout</td>	Logout
ACD Agent No Answer Timer <td>15</td>	15
ACD Make Busy on Login <td>No</td>	No
ACD Silent Monitor Accept <td>No</td>	No
ACD Silent Monitor Accept Monitoring Non-Prime Lines <td>No</td>	No
ACD Silent Monitor Allowed <td>No</td>	No
ACD Silent Monitor Notification <td>No</td>	No
Follow 2nd Alternate Reroute for Recall to Busy ACD Agent <td>No</td>	No
Work Timer <td>0</td>	0

Announce


Option	Value
Call Announce Line <td>No</td>	No
Handsfree AnswerBack Allowed <td>No</td>	No
Off-Hook Voice Announce Allowed <td>No</td>	No

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.


**Network Elements**

Name	POLYAI
Type	Other
FQDN or IP Address	34.239.17.18
Local	False
Version	
Zone	1
ARID	
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 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).

**Change**

**Network Elements**

Name	MBG62_UDP
Type	Outbound Proxy ▼
FQDN or IP Address	192.168.10.62
Local	False
Version	
Zone	1
ARID	
<b>Outbound Proxy Specific</b>	
Outbound Proxy Transport Type	UDP ▼
Outbound Proxy Port	5060

**Save** **Cancel**

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.

The screenshot shows the 'Trunk Attributes' configuration form. Several fields are highlighted with red rectangular boxes:

- Trunk Service Number:** A text field containing the value '2'.
- Direct Inward Dialing Service:** A radio button group where 'On' is selected.
- Class of Service:** A text field containing the value '2'.
- Dial In Trunks Incoming Digit Modification - Absorb:** A text field containing the value '0'.

Other visible fields and their values include:

- Release Link Trunk:** A dropdown menu set to 'No'.
- Call Recognition Service:** A dropdown menu set to 'Off'.
- Caller Based Routing Service:** A radio button group where 'Off' is selected.
- Class of Restriction:** A text field containing the value '1'.
- Baud Rate:** A dropdown menu set to '300'.
- Intercept Number:** A text field containing the value '1'.
- Non-dial In Trunks Answer Point - Day, Night 1, Night 2:** Empty text fields.
- Dial In Trunks Incoming Digit Modification - Insert:** An empty text field.
- Dial In Trunks Answer Point:** An empty text field.
- Dial In Trunks Insert Forwarding Information:** A radio button group where 'No' is selected.
- Trunk Label:** A text field containing the value 'POLYAI'.

At the bottom right of the form are two buttons: 'Save' and 'Cancel'.

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
<div>Basic   Call Routing   Calling Line ID   SDP Options   Signaling and Header Manipulation   Timers   Key Press Event   Outgoing DID Ranges   Profile Information</div>				
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-Caller-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



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

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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	<b>Profile Information</b>
Creator								
Date Created								
Created with Version								
Service Provider								
Vendor Notes								

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).

**Mitel | MiVoice Business**

Local\_94

ARS Digit Modification Plans on Local\_94

Change Change Page Change All Clear

Page 1 of 55 Go to Value Go

**ARS Digit Modification Plans**

Digit Modification Number	Number of Digits to Absorb
1	0
2	3
3	3
4	0
5	0
6	0
7	0
8	0
9	0
10	0

**Call Routing**

**Automatic Route Selection (ARS)**

ARS Call Progress Tone Detection

**ARS Digit Modification Plans**

ARS Maximum Dialed Digits

ARS Routes

ARS Route Lists

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.

**ARS Routes**

Route Number	2
Routing Medium	SIP Trunk ▼
Trunk Group Number	
SIP Peer Profile	POLYAI ▼
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	2
Digits Before Outpulsing	▼
Route Type	PSTN Access Via DPNSS ▼
Compression	Off ▼

**Save** **Cancel**

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.

**Change Range Programming - ARS Digits Dialed** [Help](#)

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

Digits Dialed	Number of Digits to Follow	Termination Type	Termination Number
111	Unknown	Route	2

1. Enter the number of records to change:

2. Define the Change Range Programming Pattern:


Field Name	Change action	Value to change	Increment by
Digits Dialed	<input type="button" value="Change to"/> ▾	<input type="text" value="111"/>	<input type="text"/>
Number of Digits to Follow	<input type="button" value="Change to"/> ▾	<input type="text" value="Unknown"/> ▾	-
Termination Type	<input type="button" value="Change to"/> ▾	<input type="text" value="Route"/> ▾	-
Termination Number	<input type="button" value="Change to"/> ▾	<input type="text" value="2"/>	<input type="text"/>

[Preview](#) [Save](#) [Cancel](#)

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.

 **Network Elements**

Name	POLYAI
Type	Other
FQDN or IP Address	34.239.17.18
Local	False
Version	
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>


SIP Peer Specific

SIP Peer Transport	TLS
SIP Peer Port	5061
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	TLS
External SIP Proxy Port	5061
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	TLS
SIP Registrar Port	5061
SIP Peer Status	Auto-Detect/Normal

Save

Cancel

Figure 20 – Network Elements – POLYAI

 **Network Elements**

Name	MBG62_TLS
Type	Outbound Proxy
FQDN or IP Address	192.168.10.62
Local	False
Version	
Zone	1
ARID	
Outbound Proxy Specific	
Outbound Proxy Transport Type	TLS
Outbound Proxy Port	5061

Save

Cancel

Figure 21 – Network Elements – MBG

SIP Peer Profile						
Network Element	SIP Peer Profile Label	Outbound Proxy Server	CPN Restriction	Trunk Service	Session Timer	Zone
Drei	Drei	MBG62_UDP	No	3	1800	1
Eckoh	Eckoh	MBG62_UDP	No	4	1800	1
POLYAI	POLYAI	MBG62_TLS	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
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_TLS							
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		*****							

Figure 22 – SIP Peer Profile

## MBG TLS trunk Configuration

Manage SIP trunk				
<b>Profile</b> Enabled <input checked="" type="checkbox"/> Name POLYAI	<b>Connection</b> Transport protocol TLS Remote trunk endpoint address 34.239.17.18 Remote trunk endpoint port 5061 Outgoing TLS trust profile No certificate validation Accept traffic from all UDP ports <input type="checkbox"/> Export root cert			
<b>Authentication</b> Authentication username Authentication password Confirm authentication password	<b>SIP adaptation</b> Receive pipeline Send pipeline			
<b>Protocol</b> PRACK support Use master setting Options keepalives Always Options interval 60 Rewrite host in PAI <input checked="" type="checkbox"/> Idle timeout (s) 3600 Use source port in contact header <input type="checkbox"/>	<b>Media</b> Local streaming between trunk calls <input type="checkbox"/> RTP address override			
<b>Trunk-side RTP security</b> Inbound SRTP or RTP Outbound SRTP only Preferred cipher AES_CM_128_HMAC_SHA1_32	<b>ICP-side RTP security</b> Inbound SRTP or RTP Outbound RTP only Preferred cipher AES_CM_128_HMAC_SHA1_32			
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				
Header match	Rule	Primary ICP	Secondary ICP	Description
1 Request URI	*	MVB_94		

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)

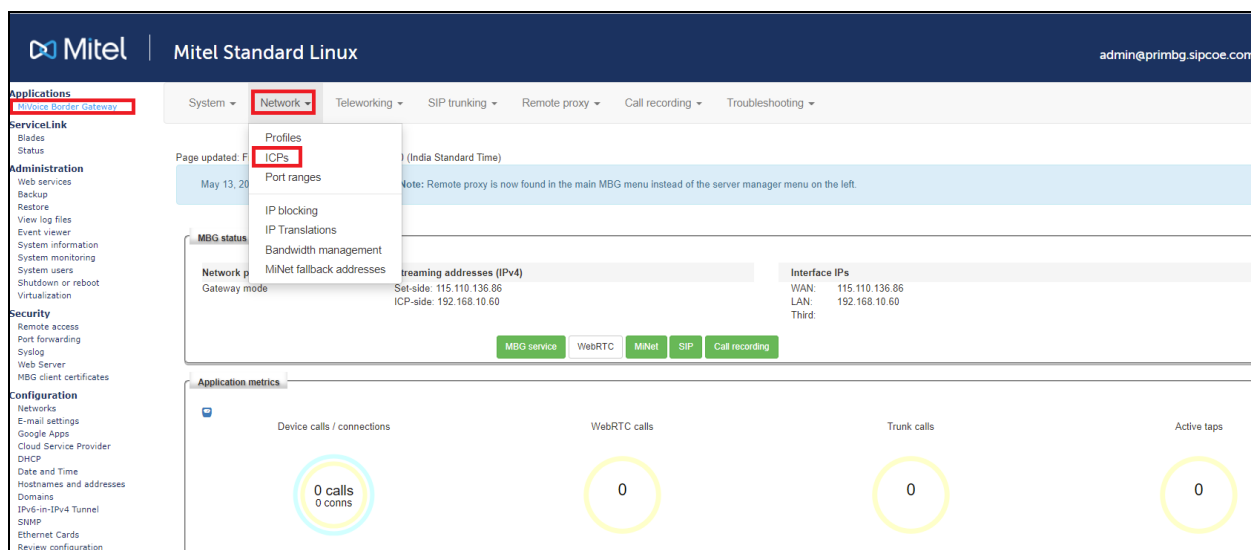


Figure 24 – MBG's Configuration page

- 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.



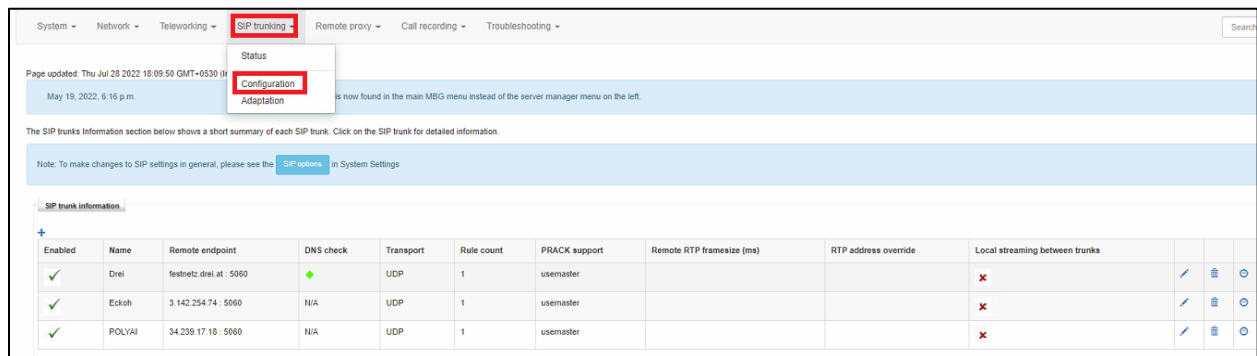


Figure 26 - MBG SIP Trunking Configuration

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.

Manage SIP trunk

**Profile**  
 Enabled ☒  
 Name POLYAI

**Connection**  
 Transport protocol UDP  
 Remote trunk endpoint address 34.239.17.18  
 Remote trunk endpoint port 5060  
 Accept traffic from all UDP ports ☐

**Authentication**  
 Authentication username  
 Authentication password  
 Confirm authentication password

**SIP adaptation**  
 Receive pipeline  
 Send pipeline

**Protocol**  
 PRACK support Use master setting  
 Options keepalives Always  
 Options interval 60  
 Rewrite host in PAI ☒  
 Idle timeout (s) 3000  
 Use source port in contact header ☐

**Media**  
 Local streaming between trunk calls ☐  
 RTP address override

**Trunk-side RTP security**  
 Inbound SRTP or RTP  
 Outbound RTP only  
 Preferred cipher AES\_CM\_128\_HMAC\_SHA1\_32

**ICP-side RTP security**  
 Inbound RTP only  
 Outbound RTP only  
 Preferred cipher AES\_CM\_128\_HMAC\_SHA1\_32

Load routing rules (1 rules) Edit loaded rules Quick add rule Save

Filter load on rule substring

Header match	Rule	Primary ICP	Secondary ICP	Description
1 Request URI	*	MVB_94		

Figure 27 - MBG SIP Trunking Configuration

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

POLYAI

**Status** ☒

**Reason**

Calls in progress / Max 0 / 2

Calls per hour / Max 0 / 719

[Reset metrics](#)

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