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# MiVB - Configure MiVoice Business 9.4 SP1 and MICC-B 9.4 for use with AudioCodes VAICC Platform

**Description:** This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB and MICCB to connect to AudioCodes VAICC platform.

**Environment:** MiVoice Business 9.4 SP1 (9.4.1.18), MiVoice Border Gateway 11.4.0.247, Mitel 69xx MiNET 01.08.00.018, MICCB 9.4.1.0

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Mitel Technical Configuration Notes – Configure MiVoice Business and MICCB for use with AudioCodes VAICC Platform

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## Overview


This document provides a reference to Mitel Authorized Solutions providers for configuring the Mitel MiVB to connect to AudioCodes VAICC Platform. 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	August, 2020	Interop with Mitel MiVB 9.1 and MICC-B 9.2 and AudioCodes Google Dialog flow Gateway.
2	December, 2022	Interop with Mitel MiVB 9.4 SP1 and MICC-B 9.4 for use with AudioCodes VAICC Platform.

## Interop Status

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

	The most common certification, which means AudioCodes VAICC Platform 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 AudioCodes VAICC Platform and the MiVB/MICC-B 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	9.4 SP1 (9.4.1.18)	NA
Mitel	MiVoice Border Gateway	11.4.0.247	NA
Mitel	MiCCB	9.4.1.0	NA
Mitel	69XX MiNET	01.08.00.018	NA
AudioCodes	VAICC	v.7.40BY.260.014	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	Transferring the call to AudioCodes VAICC Platform and AudioCodes VAICC Platform transferring the call back to MiVB	<input checked="" type="checkbox"/>
Codec	G711 codec	<input checked="" type="checkbox"/>
TLS/SRTP	Transferring the call to AudioCodes VAICC Platform and AudioCodes VAICC Platform transferring the call back to MiVB	<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 AudioCodes VAICC Platform is connected to the MiVB.

Feature	Problem Description
Media Negotiation	<p>In case of TLS/SRTP testing, we have used RTP towards MiVB/MiCCB from the MBG, and towards AudioCodes VAICC platform used as SRTP from the MBG.</p> <p><b>Recommendation:</b> Please contact Mitel Support for more information on this.</p>

## Network Topology

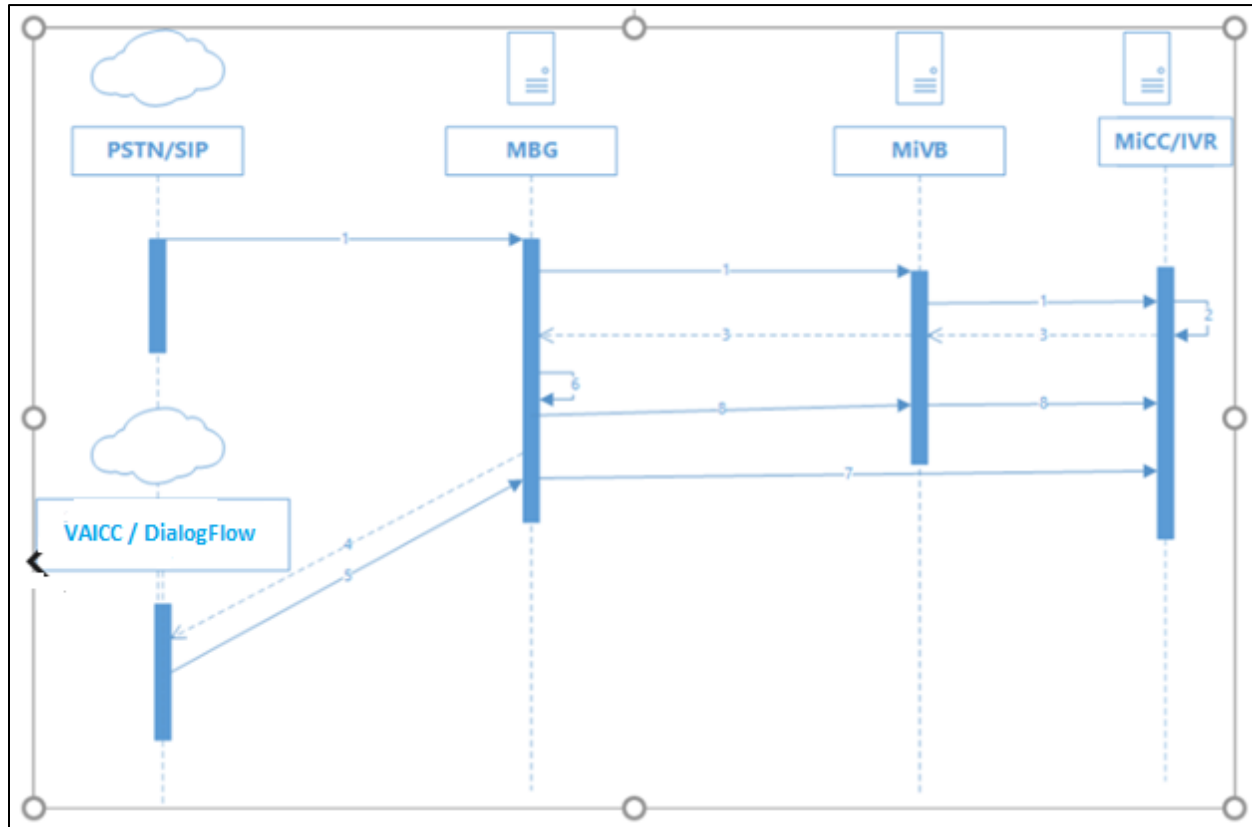


Figure 1 – Network Topology

*Scenario:*

- 1) When PSTN calls and the call lands on IVR port.
- 2) IVR Workflow/Subroutine calls REST API <http://192.168.10.134/DialogFlowDataService/PhoneNumber/{ANI}>. This is a PUT request. This will return a number that the IVR will transfer to and will store the CLI in the DialogFlowDataService DB against the phone number used to transfer.
- 3) IVR Workflow/Subroutine transfers the call to returned number.
- 4) Call is processed through MiVB/MBG. It is sent via SIP trunk to the AudioCodes VAICC platform, and it is processed by the assigned agent in Dialogflow.
- 5) If the call cannot be completed entirely by the agent, then the final intent will have a Custom Payload configured. This will send the data to a SIP endpoint and attach any data gathered as parameters within the agent. The call is sent back to the MiCC IVR as a SIP REFER. The data captured is appended to the REFER-TO message as query parameters e.g., sip: <3006@192.168.10.96?param1=val1&param2=val2>. The number being sent back to must be an endpoint on the MiVB that is processed by MiCC IVR ports, so that the data returned can be processed.
- 6) The MBG SIP trunk configured for the AudioCodes VAICC platform, has a SIP adaptation receive pipeline configured. This is a Lua plugin that checks for REFER messages, takes the query parameters, and writes these in a JSON format to a file on the MBG with the FROM address as the title + .json. This file is written to the /home/refeto folder.
- 7) There is a systemd process running a Linux executable called file watcher. The file watcher process monitors the folder, it opens the file on creation, extracts the json data, and POSTs it to the REST API on the MiCC server <http://192.168.10.134/DialogFlowDataService/referto>

The IVR gets the call that is transferred back from AudioCodes VAICC platform, the Workflow/Subroutine calls a GET request to the REST API endpoint <http://192.168.10.134/DialogFlowDataService/PhoneNumber/{ANI}>, the ANI in this case will be the number that was used to transfer to in step 1). This Endpoint returns the data that was captured by the Dialogflow agent and sets this data as variables that are configured to send back to the agent desktop

## 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 AudioCodes VAICC Platform MiVB programming was configured in our test environment.

*Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.*

### MiVB Configuration Notes

The following steps show how to program a MiVB to interconnect with AudioCodes VAICC Platform.

#### *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 the MiVB documentation on how the Import functionality is used.

#### *Network Requirements*

- There must be adequate bandwidth to support the voice over IP. 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 AudioCodes VAICC Platform. 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 s, applications, and SIP trunking devices.

The screenshot shows the 'License and Option Selection' form for 'Local\_96'. The left sidebar contains a navigation menu with 'Licenses' highlighted. The main content area displays a table of licenses. The 'SIP Trunks' row is highlighted with a red box.

License Options	Locally Consumed	Locally Allocated	Available for Allocation	Purchased	Licenses Allowed	Can be Over Allocated
Users						
IP Users	18	510	0	510	Unrestricted	Yes
External Hot Desk Users	0	0	20	0	Unrestricted	Yes
ACD Active Agents	3	10	0	10	Unrestricted	No
HTML Applications	0	500	0	500	Unrestricted	Yes
Single Line Users	0	0	20	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	No
Messaging						
Embedded Voice Mail	15	100	0	100	Unrestricted	Yes
Embedded Voice Mail PMS	0	No	1	0	Unrestricted	Yes
Trunking / Networking						
Digital Links	0	0	2	0	Unrestricted	Yes
Compression		80	0	80	Unrestricted	Yes
FAX Over IP (T.38)		4	0	4	Unrestricted	Yes
<b>SIP Trunks</b>	<b>0</b>	<b>100</b>	<b>0</b>	<b>100</b>	<b>Unrestricted</b>	<b>Yes</b>
Others						
IDS Connection	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.

The screenshot displays the Mitel MiVoice Business web interface. The left sidebar contains a navigation menu with categories: Licenses, LAN/WAN Configuration, Voice Network, System Properties (highlighted), System Settings, System Feature Settings (highlighted), System Options, Shared System Options, Class of Service Options (highlighted), SIP Device Capabilities, Class of Restriction Groups, System Access Points, Feature Access Codes, Independent Account Codes, Default Account Codes, System Account Codes, System Speed Calls, Tenants, SMDR Options, Traffic Report Options, Inward Dialing Modification, Outward Dialing Modification, System IP Ports, Location Based Numbers, System Administration, Hardware, Trunks, Users and Devices, Integrated Directory Services, Voice Mail, and Call Routing.

The main content area is titled 'Class of Service Options on Local\_96'. It includes a search bar, 'Change' and 'Copy' buttons, and a 'Print...' button. Below this is a table of Class of Service Options. The table has two columns: 'Class Of Service Number' and 'Value'. The first row shows '9' and 'VAICC'. The table is divided into sections: General, Advanced, ACD, Announce, Busy Override, and Call Control Timer.

Class Of Service Number	Value
9	VAICC
Comment	VAICC
ACD	
ACD Agent Behavior on No Answer	Logout
ACD Agent No Answer Timer	15
ACD Make Busy on Login	No
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 Route 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
Busy Override	
Busy Override Security	No
Disable Executive Busy Override Tone	No
Executive Busy Override	No
Call Control Timer	
Busy Tone Timer	30
Dialing Conflict Timer	3
First Digit Timer	15
Inter Digit Timer	10

Figure 3 – Class of Service

## Network Element Assignment

Create a network element for AudioCodes VAICC Platform. In this example, the soft switch is reachable by an IP Address and is defined as “AudioCodes VAICC Platform” 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 AudioCodes VAICC Platform 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 for the UDP trunk and the TLS and port to 5061 for the TLS trunk (see the below screen shots for reference).

The screenshot shows a web-based configuration form titled "Network Elements". The form contains the following fields and values:

Field	Value
Name	VAICC
Type	Other
FQDN or IP Address	1.1.1.1
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	Always Active

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

Figure 4 – Network Element Assignment for UDP


**Network Elements**

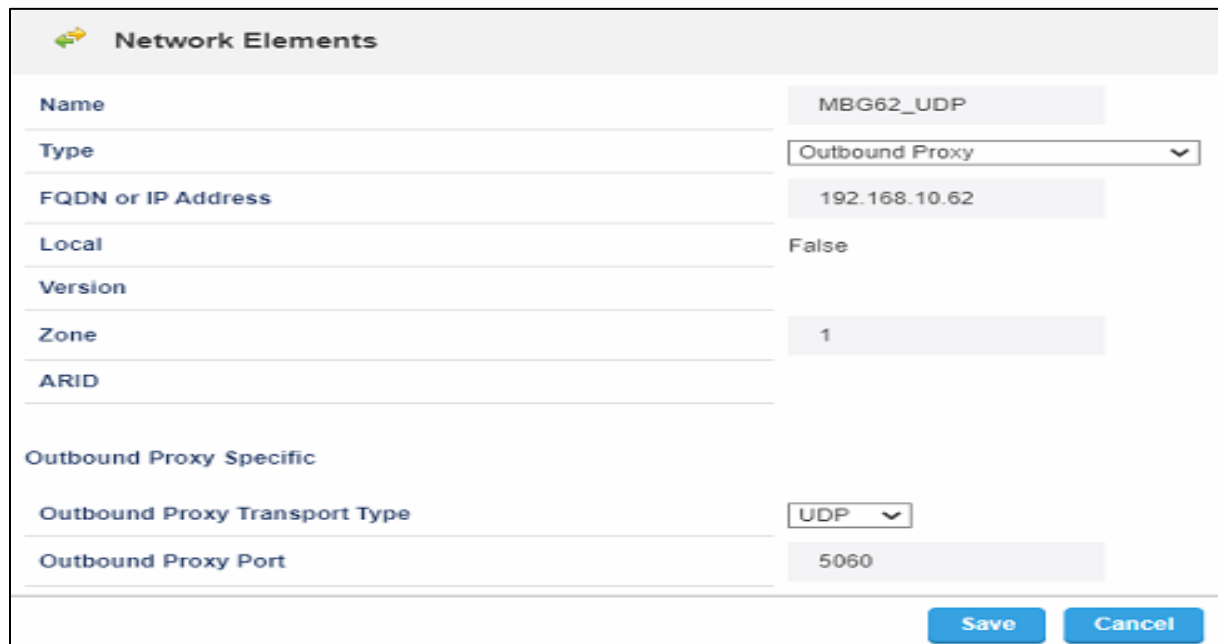
Name	VAICCTLS
Type	Other
FQDN or IP Address	2.2.2.2
Local	False
Version	
Zone	1
ARID	
SIP Peer	<input checked="" type="checkbox"/>
SIP Peer Specific	
SIP Peer Transport	TLS
SIP Peer Port	5061
External SIP Proxy FQDN or IP Address	
External SIP Proxy Transport	TLS
External SIP Proxy Port	5061
SIP Registrar FQDN or IP Address	
SIP Registrar Transport	TLS
SIP Registrar Port	5061
SIP Peer Status	Always Active

Save
Cancel

Figure 5 – Network Element Assignment for TLS

### Network Element Assignment (Proxy)

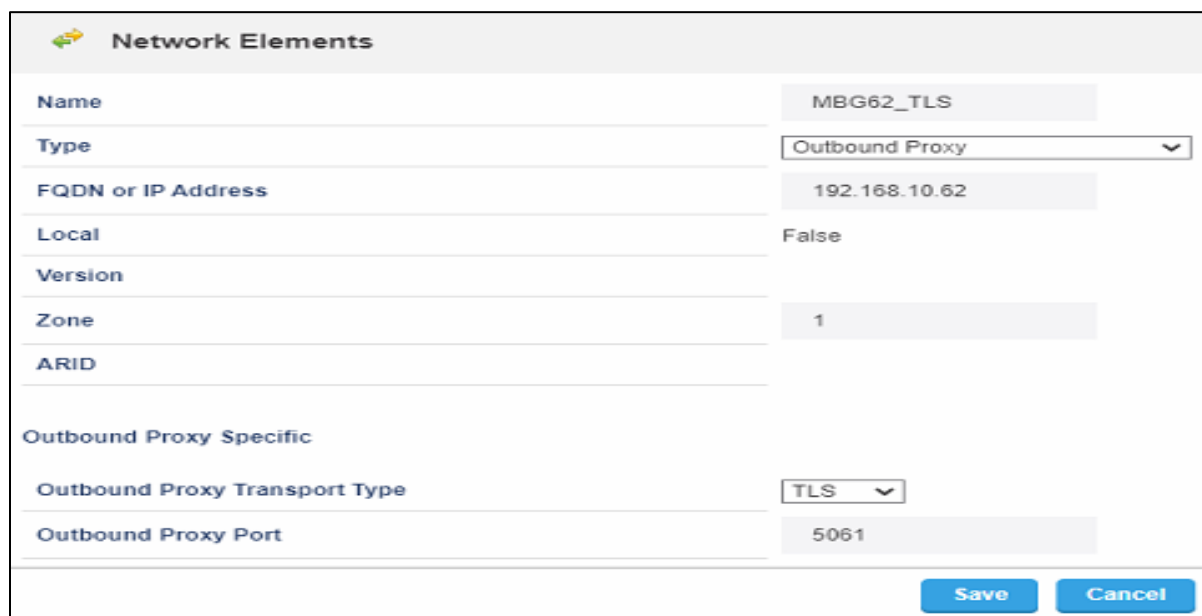
In addition, depending on your configuration, a Proxy may need to be configured to route SIP data to the service provider. If you have a Proxy server installed in your network, the MiVB will require knowledge of this by programming the Proxy as a network element then referencing this proxy in the SIP Peer profile assignment (later in this document).



The screenshot shows the 'Network Elements' configuration form. The form has a header with a green arrow icon and the title 'Network Elements'. Below the header, there are several fields for configuration. The 'Name' field is 'MBG62\_UDP'. The 'Type' field is a dropdown menu set to 'Outbound Proxy'. The 'FQDN or IP Address' field is '192.168.10.62'. The 'Local' field is 'False'. The 'Version' field is empty. The 'Zone' field is '1'. The 'ARID' field is empty. Below these fields, there is a section titled 'Outbound Proxy Specific'. The 'Outbound Proxy Transport Type' field is a dropdown menu set to 'UDP'. The 'Outbound Proxy Port' field is '5060'. At the bottom right, there are two buttons: 'Save' and 'Cancel'.

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

Figure 6 – Network Element Assignment (Proxy) for UDP



The screenshot shows the 'Network Elements' configuration form. The form has a header with a green arrow icon and the title 'Network Elements'. Below the header, there are several fields for configuration. The 'Name' field is 'MBG62\_TLS'. The 'Type' field is a dropdown menu set to 'Outbound Proxy'. The 'FQDN or IP Address' field is '192.168.10.62'. The 'Local' field is 'False'. The 'Version' field is empty. The 'Zone' field is '1'. The 'ARID' field is empty. Below these fields, there is a section titled 'Outbound Proxy Specific'. The 'Outbound Proxy Transport Type' field is a dropdown menu set to 'TLS'. The 'Outbound Proxy Port' field is '5061'. At the bottom right, there are two buttons: 'Save' and 'Cancel'.

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

Figure 7 – Network Element Assignment (Proxy) for TLS

## 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.

Trunk Attributes	
Trunk Service Number	2
Release Link Trunk	No
Call Recognition Service	Off
Direct Inward Dialing Service	<input type="radio"/> Off <input checked="" type="radio"/> On
Caller Based Routing Service	<input checked="" type="radio"/> Off <input type="radio"/> On
Class of Service	9
Class of Restriction	1
Baud Rate	300
Intercept Number	1
Non-dial In Trunks Answer Point - Day	
Non-dial In Trunks Answer Point - Night 1	
Non-dial In Trunks Answer Point - Night 2	
Dial In Trunks Incoming Digit Modification - Absorb	0
Dial In Trunks Incoming Digit Modification - Insert	
Dial In Trunks Answer Point	
Dial In Trunks Insert Forwarding Information	<input checked="" type="radio"/> No <input type="radio"/> Yes
Trunk Label	VAICC
<div>Save Cancel</div>	

Figure 8 – 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 "AudioCodes VAICC Platform" Network Element.

**Registration User Name:** The Mitel MiVB does not support Bulk Registration; therefore, trunks will have to be registered individually. Enter the Value assigned by AudioCodes VAICC Platform 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** AudioCodes VAICC Platform. 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 AudioCodes VAICC Platform.

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

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
<b>SIP Peer Profile Label</b>								VAICC
<b>Network Element</b>								VAICC
<b>Local Account Information</b>								
Registration User Name								
Address Type								IP Address: 192.168.10.96
<b>Administration Options</b>								
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
<b>Authentication Options</b>								
User Name								
Password								*****
Confirm Password								*****
Authentication Option for Incoming Calls								No Authentication
Subscription User Name								
Subscription Password								*****
Subscription Confirm Password								*****
<b>Gateway Options</b>								
Digital Trunk Licenses								0
Maximum Digital/Analog Channels								0

Figure 9 – SIP Peer Profile Assignment- Basic

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Alternate Destination Domain Enabled								No
Alternate Destination Domain FQDN or IP Address								
Enable Special Re-invite Collision Handling								No
Only Allow Outgoing Calls								No
Private SIP Trunk								No
Reject Incoming Anonymous Calls								No
Route Call Using P-Called-Party-ID (if present)								Yes
Route Call Using To Header								No

Figure 10 – 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
<b>Default CPN</b>								
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 11 – SIP Peer Profile Assignment- Calling Line ID





Use P-Call-Leg-ID Header	No
Use P-Early-Media Header	No
Use P-Preferred Identity Header	No
Use Restricted Character Set For Authentication	No
Use To Address in From Header on Outgoing Calls	No
Use user=phone	No
Use user=phone for Diversion Header	No
User-Defined Header Name	
User-Defined Header Value	

Figure 13 – SIP Peer Profile Assignment- Signaling and Header Manipulation

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Keep-Alive (OPTIONS) Period								120
Registration Period								3600
Registration Period Refresh (%)								50
Registration Maximum Timeout								90
Session Timer								1800
Session Timer: Local as Refresher								No
Subscription Period								3600
Subscription Period Minimum								300
Subscription Period Refresh (%)								80
Invite Ringing Response Timer								0

Figure 14 – SIP Peer Profile Assignment- Timers

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Allow In-Subscriptions for Local Digit Monitoring								No
Allow Out-Subscriptions for Remote Digit Monitoring								No
Force Out-Subscriptions for Remote Digit Monitoring								No
Request Outbound Proxy to Handle Out Subscriptions								No
KPML Transport								default
KPML Port								0

Figure 15 – SIP Peer Profile Assignment- Key Press Event

Basic	Call Routing	Calling Line ID	SDP Options	Signaling and Header Manipulation	Timers	Key Press Event	Outgoing DID Ranges	Profile Information
Index		DID Range						CPN Substitution

Figure 16 – 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	Profile Information
Creator								
Date Created								
Created with Version								
Service Provider								
Vendor Notes								

Figure 17 – SIP Peer Profile Assignment- Profile Information

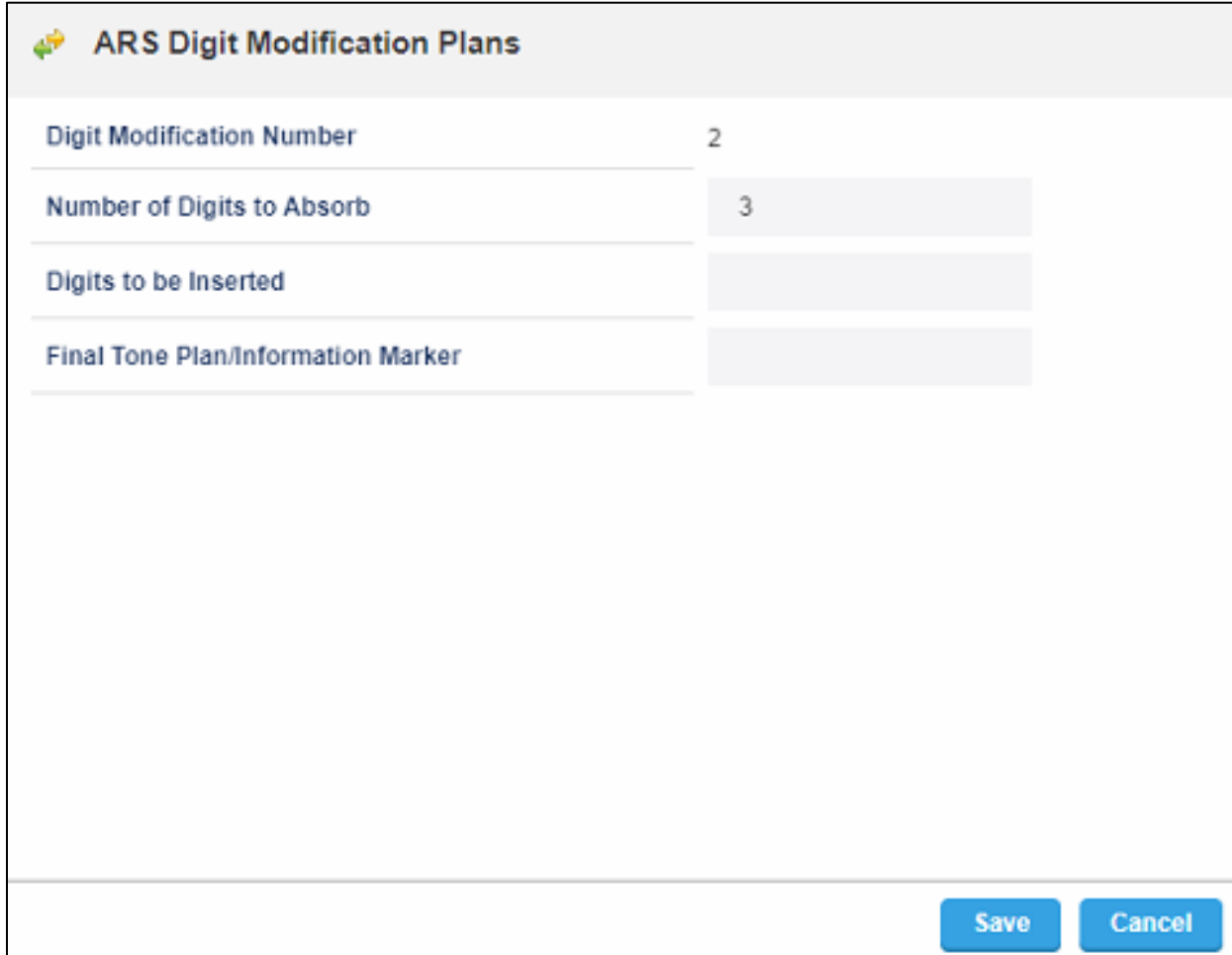
**Note** – All the above mentioned SIP peer profile screen shots are the same for the TLS trunk as well with AudioCodes VAICC platform except the below one (figure 18).

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								VAICCTLS
Network Element								VAICCTLS
Local Account Information								
Registration User Name								
Address Type								IP Address: 192.168.10.96
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								*****
Gateway Options								
Digital Trunk Licenses								0
Maximum Digital/Analog Channels								0

Figure 18 – SIP Peer Profile Assignment- Basic for TLS

### ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to AudioCodes VAICC Platform 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).



The image shows a web-based configuration form titled "ARS Digit Modification Plans". The form contains four input fields with labels on the left and values or input areas on the right. The first field, "Digit Modification Number", has the value "2". The second field, "Number of Digits to Absorb", has the value "3". The third field, "Digits to be Inserted", and the fourth field, "Final Tone Plan/Information Marker", are currently empty. At the bottom right of the form, there are two blue buttons labeled "Save" and "Cancel".

Field Label	Value
Digit Modification Number	2
Number of Digits to Absorb	3
Digits to be Inserted	
Final Tone Plan/Information Marker	

Figure 19 – Digit Modification Assignment

## ARS Routes

Create a route for SIP Trunks connecting a trunk to AudioCodes VAICC Platform. 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	<div>SIP Trunk</div>
Trunk Group Number	
SIP Peer Profile	<div>VAICC</div>
PBX Number / Cluster Element ID	
COR Group Number	1
Digit Modification Number	2
Digits Before Outpulsing	<div></div>
Route Type	<div>PSTN Access Via DPNSS</div>
Compression	<div>Off</div>
<div>SaveCancel</div>	

Figure 20 – 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 followed by any number, the call will be routed to AudioCodes VAICC Platform.

**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	5

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="text" value="Change to"/>	<input type="text" value="111"/>	<input type="text"/>
Number of Digits to Follow	<input type="text" value="Change to"/>	<input type="text" value="Unknown"/>	-
Termination Type	<input type="text" value="Change to"/>	<input type="text" value="Route"/>	-
Termination Number	<input type="text" value="Change to"/>	<input type="text" value="2"/>	<input type="text"/>

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

Figure 21 – ARS Digit Dialed Assignment

## Hunt Group Configuration

Hunt Group mapped to DID. Call will be forwarded to MICCB, Created Transfer Queue in MICCB which.

Hunt Groups						
3005	Terminal	Hunt Group_Bot	1	Voice	Local_96	Not Assigned
Hunt Group				3005		
Local-only DN				False		
Hunt Group Mode				Terminal		
Hunt Group Name				Hunt Group_Bot		
Class of Service - Day				3		
Class of Service - Night1				3		
Class of Service - Night2				3		
Zone ID						
Home Element				Local_96		
Secondary Element				Not Assigned		
First RAD						
Second RAD						
Night Answer RAD						
Hunt Group Priority				1		
Hunt Group Type				Voice		
Phase Timer Ring						
						<a href="#">Add Member</a> <a href="#">Change Member</a> <a href="#">Delete Member</a>
Hunt Group Members						
Member Index	Number	Presence	Name	Home Element	Secondary Element	
1	1502	Present	IVR PORT3	Local_96		
2	1503	Present	IVR Port4	Local_96		

Figure 22 – Hunt Group

AudioCodes VAICC Platform transfer is out of the Dialogflow / AudioCodes VAICC Platform, and back to the MiVB Hunt Group

Hunt Groups						
3006	Terminal		64	Voice	Local_96	Not Assigned
Hunt Group				3006		
Local-only DN				False		
Hunt Group Mode				Terminal		
Hunt Group Name						
Class of Service - Day				3		
Class of Service - Night1				3		
Class of Service - Night2				3		
Zone ID						
Home Element				Local_96		
Secondary Element				Not Assigned		
First RAD						
Second RAD						
Night Answer RAD						
Hunt Group Priority				64		
Hunt Group Type				Voice		
Phase Timer Ring						
						<a href="#">Add Member</a> <a href="#">Change Member</a> <a href="#">Delete Member</a>
Hunt Group Members						
Member Index	Number	Presence	Name	Home Element	Secondary Element	
1	1504	Present	Agent_IVR	Local_96		
2	1505	Present	Agent IVr4	Local_96		

Figure 23 – Hunt Group – Transfer from AudioCodes VAICC Platform

## MICC-B Configuration

When configuring MICCB (MBG), you need to identify the working MiVB ICP where MICCB and MIVB communicate through MITAI Messaged

Need to Create Appropriate COS for respective Ports

The screenshot shows the 'Media servers' configuration page in the YourSite interface. The left sidebar contains a tree view with 'Enterprise' and 'Devices' sections. The main content area is titled 'Media servers' and includes a search bar, a table of media servers, and a detailed configuration form for the selected server 'MIVB-96'.

Name	Historical	Site	Type	Last modified	Last modified by
MIVB-96	<input type="checkbox"/>	Default Site	ICP3300 with MITAI	12/19/2022 2:04:03 AM	Data Synchronization Service

1 of 1 selected. Total: 1

### MIVB-96

General | MITAI Options | Location | Telephone system settings | Data summary options | Call recording options | Data collection | Record Agent Greeting

**General**

Name: MIVB-96  
Media Server ID: 3  
Site: Default Site  
Computer name: WIN-9HBK85GMQ1K  
Telephone System Version: 20.4.1.34

☒ Enabled for alarms ☒ SDS Mode  
☒ Uses hot desking agents

Licensing  
☒ Licensed for Business Reporter Make historical

**Telephone system connection settings**

IP address / DNS name: 192.168.10.96  
Username: system  
Password: \*\*\*\*\*  
Confirm Password: \*\*\*\*\*  
Test Connection

Figure 24 – MICCB Media Server Configuration page

## Create Extension in MICCB for IVR and Port Membership

Need to create Appropriate extension type for IVR routing Workflow

The screenshot shows the 'Extensions' configuration page in the YourSite interface. The left sidebar contains a tree view with 'Enterprise' and 'Devices' sections. The main content area is titled 'Extensions' and includes a search bar, a table of extensions, and a detailed configuration form for the selected extension.

Name	Reporting number	Extension type	Media server	Failover media server	Workflow	Server Name	Licensed	Real time	Last modified
1010.1010	1010	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/13/2022 2:04:20 AM
2012.softphon	2103	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/14/2022 2:03:39 AM
5330e.530X	1002	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/28/2022 12:50:07 P.
6920.MINET	1000	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/28/2022 12:50:07 P.
6930.MINET	1001	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/28/2022 12:50:07 P.
6940.6940	1011	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/23/2022 3:16:45 PM
6940.myfab	2102	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/13/2022 2:04:20 AM
Agent IVr4	1505	Messaging port 5020 IP	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/8/2022 6:15:13 PM
Agent IVR	1504	Messaging port 5020 IP	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/8/2022 6:15:13 PM
Agent2.Agen53xx	2101	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/13/2022 2:04:20 AM
Base.Ext1	2100	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/23/2022 3:25:46 PM
IVR PORT3	1502	Messaging port 5020 IP	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/29/2022 2:08:12 AM
IVR Port4	1503	Messaging port 5020 IP	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/29/2022 2:08:12 AM
Ivr2	1501	Messaging port 5020 IP	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/7/2022 5:12:12 PM
MINET.6940	1003	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/15/2022 2:05:06 AM
SIP.user	2104	Voice	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	11/26/2022 2:04:07 AM
ucadmin.ucadmin	2116	Voice	MIVB-96				<input type="checkbox"/>	<input type="checkbox"/>	11/29/2022 2:08:12 AM
UVR Port	1500	Messaging port 5020 IP	MIVB-96				<input type="checkbox"/>	<input checked="" type="checkbox"/>	12/7/2022 5:12:12 PM

0 of 18 selected. Total: 18

Figure 25 – MICCB Extensions



## IVR Routing

The first leg of the call, when it initially enters the MiCC workflow, will call a REST API running from the MiCC server. This is running from IIS and is called DialogFlowDataService. The endpoint that is being called is PhoneNumber. The workflow will call this PUT request (see below) passing in the ANI/CLI of the caller. The ANI/CLI is written to the DB DialogFlowDataService table tblDialogFlowData CLI field. The PhoneNumber that is returned is assigned to a variable TX\_DEST. This number will be ARS digits (one of one thousand numbers in the table) that will be used to transfer the caller to the AudioCodes VAICC platform.

The screenshot displays the 'Subroutines' configuration window in the IVR Routing application. The left sidebar shows a tree view with categories like Workflows, Subroutines, Prompts, Rules, Holidays, Data providers, Variables, Queues, Extensions, and Hunt Groups. The main area is titled 'Subroutines' and contains a table listing various subroutines.

Name	Type	IVR Sync Status	Validate workflow	Last modified
Default Dial Out Of Queue Callback Request Subroutine	Inbound	Sync complete	<input checked="" type="checkbox"/>	11/22/2022 4:00:57 PM
Default Management Subroutine	Management	Sync complete	<input checked="" type="checkbox"/>	2/27/2020 3:30:02 PM
Default Inbound Voice Callback Subroutine	Callback Inbound	Sync complete	<input checked="" type="checkbox"/>	4/13/2020 12:53:54 PM
Default Outbound Voice Callback Subroutine	Outbound	Sync complete	<input checked="" type="checkbox"/>	8/13/2020 7:48:16 PM
DialogFlowReturn	Inbound	Sync complete	<input checked="" type="checkbox"/>	11/30/2022 8:27:57 PM
DialogFlowTransfer	Inbound	Sync complete	<input checked="" type="checkbox"/>	12/19/2022 6:40:32 PM

Below the table, it indicates '1 of 6 selected. Total : 6'. The 'DialogFlowTransfer' subroutine is selected, and its workflow is displayed in a diagram. The workflow starts with an 'Execute' step: 'http://localhost/dialogflowdataservice/api/phoneNumber'. This leads to a 'Transfer' step: '<<TxDest>>'. The 'Transfer' step has three paths: 'Success', 'Abandoned', and 'Failure'. Each path leads to a 'Hang Up' step. There are also 'Timeout' and 'Failure' steps with 'Drop Activities Here' options.

Figure 26 – Transfer to User Workflow

Web Service | <http://localhost/dialogflowdataservice/api/phonenumbers> | Rest

**Process Setup** Edit Process

URI:

Http Action:

Web Service Type:

Username:

Password:

**Headers**

Add Remove

Parameter	Value	Test Value
-----------	-------	------------

**Input Parameters**

Add Remove

Parameter	Value	Test Value
ani	ANI	12345
dnis	DNIS	54321

<http://localhost/dialogflowdataservice/api/phonenumbers/12345/54321>

Execute

**Output Mappings**

Output	Mapping
PhoneNumber	TxDest
CLI	
TimeCalled	
Data	
DNIS	
ExitCode	

OK Cancel

Figure 27 – API Configuration

**IVR Routing**

Extensions X Workflows X Subroutines X

**Workflows**

Search Add Edit Delete Import Export Filter

Name	Type	IVR Sync Status	Validate workflow	Always run	Last modified
Default Callback Outbound Workflow	Outbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	4/13/2020 12:56:27 PM
Queue - Agent (2022-11-23 12:44:52 PM)	Agent	Sync complete	<input checked="" type="checkbox"/>	<input type="checkbox"/>	12/9/2022 11:09:09 AM
Transfer AC Bot	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	12/9/2022 11:36:03 AM
Venky Queue	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	12/9/2022 11:17:05 AM
Vinod Queue	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	12/16/2022 11:39:59 AM
Voice (2020-06-26 03:14:28 PM)	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	7/1/2020 12:30:12 PM

1 of 6 selected. Total: 6

**Vinod Queue** Sync complete

Designer **Hunt Group Membership** Port Membership

Available members

Search

Name	Dialable number	Media server
------	-----------------	--------------

Selected members

Search

Name	Dialable number	Media server
Hunt group - 3006	3006	MIVB-96

Figure 28 – Hunt Group Membership for Workflow

## Workflow for calls to transfer to Agent

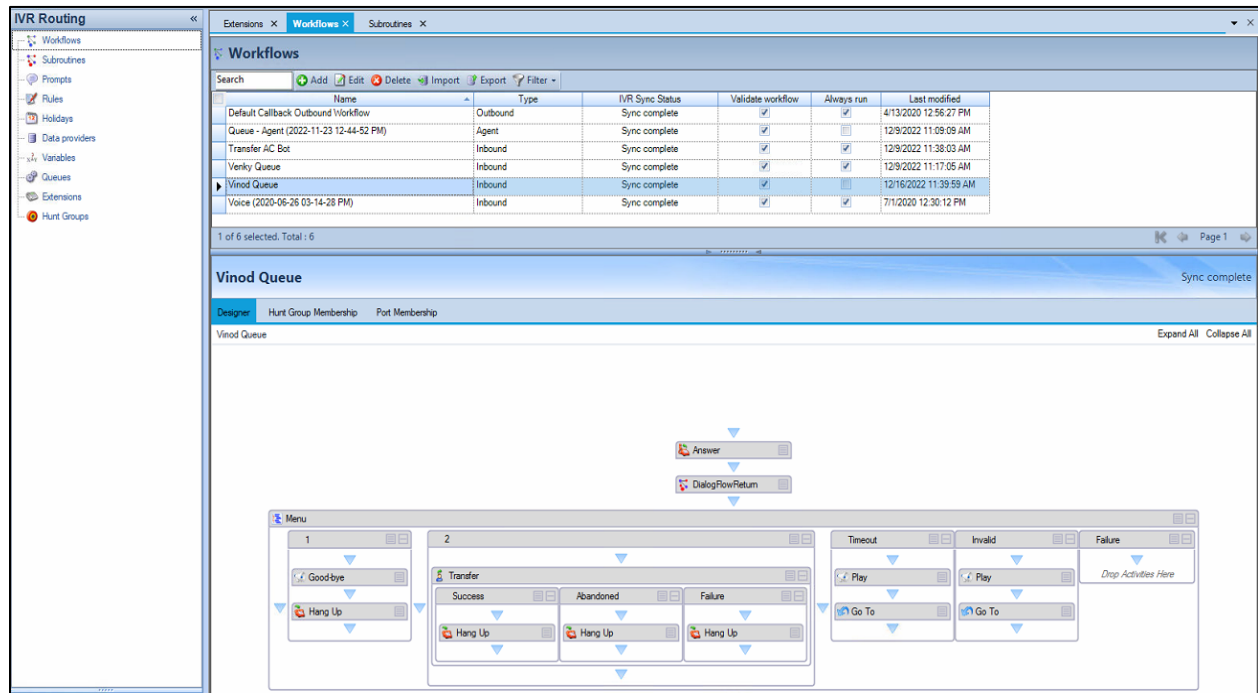


Figure 29 – Workflow

Web Service | <http://localhost/dialogflowdataservice/api/phoneNumber/{ani}> | Rest

**Process Setup** Edit Process

URI:

Http Action:

Web Service Type:

Username:

Password:

**Headers**

Add Remove

Parameter	Value	Test Value
-----------	-------	------------

**Input Parameters**

Add Remove

Parameter	Value	Test Value
ani	ANI	test

<http://localhost/dialogflowdataservice/api/phoneNumber/{ani}> Execute

**Output Mappings**

Output	Mapping
root	
PhoneNumber	
CLI	
DNIS	
TimeCalled	
Data	
RawData	

OK Cancel

Figure 30 – API Configuration

IVR Routing

Extensions X Workflows X Subroutines X

**Workflows**

Search Add Edit Delete Import Export Filter

Name	Type	IVR Sync Status	Validate workflow	Always run	Last modified
Default Callback Outbound Workflow	Outbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	4/13/2020 12:56:27 PM
Queue - Agent (2022-11-23 12:44:52 PM)	Agent	Sync complete	<input checked="" type="checkbox"/>	<input type="checkbox"/>	12/9/2022 11:09:09 AM
Transfer AC Bot	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	12/9/2022 11:36:03 AM
Venky Queue	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	12/9/2022 11:17:05 AM
Vinod Queue	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	12/16/2022 11:39:59 AM
Voice (2020-06-26 03:14:28 PM)	Inbound	Sync complete	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	7/1/2020 12:30:12 PM

1 of 6 selected. Total: 6

**Vinod Queue** Sync complete

Designer **Hunt Group Membership** Port Membership

Available members

Search

Name	Dialable number	Media server
------	-----------------	--------------

Selected members

Search

Name	Dialable number	Media server
Hunt group - 3006	3006	MIVB-96

Figure 31 – Hunt Group Membership for Workflow

## 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 right pane, click **Network** tab and then **ICPs** (see Figure 32 for details)

Page updated: Mon Dec 19 2022 18:58:34 GMT+0530 (India Standard Time)  
To test connectivity to your configured ICPs, or to run a DNS resolution test on configured hostnames, see the [Diagnostics](#) page.

ICP Information

Default for MiNet	Default for SIP	Name	Hostname or IP address	Type	Installer password	SIP capabilities	Indirect call recording capable	Associated connectors	Associated sets (MiNet/SIP)	Associated trunk rules (pri/sec)			
<input type="radio"/>	<input type="radio"/>	MiVB_69	192.168.10.69	MiVoice Business		UDP TCP TLS	✗	✗	2 / 0	0 / 0			
<input type="radio"/>	<input type="radio"/>	MiVB_94	192.168.10.94	MiVoice Business		UDP TCP TLS	✗	✗	7 / 2	0 / 0			
<input type="radio"/>	<input type="radio"/>	MiVB_95	192.168.10.95	MiVoice Business		UDP TCP TLS	✗	✗	0 / 0	0 / 0			
<input checked="" type="radio"/>	<input checked="" type="radio"/>	MiVB_96	192.168.10.96	MiVoice Business		UDP TCP TLS	✗	✗	0 / 0	3 / 0			
<input type="radio"/>	<input type="radio"/>	MiVO250	192.168.10.162	MiVoice Office 250		UDP	✗	✗	0 / 1	0 / 0			
<input type="radio"/>	<input type="radio"/>	MiVO250_169	192.168.10.169	MiVoice Office 250		UDP TCP TLS	✗	✗	0 / 0	0 / 0			

Figure 32 – MBG's Configuration page

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

System ▾ Network ▾ Teleworking ▾ SIP trunking ▾ Remote proxy ▾ Call recording ▾ Troubleshooting ▾ Search

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The following is a form for modifying an icp entry. You may edit this information as you wish, and click on the "Save" button below when you are done.

Manage ICP

Name	MiVB_96	Hostname or IP address	192.168.10.96
Type	MiVoice Business	MiNet installer password	
SIP capabilities	UDP, TCP, TLS	Indirect call recording capable	<input type="checkbox"/>

Save

Figure 33 – ICP configuration page

- Next configure the SIP trunking by click on the “Sip Trunking menu” and select Configuration
- On the SIP Trunking page click on the ‘+’ symbol and add AudioCodes VAICC Platform trunk, see Figure 34 and Figure 35.

System ▾ Network ▾ Teleworking ▾ SIP trunking ▾ Remote proxy ▾ Call recording ▾ Troubleshooting ▾ Search

Page updated: Mon Jul 20 2020 17:15:40 GMT+0530 (India Standard Time)

March 17, 2020, 5:47 a.m. **Note:** Remote proxy is now found in the main MBG menu instead of the server manager menu on the left. ✕ Dismiss

The SIP trunks information section below shows a short summary of each SIP trunk. Click on the SIP trunk for detailed information.

**Note:** To make changes to SIP settings in general, please see the [SIP options](#) in System Settings

**SIP trunk information**

+

Figure 34: MBG SIP Trunking Configuration

**Manage SIP trunk**

**Profile**

Enabled ☒

Name

**Authentication**

Authentication username

Authentication password

Confirm authentication password

**Protocol**

PRACK support

Options keepalives

Options interval

Rewrite host in PAI ☒

Idle timeout (s)

Use source port in contact header ☐

**Trunk-side RTP security**

Inbound

Outbound

Preferred cipher

**Connection**

Transport protocol

Remote trunk endpoint address

Remote trunk endpoint port

Accept traffic from all UDP ports ☒

**SIP adaptation**

Receive pipeline

Send pipeline

**Media**

Local streaming between trunk calls ☐

RTP address override

**Icp-side RTP security**

Inbound

Outbound

Preferred cipher

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	*	MIVB_96		

Figure 35: MBG SIP Trunking Configuration

Enter the SIP Trunking details as shown in Figure 35:

**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. AudioCodes VAICC Platform.

**Remote trunk endpoint port:** 5060.

**Options Keepalives:** Never.

**Options interval:** 60

**Remote RTP frame size (MS):** Is the packetization rate you want to set on this trunk. This option is typically set to Auto.

**RTP address override:** Leave blank.

**PRACK support:** Use Master Setting.

**SIP Adaptation Receive Pipeline:** Select Configured Plugin (Attached in KB Article)

**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.

- Check status: click SIP Trunking and then click Status, see Figure 36



Figure 36 – SIP Trunk Status

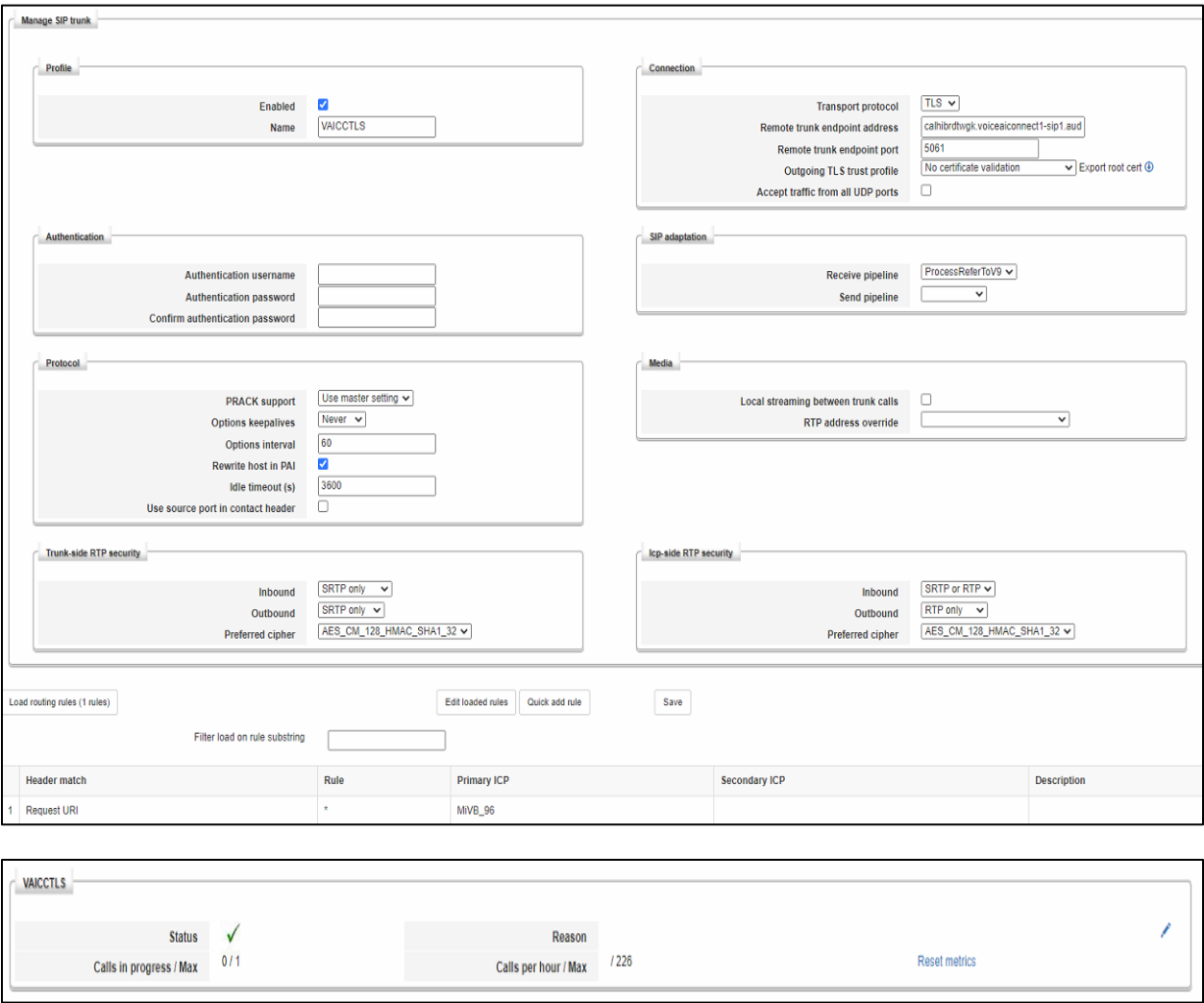


Figure 37: MBG SIP Trunking Configuration for TLS

**Note:**

The SIP Trunk used to access the AudioCodes VAICC will have a SIP Adaptation Receive Pipeline configured. This is a custom adaptation plugin (Lua script) written to process any REFER messages sent in the receive pipeline.

The AudioCodes VAICC transfers the caller using a REFER message. Any data to be sent with the call is appended to the REFER-To message as query parameters e.g. e.g. sip:

**3006@192.168.10.96?D1=%24session.params.main\_ivr\_selection**

The plugin will process these query parameters. It will use the FROM address as a file name appended with .json e.g. 58081000.json and will create this file using this file name in the folder /home/referto on the MBG server file system. It will write the query parameters converted to a JSON object into this file.

There is a file watcher process running on the MBG server. This is a Linux executable that monitors the folder /home/referto for any new file creations (it uses the Linux subsystem iNotify events for this purpose).

Any new file created in this folder will be read by the file watcher process. The data will then be posted to the DialogFlowDataService/API/ReferTo endpoint on the MiCC server (LibCurl is used for this purpose).

The file will then be deleted.

The file watcher process is configured a systemd service, this is configured in

/etc/systemd/system/file watcher.service

The file watcher service is configured to start at run time (systemctl enable file watcher)

The status can be checked by using systemctl status file watcher

The file watcher process writes logs to /var/logs/messages.



## Glossary

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