Mitel Technical Configuration Notes – HO4909

December 26, 2022

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

NOTICE

TRADEMARKS

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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:

COMPATIBLE	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.

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	☑
Codec	G711 codec	☑
TLS/SRTP	Transferring the call to AudioCodes VAICC Platform and AudioCodes VAICC Platform transferring the call back to MiVB	☑

☑ - 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	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. **Recommendation**: Please contact Mitel Support for more information on this.

Network Topology

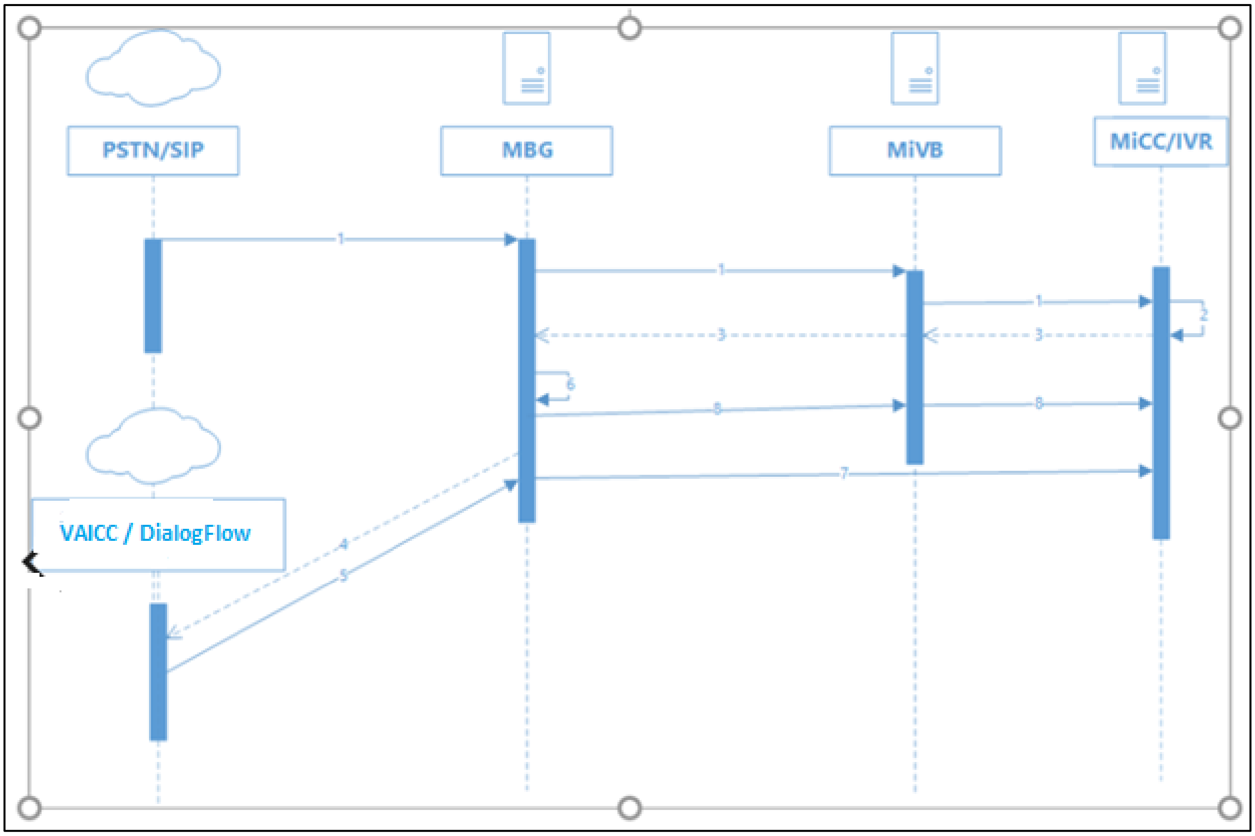


Figure 1 – Network Topology

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.

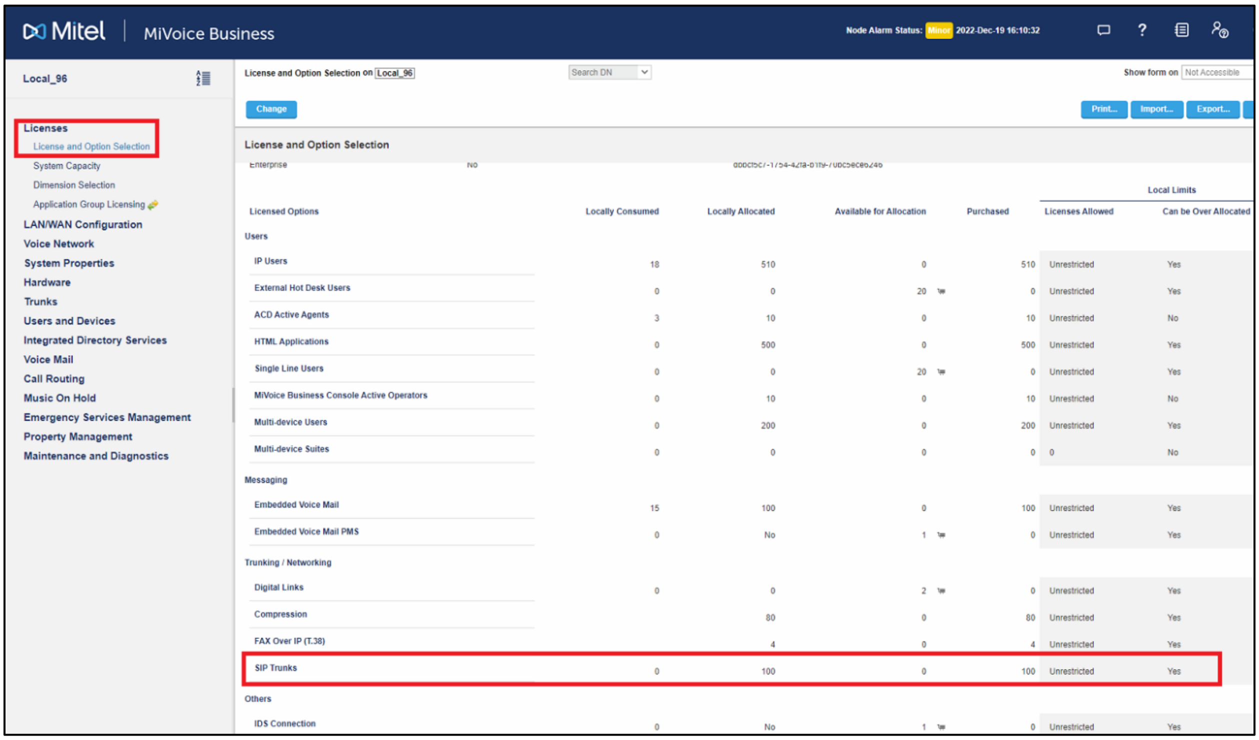


Figure 2 – License and Option Selection

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.

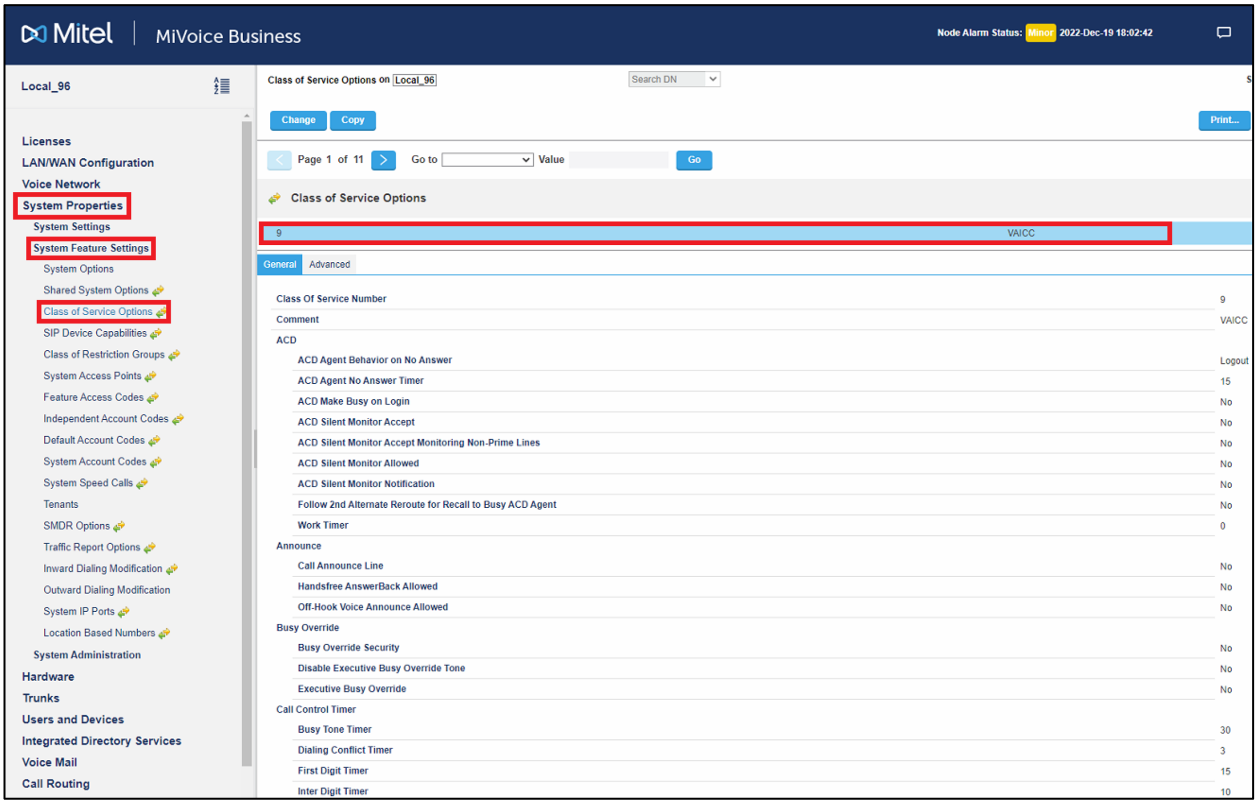


Figure 3 – Class of Service

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).

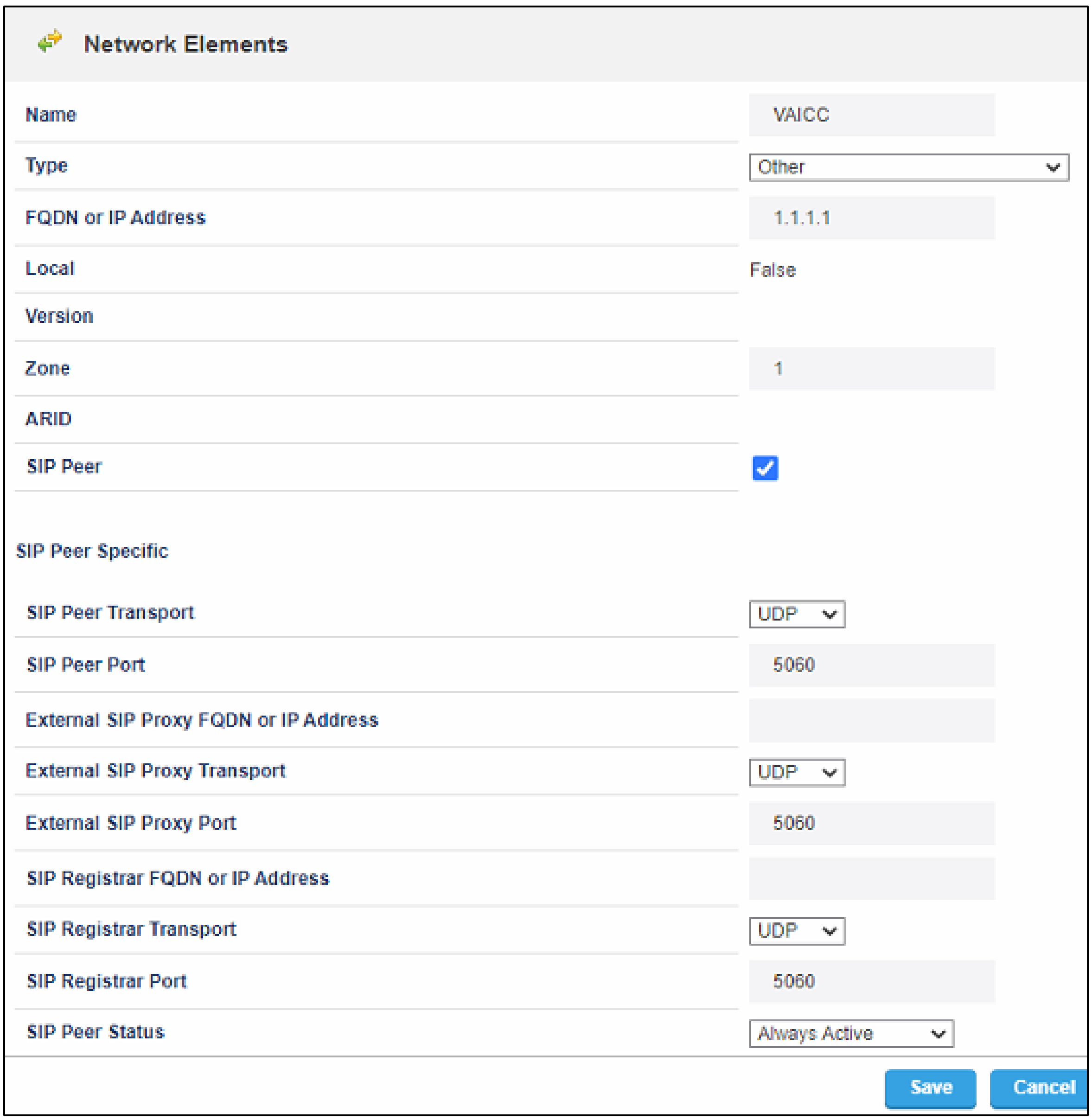


Figure 4 – Network Element Assignment for UDP

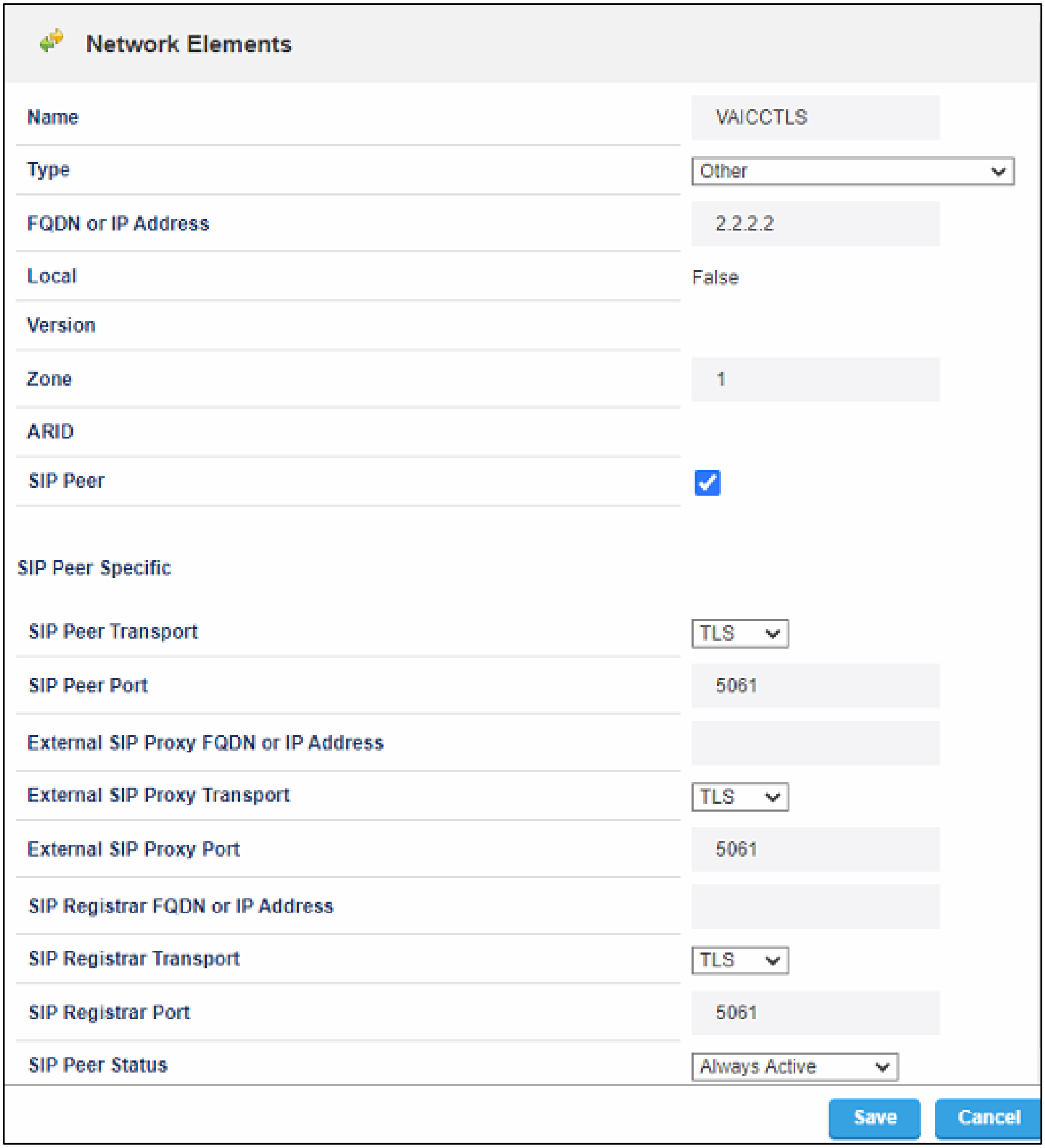


Figure 5 – Network Element Assignment for TLS

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).

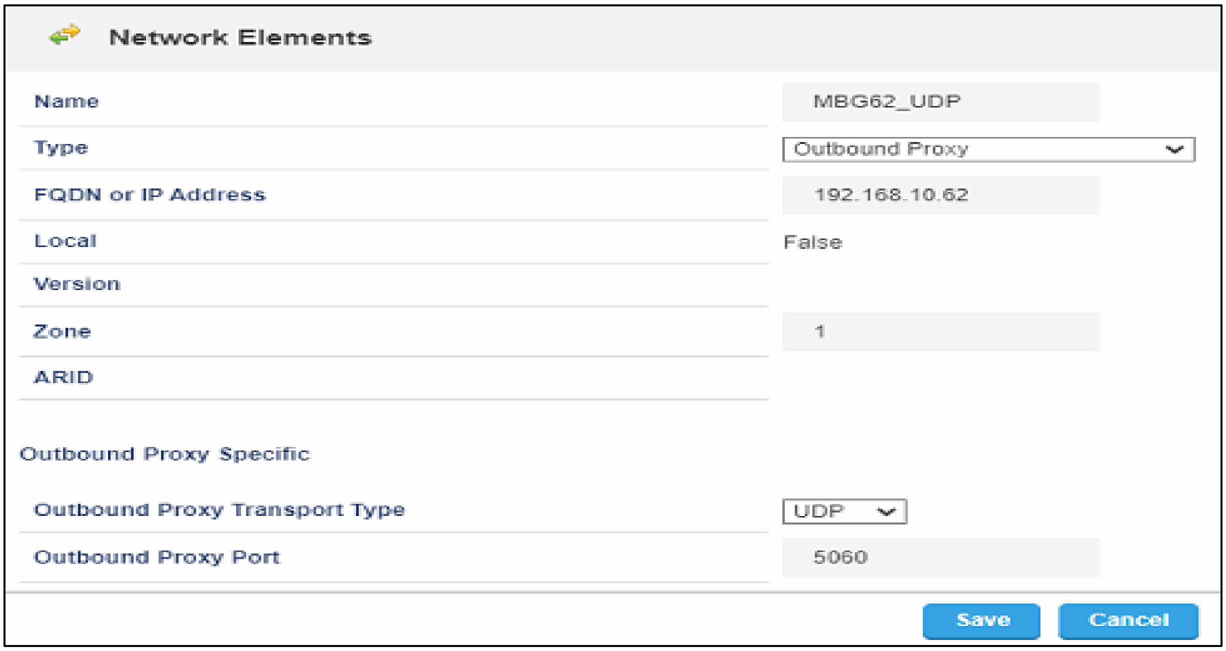


Figure 6 – Network Element Assignment (Proxy) for UDP

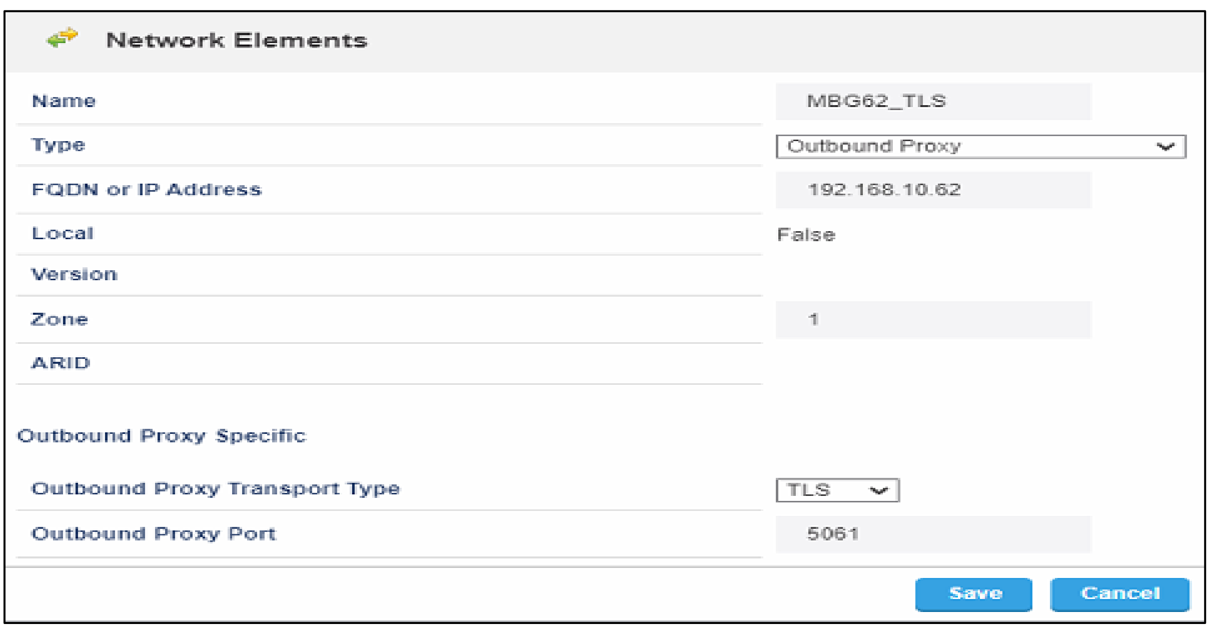


Figure 7 – Network Element Assignment (Proxy) for TLS

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.

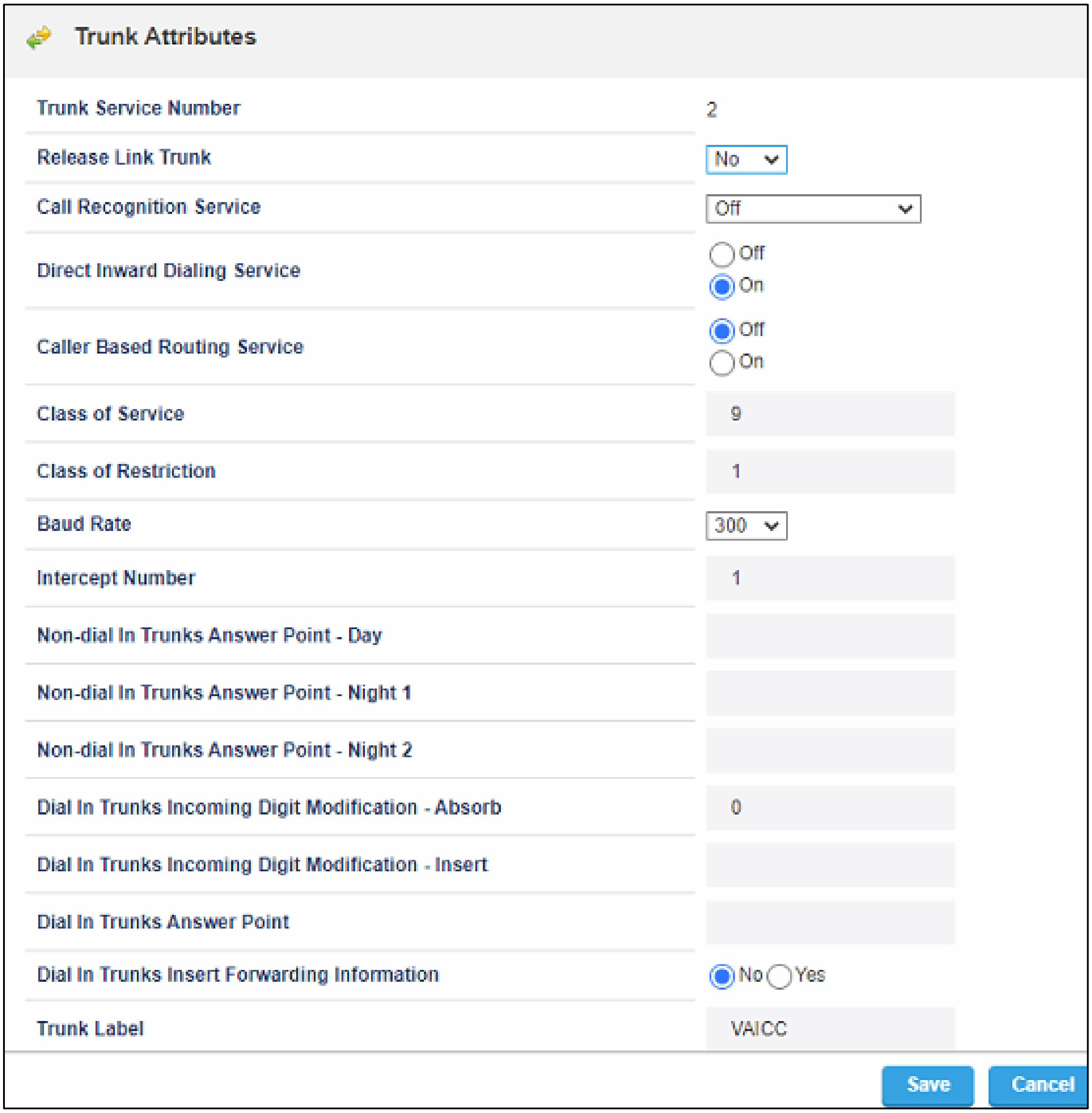


Figure 8 – Trunk Attributes

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.

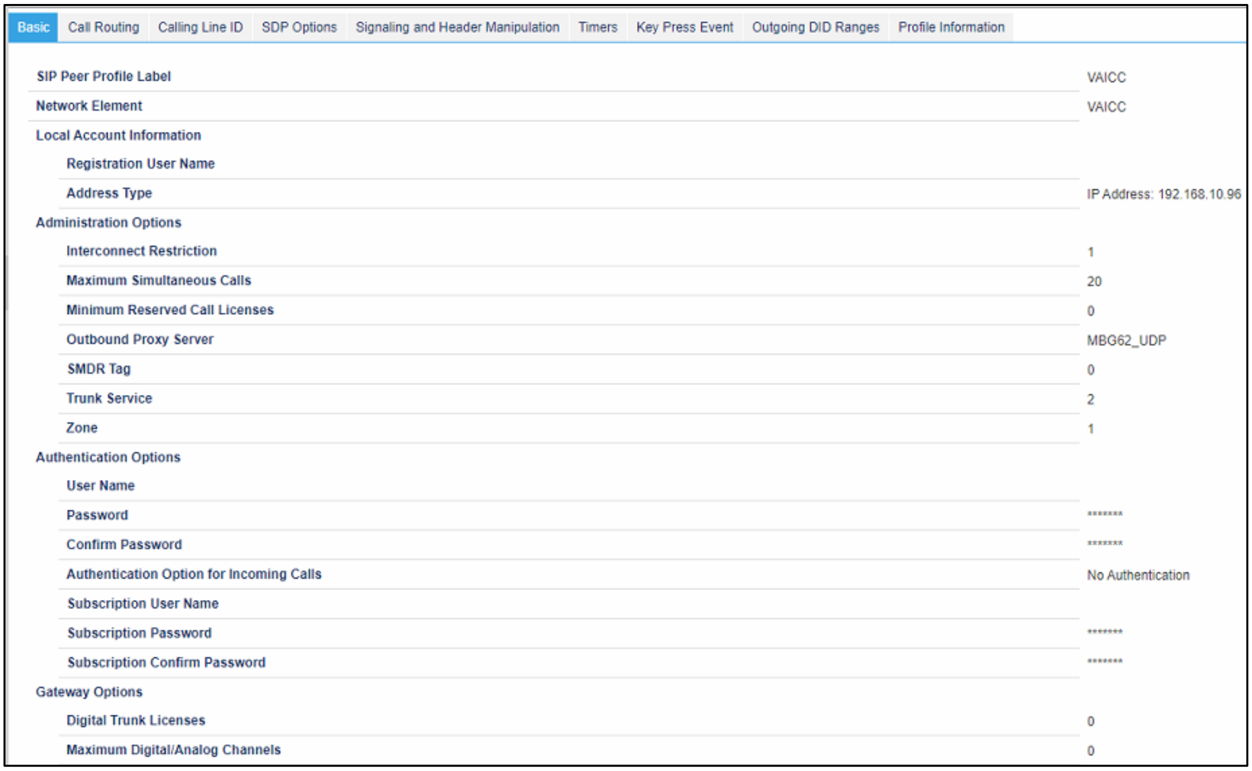
Calling Line ID: The default CPN is applied to all calls unless there is a match in the "Outgoing DID Ranges" of the SIP Peer Profile. **This number will be provided by** AudioCodes VAICC Platform**.** Do not use a Default CPN if you want public numbers to be preserved through the SIP interface. Add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

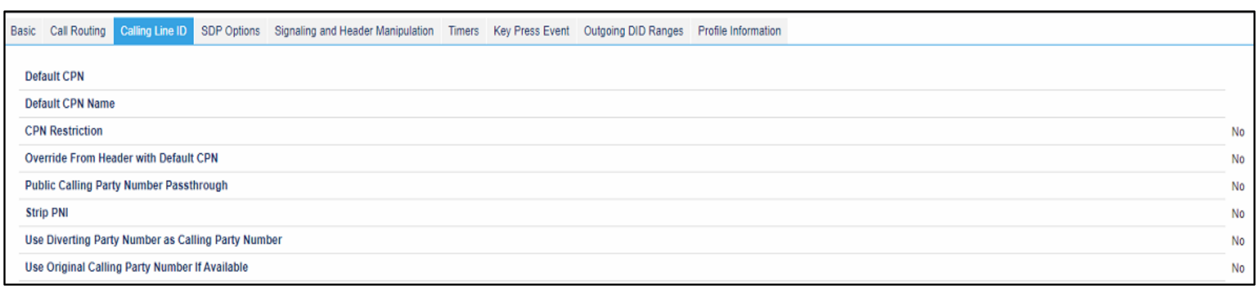
SIP Peer Profile Label	VAICC
Network Element	VAICC
Local Account Information	
Registration User Name	
Address Type	IP Address: 192.168.10.96
Administration Options	
Interconnect Restriction	1
Maximum Simultaneous Calls	20
Minimum Reserved Call Licenses	0
Outbound Proxy Server	MBG62_UDP
SMDR Tag	0
Trunk 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 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

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 11 – 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 12 – 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
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-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 Inc Subscriptions for Local Digit Monitoring	No
Allow Out Subscriptions for Remote Digit Monitoring	No
Force Out Subscriptions for Remote Digit Monitoring	No
Request Outbound Proxy to Handle Out Subscriptions	No
KPML Transport	default
KPML Port	0

Figure 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

Update

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).

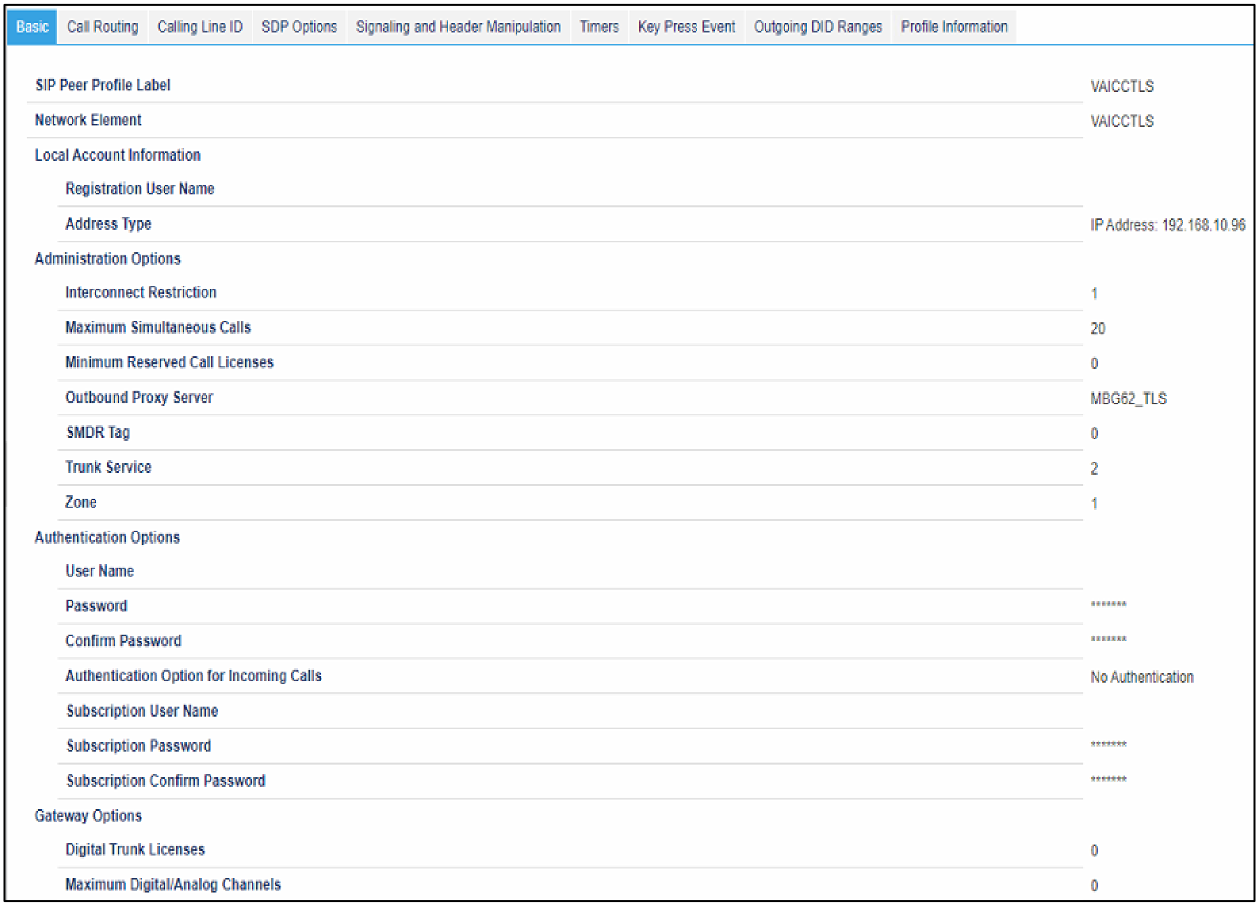


Figure 18 – SIP Peer Profile Assignment- Basic for TLS

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).

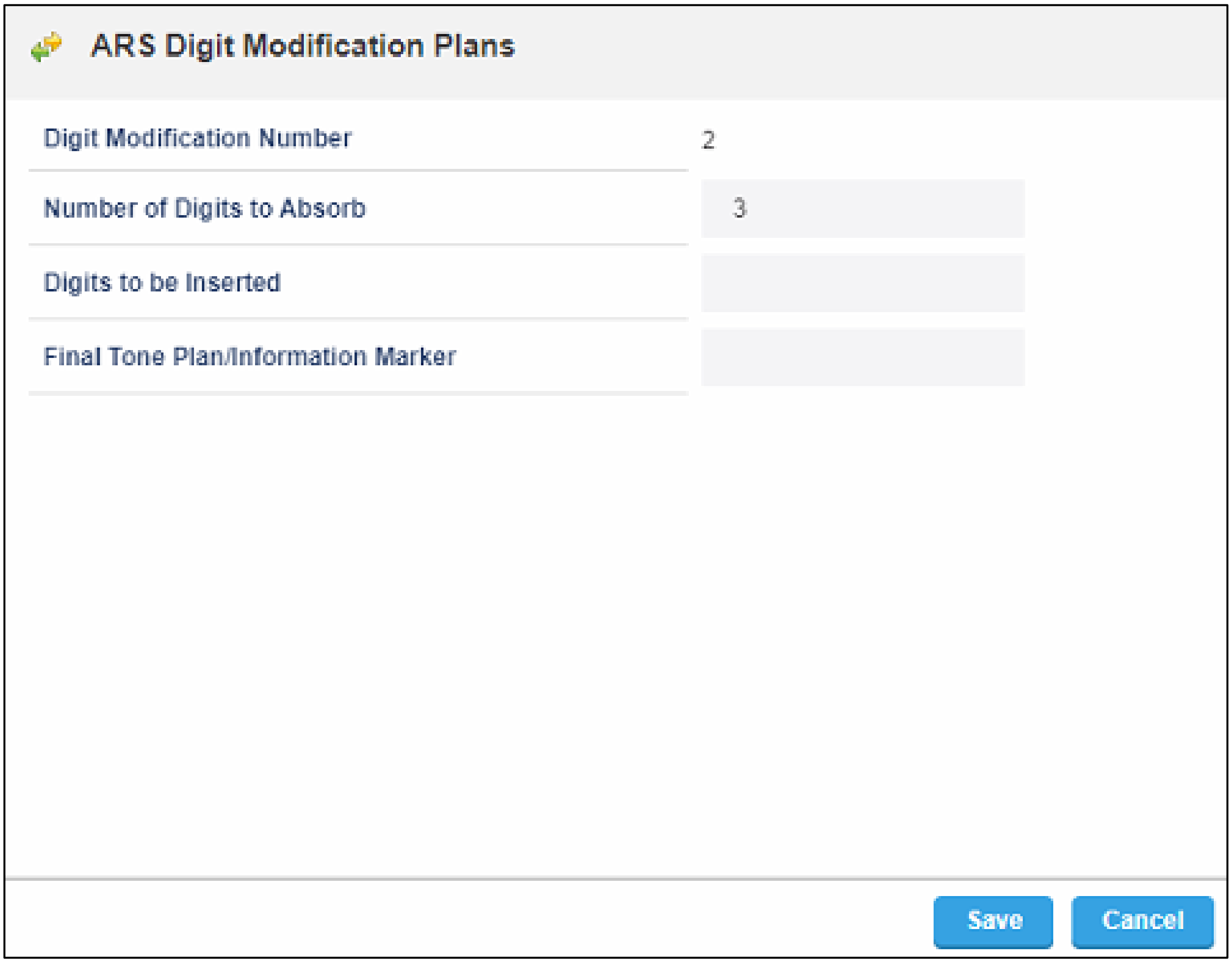


Figure 19 – Digit Modification Assignment

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.

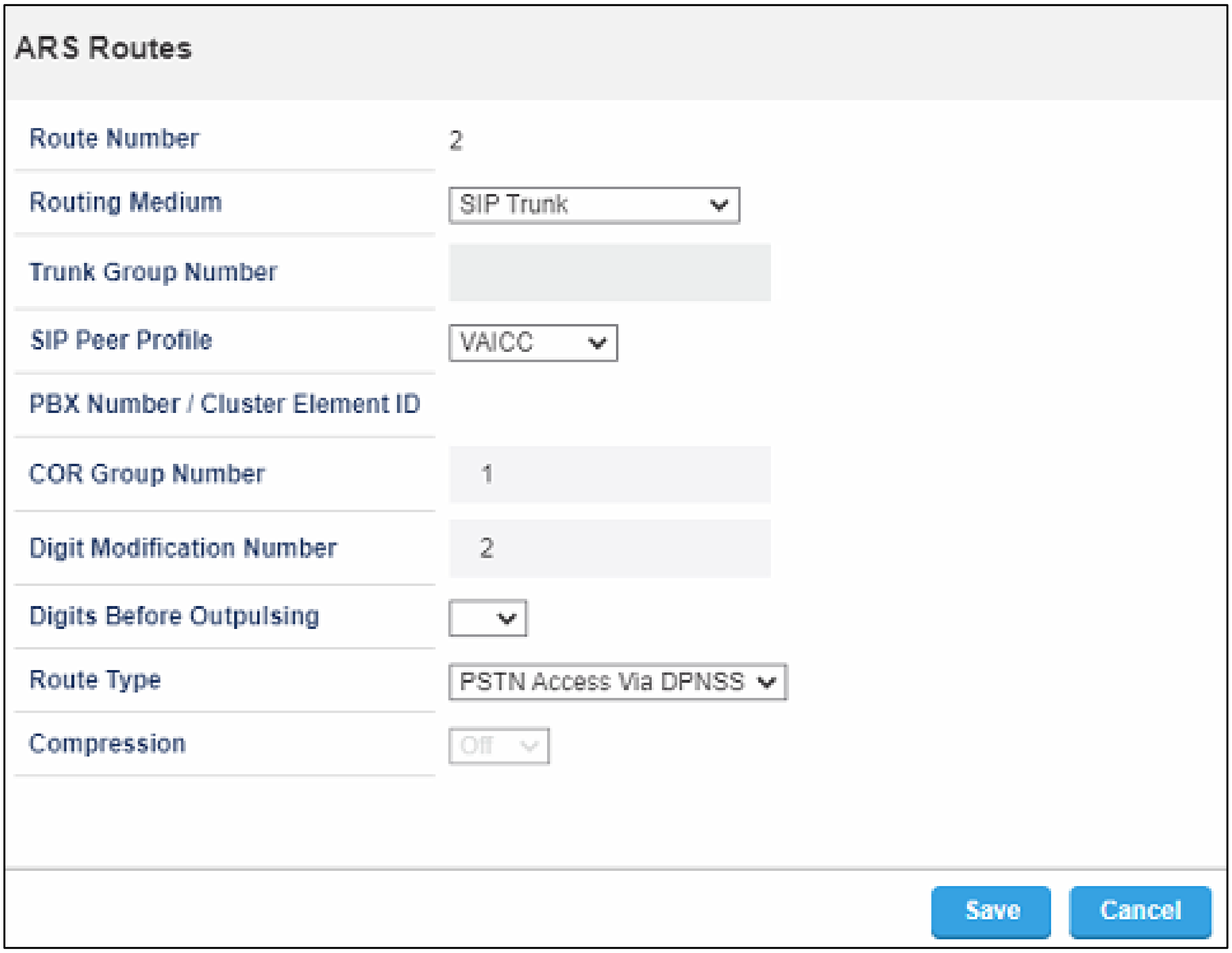


Figure 20 – SIP Trunk Route Assignment

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.

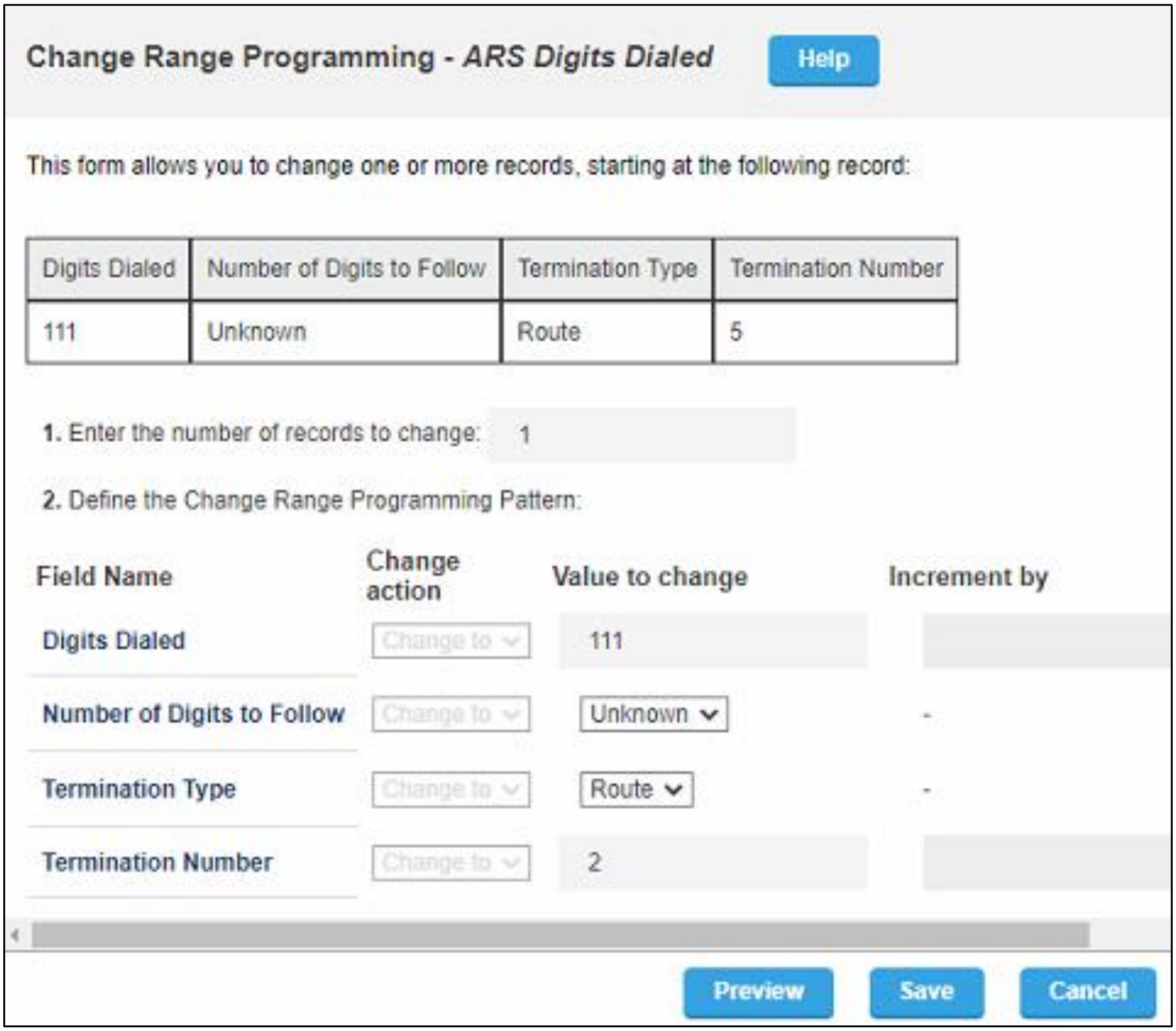


Figure 21 – ARS Digit Dialed Assignment

Hunt Group Configuration

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

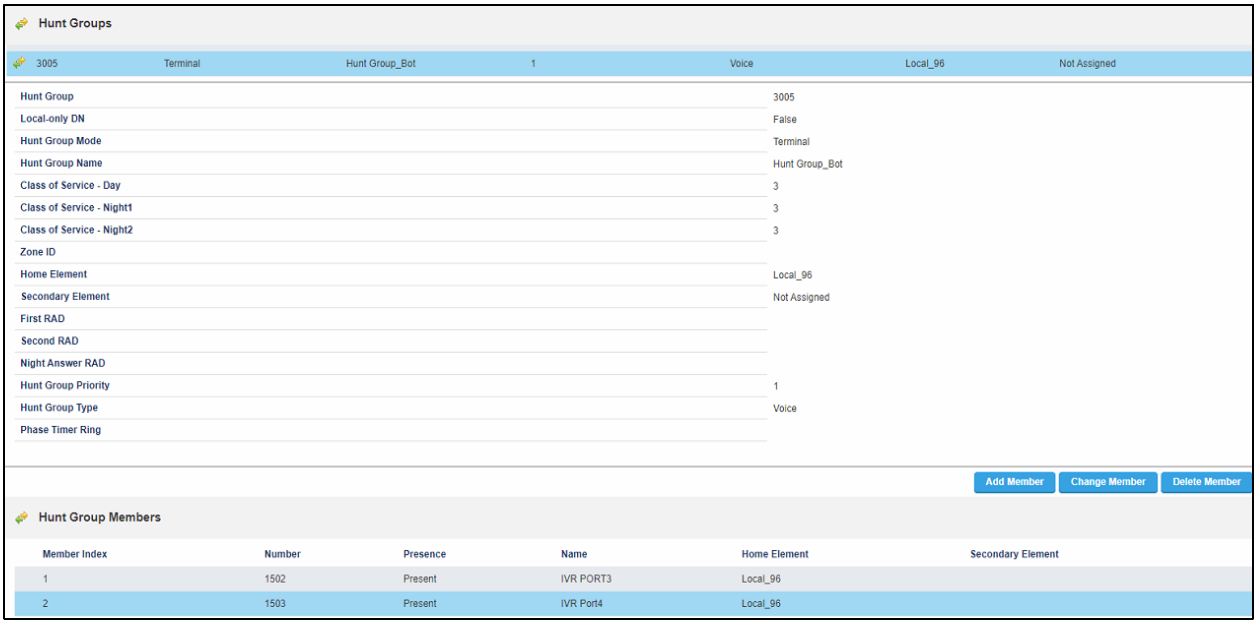


Figure 22 – Hunt Group

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

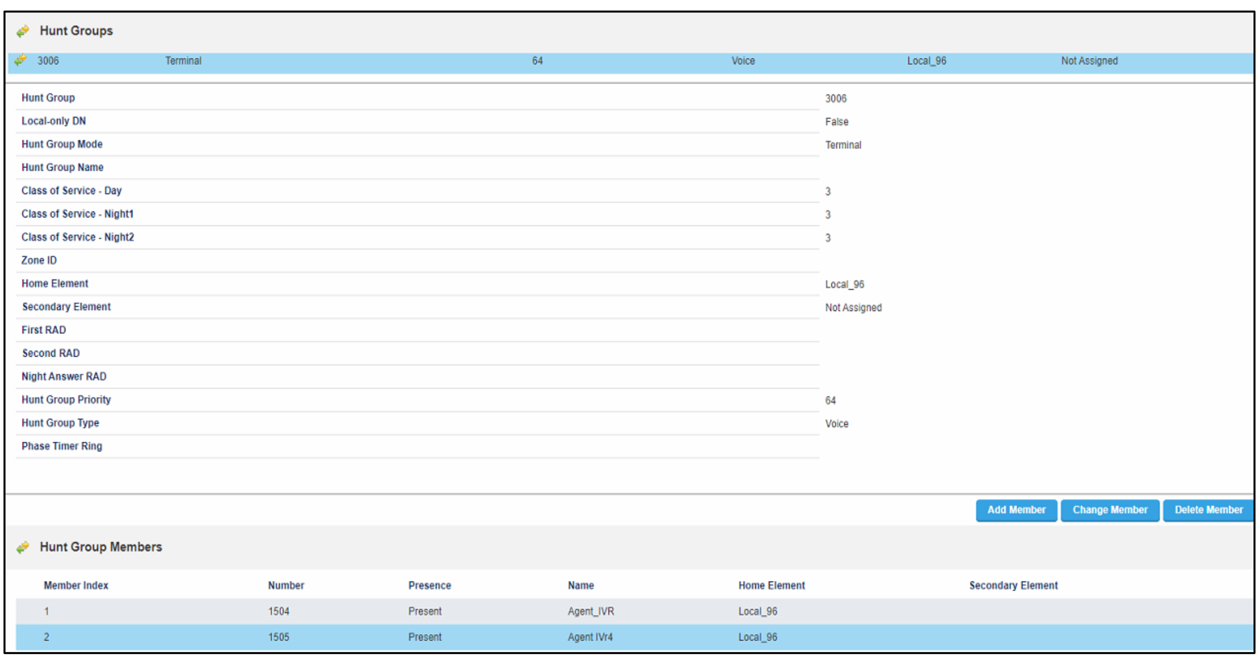


Figure 23 – Hunt Group – Transfer from AudioCodes VAICC Platform

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

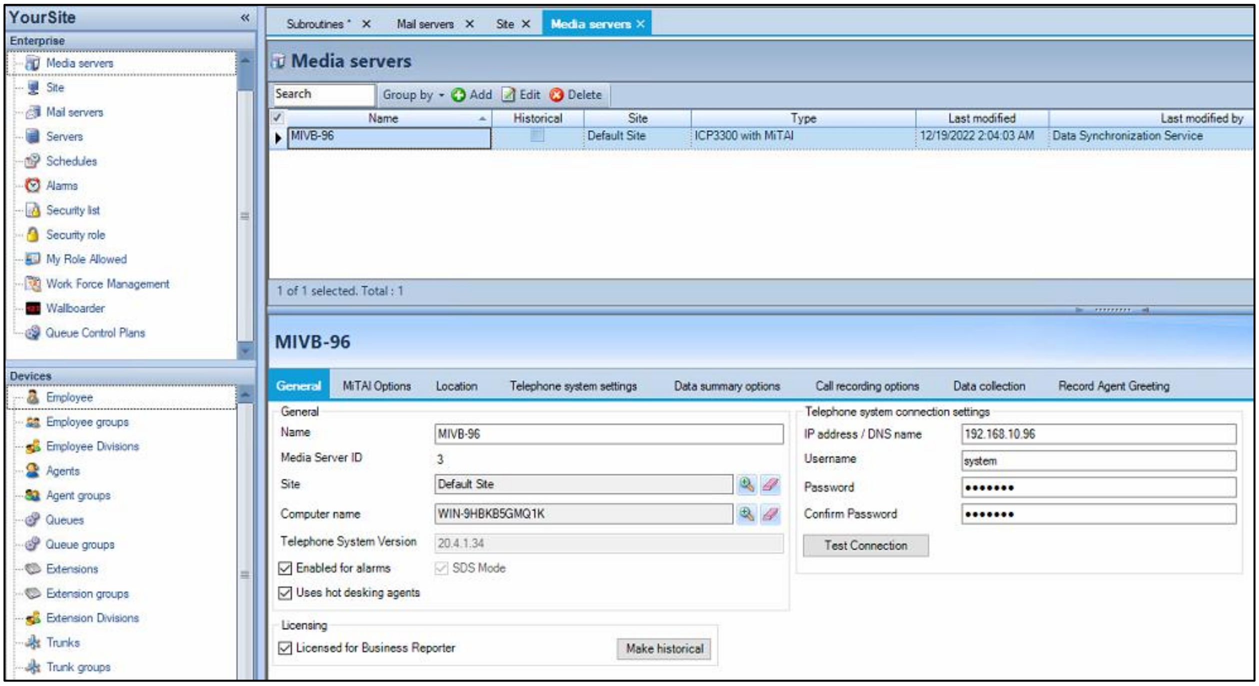


Figure 24 – MICCB Media Server Configuration page

Create Extension in MICCB for IVR and Port Membership

Need to create Appropriate extension type for IVR routing Workflow

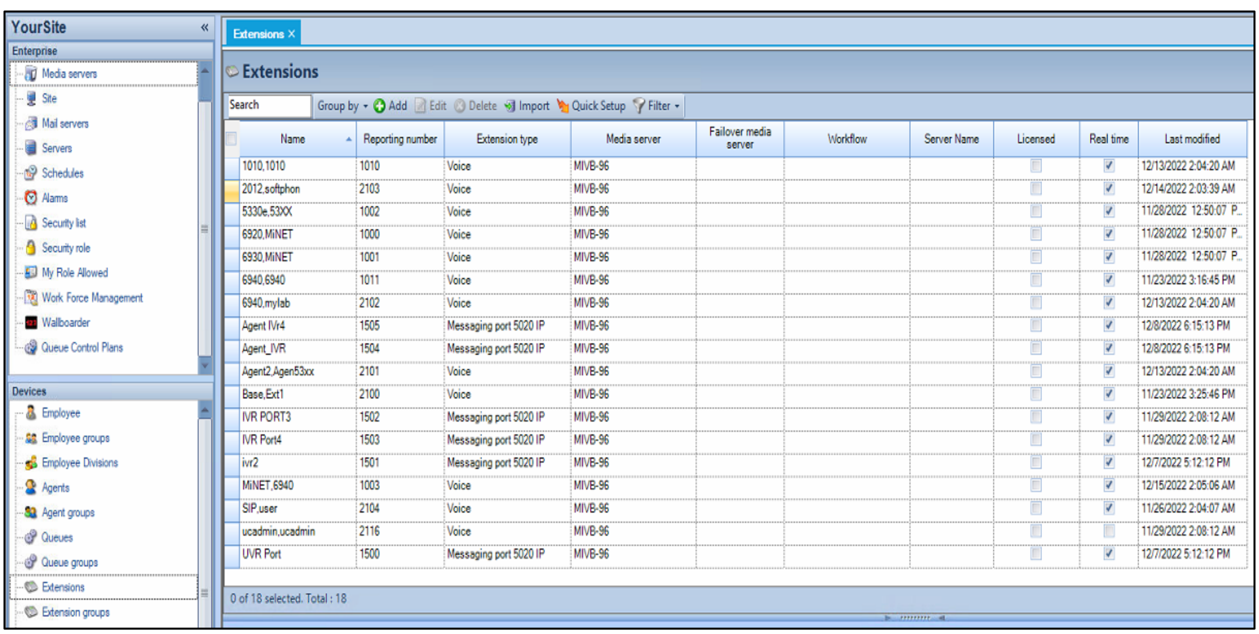


Figure 25 – MICCB Extensions

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.

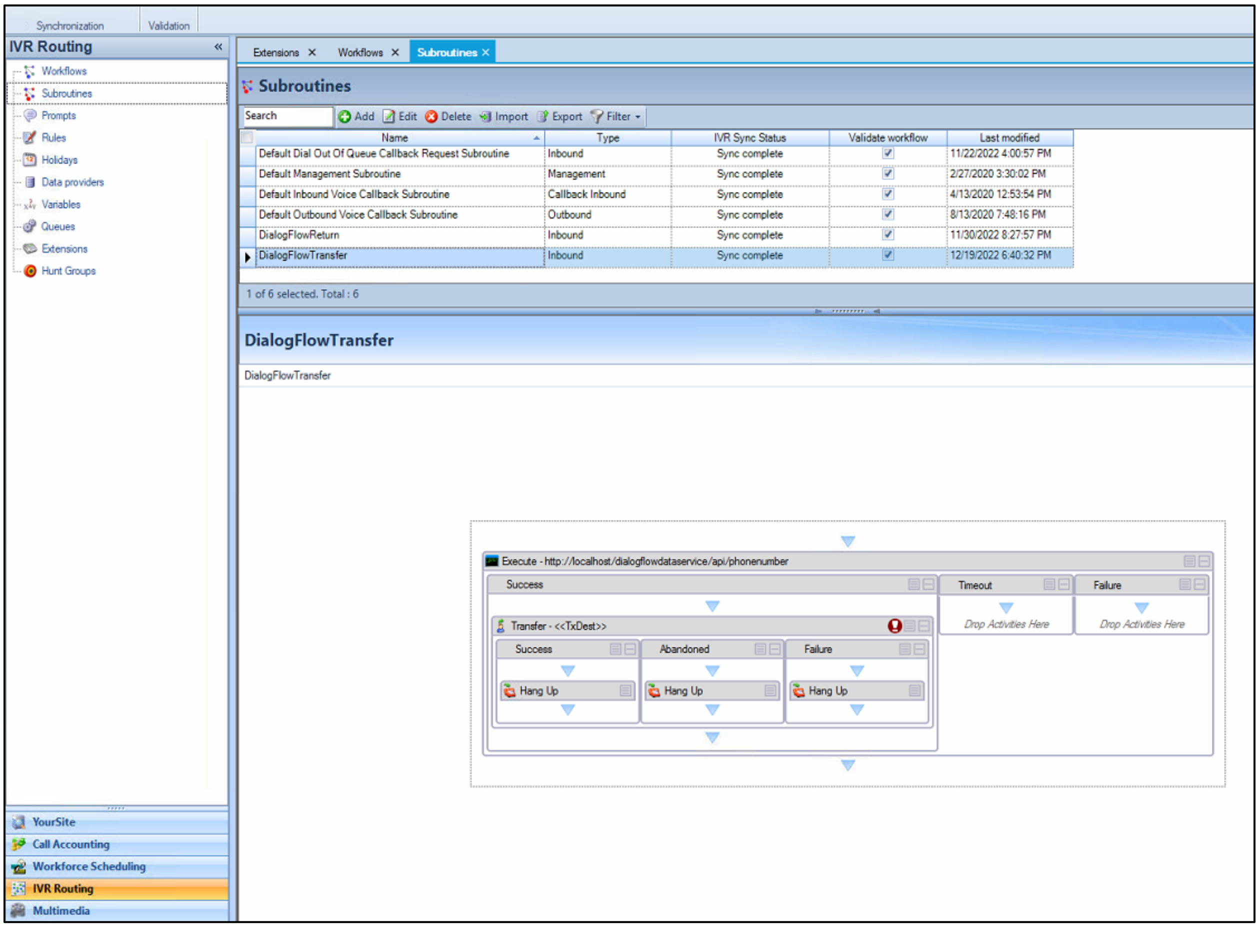


Figure 26 – Transfer to User Workflow

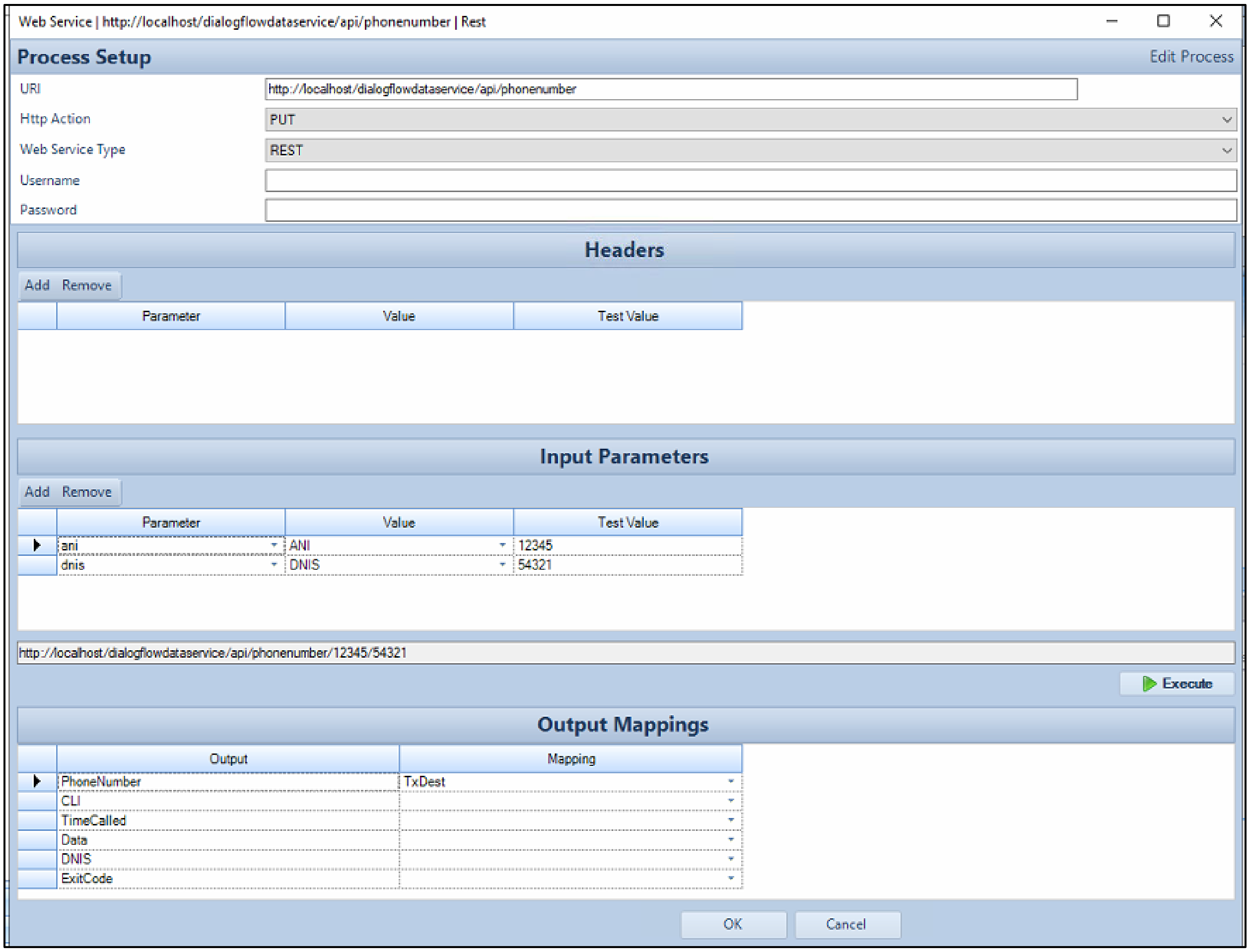


Figure 27 – API Configuration

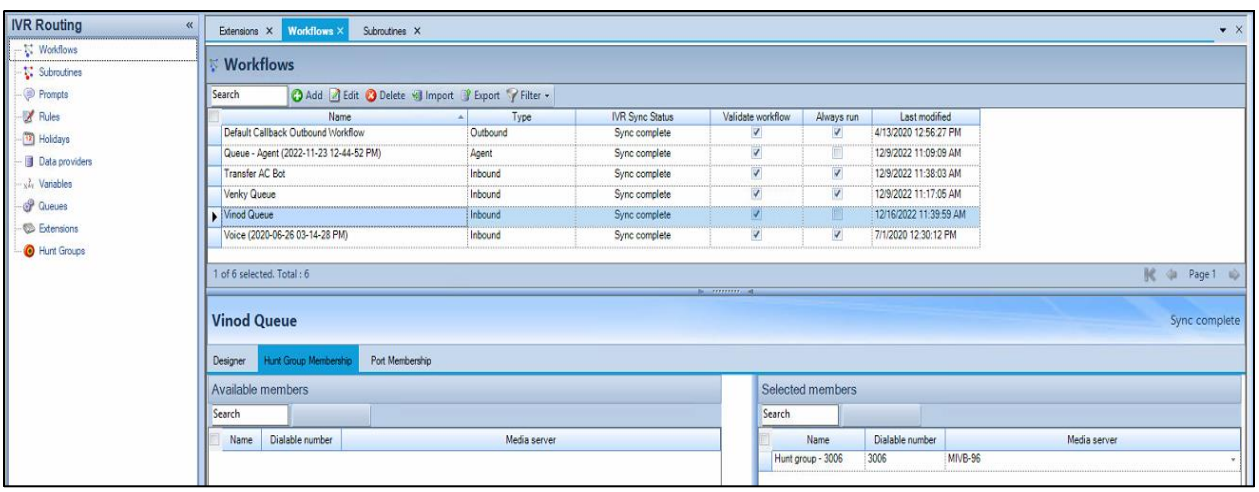


Figure 28 – Hunt Group Membership for Workflow

Workflow for calls to transfer to Agent

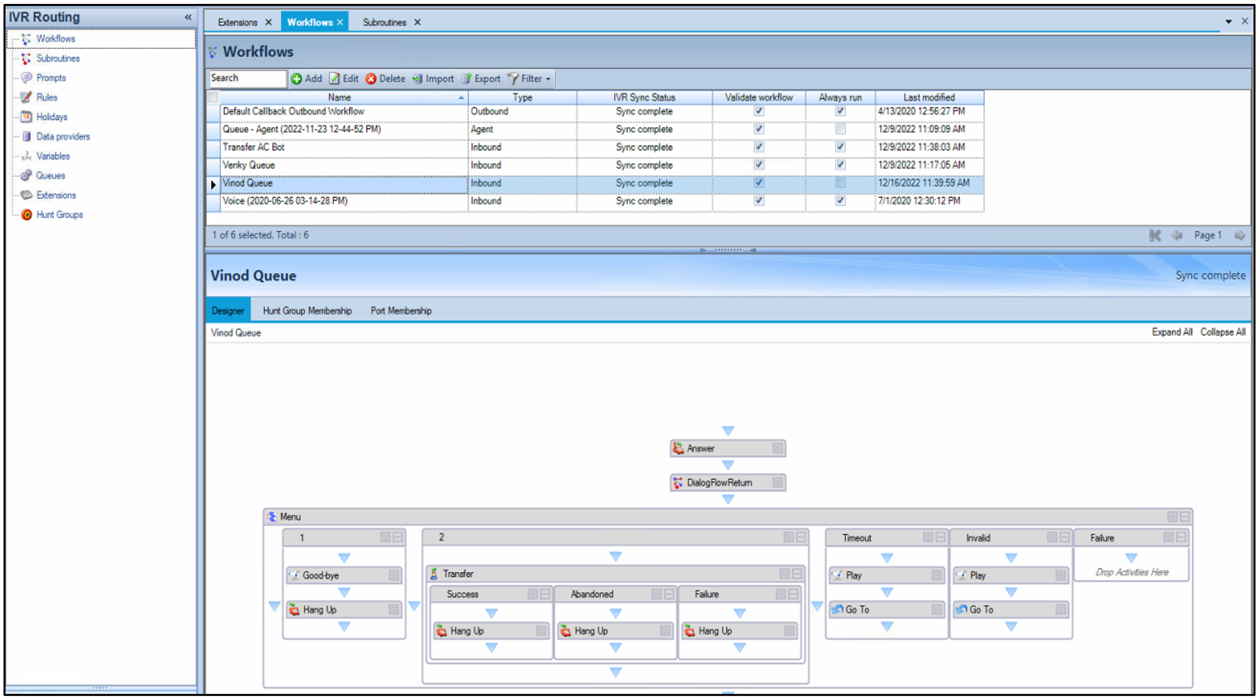


Figure 29 – Workflow

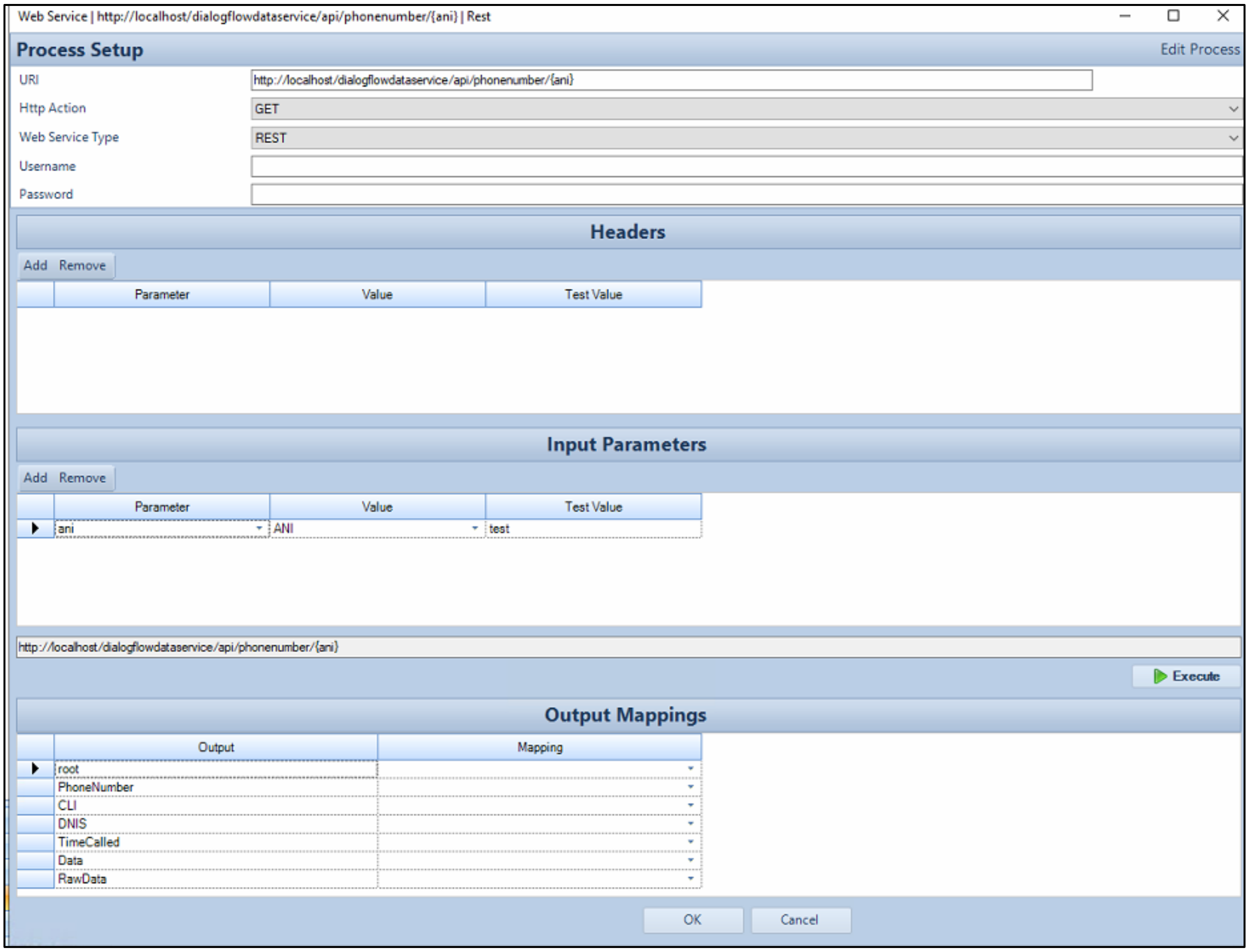


Figure 30 – API Configuration

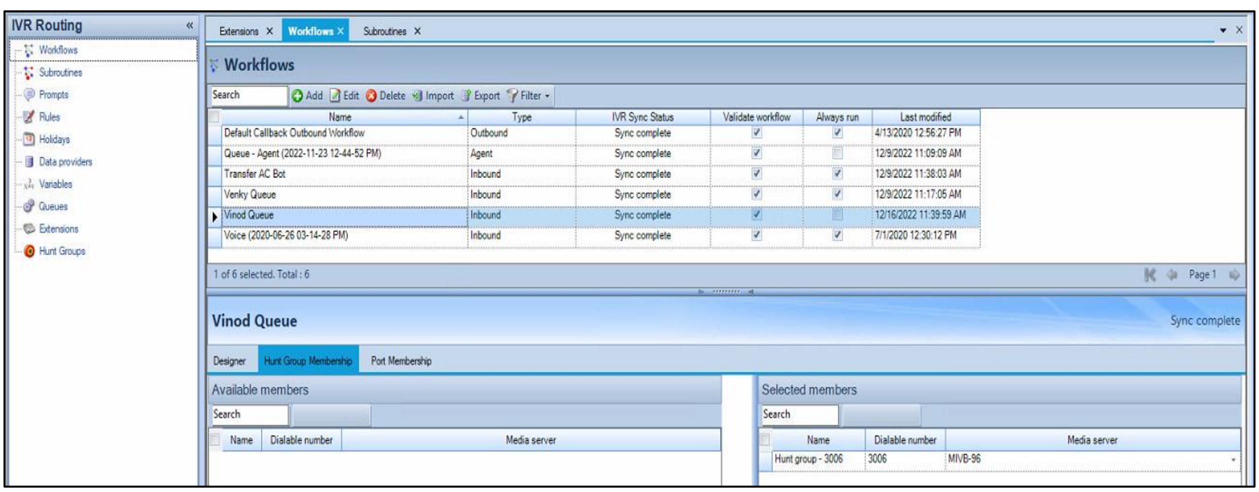


Figure 31 – Hunt Group Membership for Workflow

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)

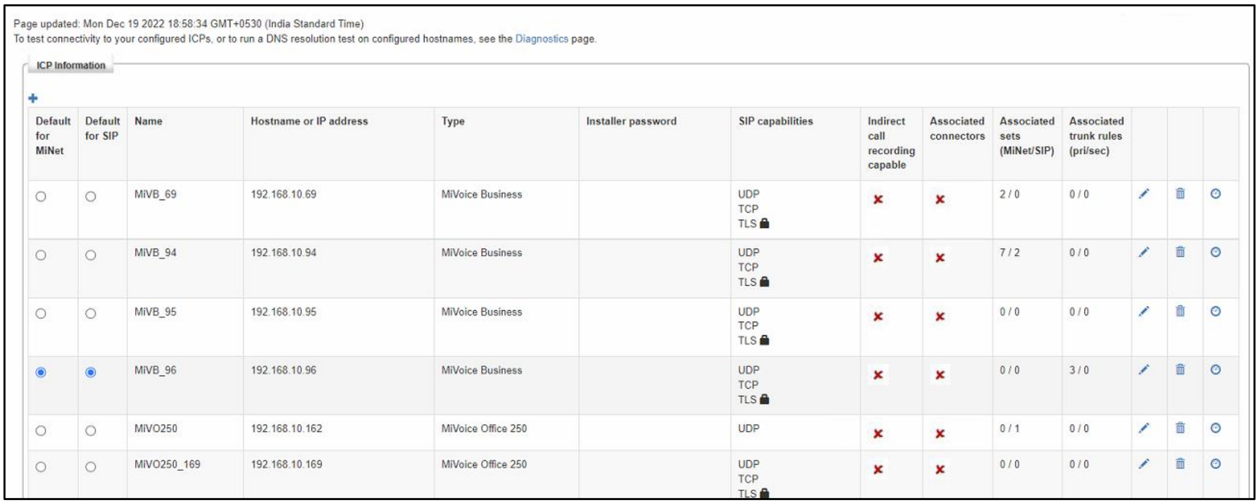


Figure 32 – MBG's Configuration page

- 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

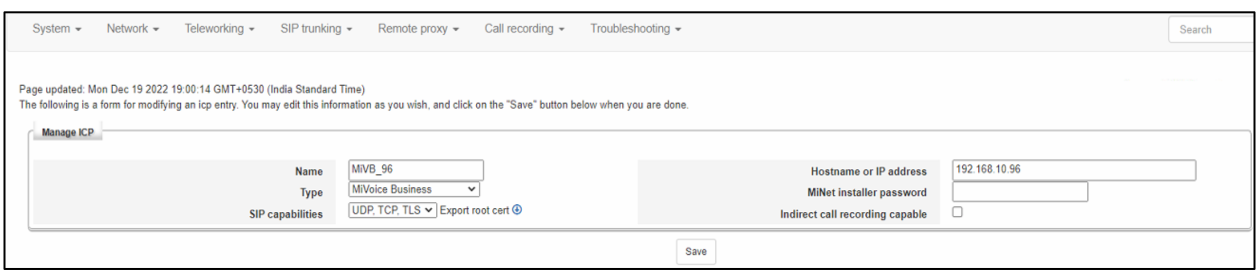


Figure 33 – ICP configuration page

- 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.

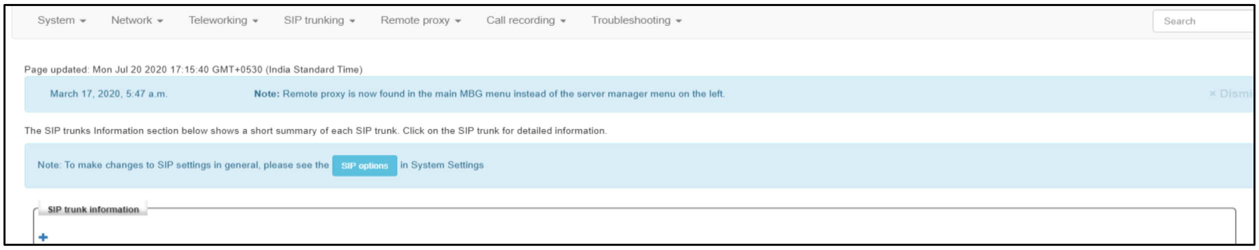


Figure 34: MBG SIP Trunking Configuration

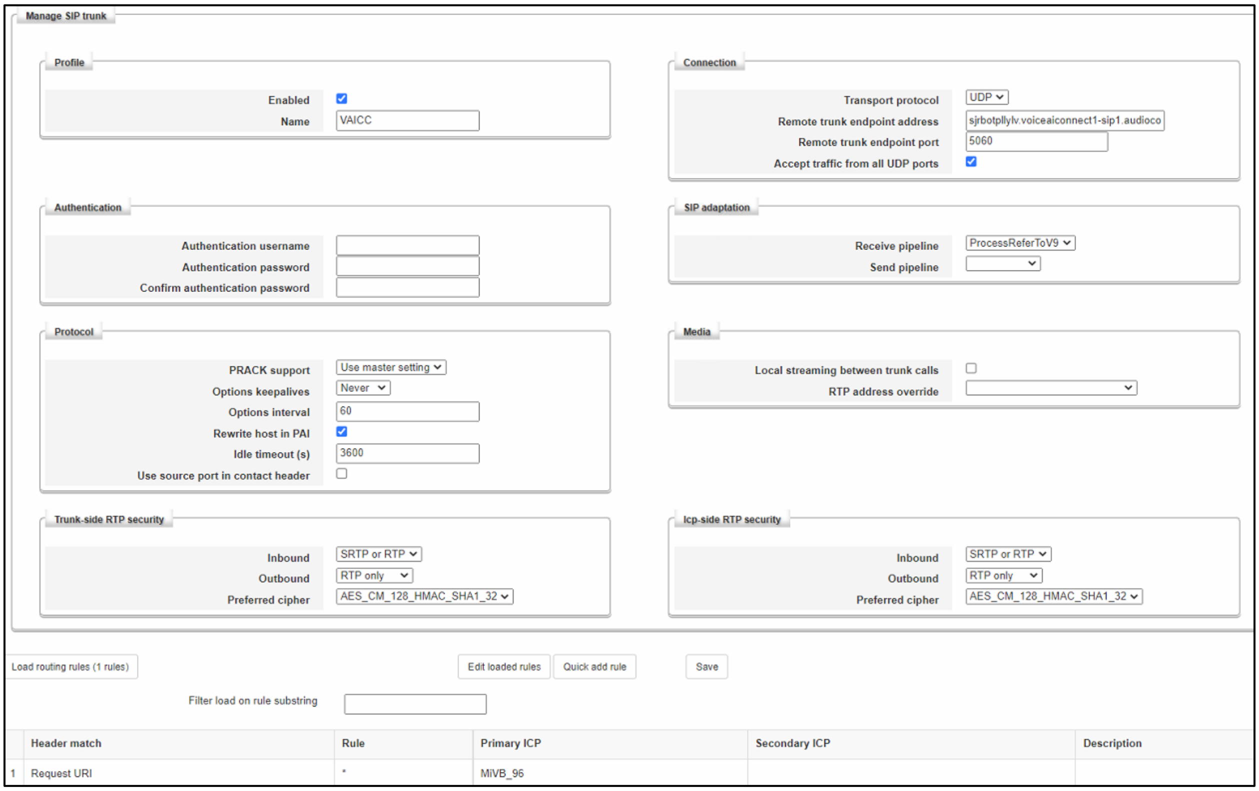


Figure 35: MBG SIP Trunking Configuration

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

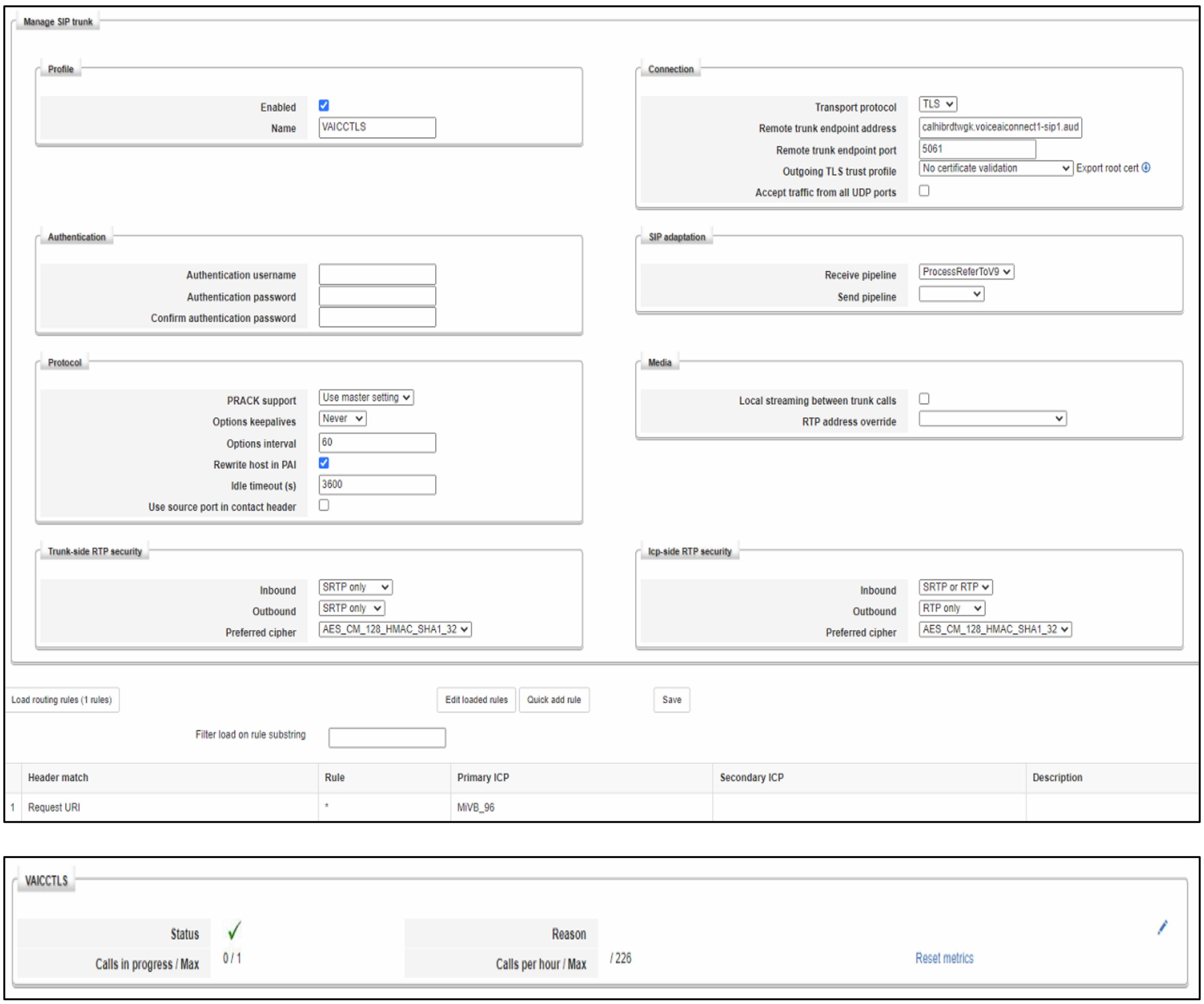


Figure 37: MBG SIP Trunking Configuration for TLS

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