Mitel Technical Configuration Notes – HO5356

September 4, 2024

Configure AudioCodes Mediant 1000 Gateway (common firmware) Lineside & Trunkside to use with MiVoice Business 10.1

Description: This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Business 10.1 with AudioCodes Mediant 1000 Gateway (Common Firmware) Lineside & Trunkside.

Environment: MiVoice Business 10.1 (10.1.1.21), MBG (12.0.2.132), MiCollab (9.8.1.108-01), AudioCodes Mediant 1000 Gateway (7.40A.501.150)

NOTICE

TRADEMARKS

Mitel Technical Configuration Notes – Configure AudioCodes Mediant 1000 Gateway (common firmware) Lineside & Trunkside for use with MiVoice Business 10.1.

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Overview

This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Business 10.1 to host the AudioCodes Mediant 1000 Gateway. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic AudioCodes Mediant 1000 Gateway setup as Endpoint with required options setup.

Interop History

Version	Date	Reason
1	September 2014	Refresh interop with MiVB 7.0 and AudioCodes Mediant 1000 Gateway SBC V 6.80A.231.002
2	March, 2019	Refresh interop with MiVoice Business 9.0 SP1 for use with AudioCodes Mediant 1000 Gateway SBC SW/v.7.20A.156.028
3	January, 2021	Interop with AudioCodes Mediant 1000 Gateway version (v.7.20A.258.271) and MiVoice Business 9.1 SP1 Both Trunkside and Lineside.
4	August, 2024	Interop with AudioCodes Mediant 1000 Gateway version (7.40A.501.150) and MiVoice Business 10.1 for both Trunkside and Lineside.

Interop Status

The Interop of the AudioCodes Mediant 1000 Gateway has been given a Certification status. This device will be included in the Mitel Interoperability Reference Guide (IRG). The status of AudioCodes Mediant 1000 Gateway achieved is:

COMPATIBLE	The most common certification which means the AudioCodes Mediant 1000 Gateway with MiVoice Business has been tested and/or validated by the Mitel Third-Party Interop Team. 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

The test setup generated basic SIP calls between the AudioCodes Mediant 1000 Gateway and the MiVoice Business 10.1.

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 Variant	Software Version	Additional Applicable Variants
Mitel	MiVoice Business	10.1 (10.1.1.21)	NA
Mitel	MBG (Teleworker)	12.0.2.132	NA
Mitel	68xx/69xx SIP	6.3.3.57	NA
Mitel	69xx MiNET	01.09.00.020	NA
Mitel	MiCollab Server	9.8.1.108-01	NA
AudioCodes	Mediant 1000 Gateway	7.40A.501.150	Mediant 500L/500/800

Tested Features

Listed below is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the Trunk side Interoperability Test Plans for detailed test cases.

Trunkside:

Feature	Feature Description	Issues
Basic Call	Making and receiving a call through the AudioCodes Mediant 1000 Gateway, call holding, transferring, conferencing, busy calls, long calls durations, variable codec.	☑
Nu-Point Voicemail	Terminating calls to a NuPoint voicemail boxes and DTMF detection.	X
Automatic Call Distribution	Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection.	☑
Packetization	Forcing the MiVoice Business to stream RTP packets through its E2T card at different intervals, from 10ms to 30ms	☑
Personal Ring Groups	Receiving calls through the AudioCodes Mediant 1000 Gateway to a personal ring group. Also moving calls to/from the prime member and group members.	☑
Teleworker	Making and receiving a call through the AudioCodes Mediant 1000 Gateway to and from Teleworker extensions.	☑
Video	Making and receiving a call through the AudioCodes Mediant 1000 Gateway with video capable devices.	X
Fax	G711 & T.38 Fax Calls.	☑
TLS/SRTP	Basic incoming/outgoing calls.	☑
Resiliency	Device able to handle resiliency when primary MiVB goes down.	☑

☑ *- No issues found* X *- Issues found, cannot recommend to use* ⚠ *- Issues found*

Lineside:

Listed below is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Line Side Interoperability Test Plans for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and Receiving calls.	☑
DTMF Signal	Sending DTMF after call setup (i.e. mailbox password)	☑
Registration/Authentication	Device registration w/o authentication	☑
Call Hold	Putting a call on hold	☑
Call Transfer	Transferring a call to another destination	☑
Call Forward	Forwarding a call to another destination	☑
Conference	Conferencing multiple calls together	☑
Redial	Last Number Redial	☑
Personal Ring Group	Multiple sets ringing when one number dialed	☑
Video	Video Capabilities	X
TLS/SRTP	Basic incoming/outgoing call. Teleworker incoming/outgoing call.	☑
Resiliency	Device able to handle resiliency when primary MiVB or MBG goes down.	☑
MWI	Message Waiting Indication	☑
G.711 and T.38 Fax	Fax Messages	☑

☑ *- No issues found* X *- Issues found, cannot recommend to use* ⚠ *- Issues found*

Resiliency

The following table lists the scenarios of resilience supported by this device when connected to the Mitel MiVoice Business 10.1.

Device	Basic	Advanced
AudioCodes Mediant 1000 Gateway	☑	⚠

☑ - *No issues found* ✗ - *Issues found, cannot recommend use* ⚠ - *Issues found*

Note: Refer to list of device limitations and known issues later in the document for recommendations.

The various scenarios are described below. The scenario names are a convenience for understanding this section of the configuration guide.

Basic: Resiliency is achieved by utilizing the ability of DNS servers to provide multiple IP addresses against a single FQDN. This is generally achieved by using DNS SRV or A records. This scenario requires nothing from a SIP Endpoint except that it supports standard DNS behavior. It can also be done by manually setting up back proxy on the phone.

Using REGISTER-301 Moved Permanently message to redirect registration to an alternate MiVoice Business element.

At a minimum, a 32-second timeout for the REGISTER, SUBSCRIBE, INVITE or OPTIONS messages should trigger a Failover

After Failover/Failback – the device must restart all subscriptions (message-summary

Advanced: There are different ways to detect the failure in this category.

P-Alternate-Server:
Use the P-Alternate-Server header in the REGISTER-200 OK message to store the HE and SE addresses.
Heartbeat
Use a light-weight heartbeat to periodically monitor the health of the MiVoice Business element to which the device is connected. This allows for the device to recover from failures faster without overloading the controlling element.
Survival Mode
Continue existing conversations when a failure is detected until at least the Session Timer expires or the user takes an action which causes termination. Displaying a message on the set is also recommended.
First Call after Failure
Implement a policy to time out a new call early if no 18x/2xx message is received.

Device Limitations

This is a list of problems or not supported features when the AudioCodes Mediant 1000 Gateway is connected to MiVoice Business 10.1.

Trunkside:

Feature	Problem Description
Outbound PSTN Call Privacy	During Outbound PSTN private call seeing Caller ID on PSTN side. MiVB is sending Anonymous but not sure whether AudioCodes is changing to Caller ID. May be some configuration details on AudioCodes. **Recommendation:** Please contact Audiocodes support for further details.
Call when other party is Busy	AudioCodes Mediant 1000 Gateway doesn't send busy tone to the caller. **Recommendation:** Contact AudioCodes support for more details.
TLS/SRTP	Force SDP and AVP only should be disabled under SIP peer profile in order to use TLS/SRTP with MiVB. **Recommendation:** Contact Mitel Support for more information.
Resiliency	Service provider resiliency feature is not tested in case of trunking environment. So, we have tested only MiVB resiliency. **Recommendation:** Since the analog trunk is connected to FXO port and created SIP trunk between AudioCodes Mediant 1000 Gateway and MiVB. So, the service provider resiliency is not applicable in this case. We have tested MiVB resiliency as a part of this testing.

Lineside:

Feature	Problem Description
TLS/SRTP	Force SDP and AVP only should be disabled under SIP Device Capability in order to use TLS/SRTP with MiVB. **Recommendation:** Contact Mitel Support for more information.
Basic Call	For basic call, when far end disconnects the call, The analog phone does not disconnect the call. So, India Re-order tones should be uploaded on AudioCodes Mediant 1000 Gateway in order to disconnect the call clearly. **Recommendation:** India Re-order tones should be uploaded on AudioCodes Mediant 1000 Gateway or Contact AudioCodes support for further clarifications on this.
Resiliency	In case of fallback scenario, Mediant 1000 sends invalid R-URI in REGISTER message to the priamry server when the primary server comes back online. Workaround – AudioCodes support has provided workaround with help of message maunipulation on Meidant 1000 in order to fix this issue. **Issue ID - SBC-25918** **Recommendation:** Contact AudioCodes support for further clarifications on this.

Network Topology

This diagram shows how the **Trunk** testing network is configured for reference.

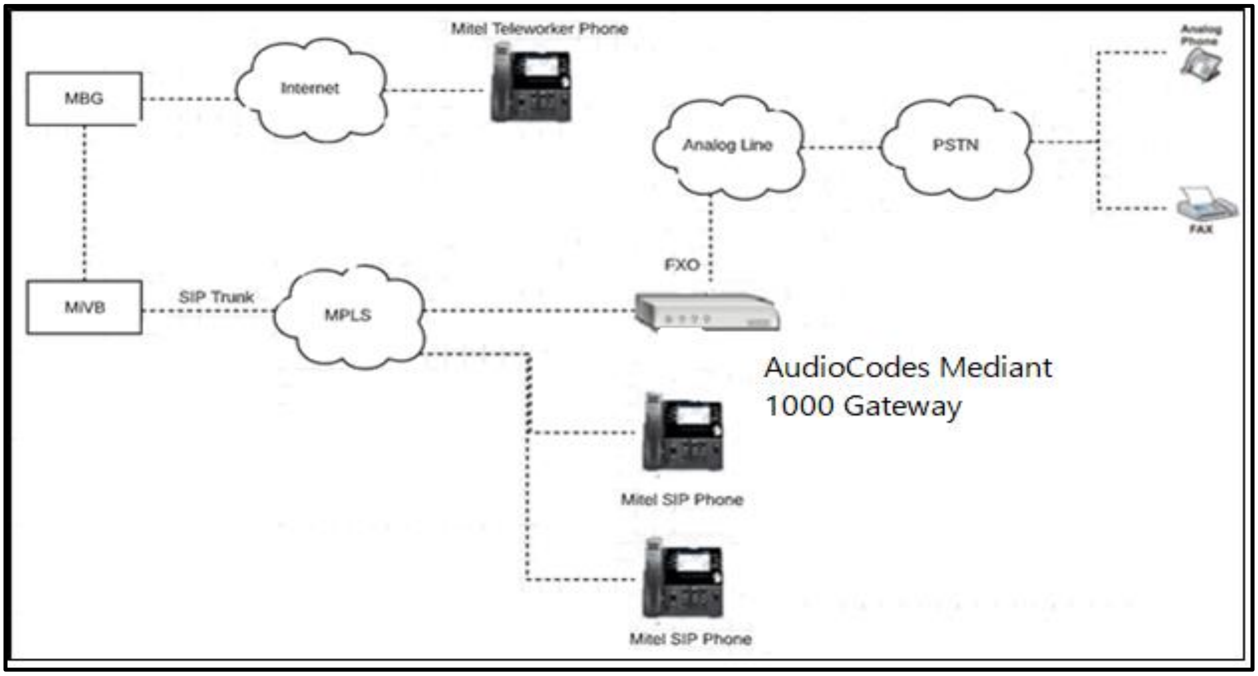


Figure 1 – Network Topology for Trunkside

This diagram shows how the **Line** testing network is configured for reference.

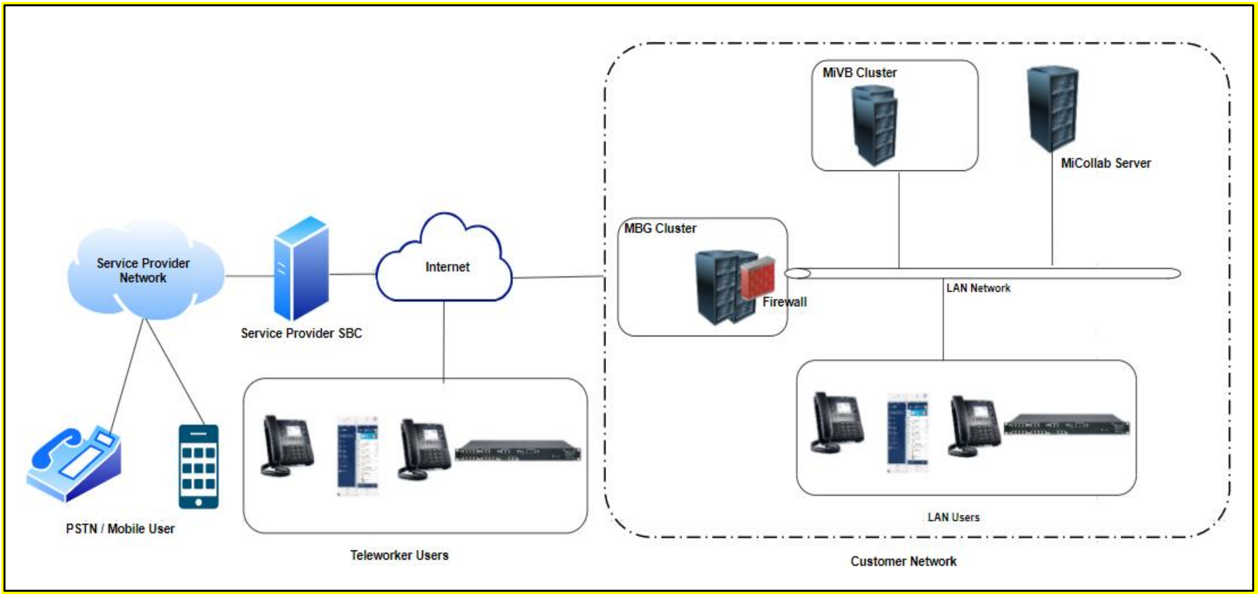


Figure 2 – Network Topology for Lineside

MiVoice Business - Configuration Notes

The following steps show how to program a MiVoice Business to connect with the AudioCodes Mediant 1000 Gateway.

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. 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 for G729. 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 the MiVoice Business Programming

The SIP signaling connection uses UDP on Port 5060.

Trunkside Configuration

Licensing and Option Selection – SIP Licensing

Ensure that the MiVB is equipped with enough SIP trunking licenses for the connection of AudioCodes Mediant 1000 Gateway. 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 service providers, applications, and SIP trunking devices.

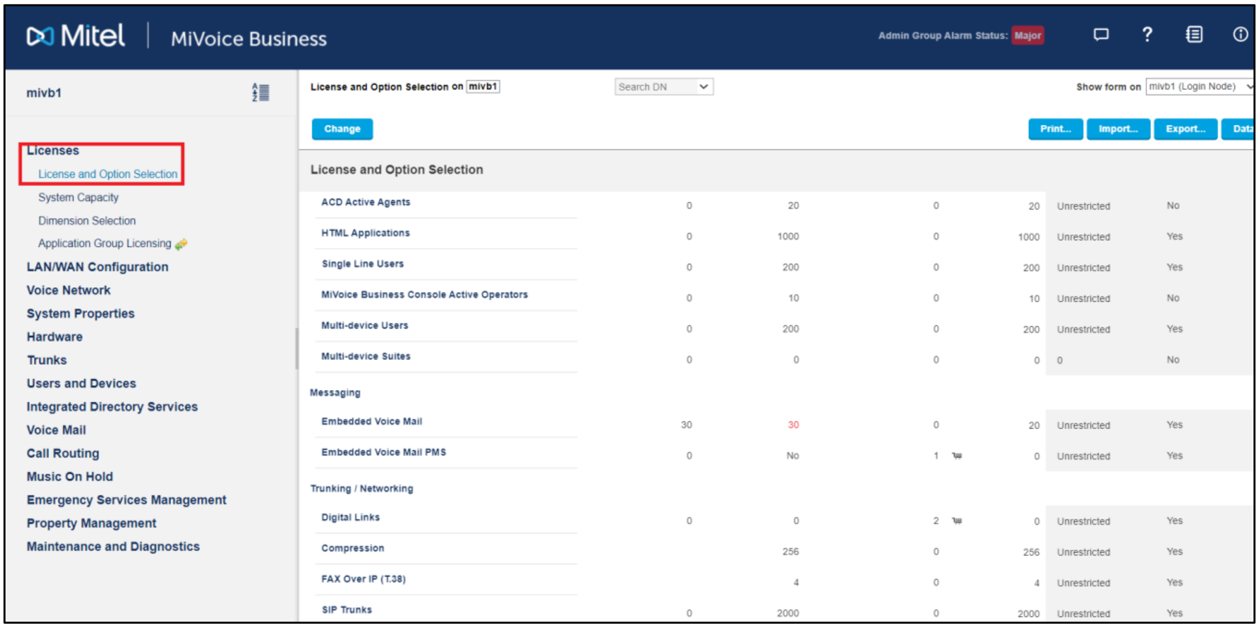


Figure 3 – License and Option Selection

Class of Service Assignment

The Class of Service Options form is used to create or edit the 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

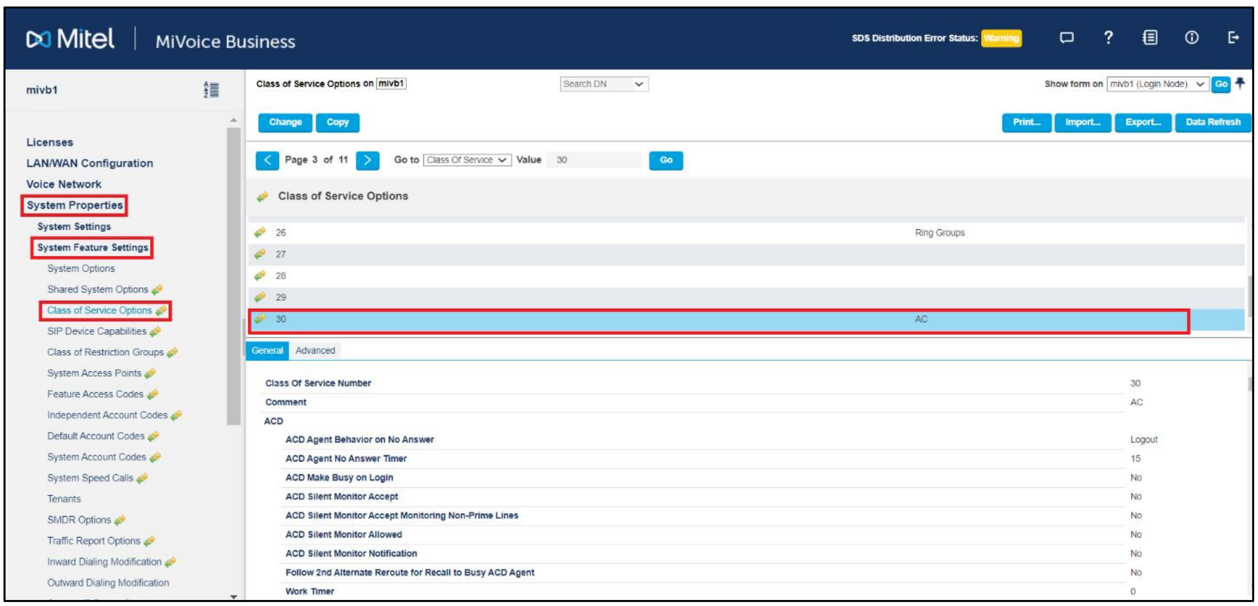


Figure 4 – Class of Service Options

Network Element Assignment

Create a network element for AudioCodes Mediant 1000 Gateway. In this example, the soft switch is reachable by an IP Address and is defined as 'Mediant' in the network element assignment form.

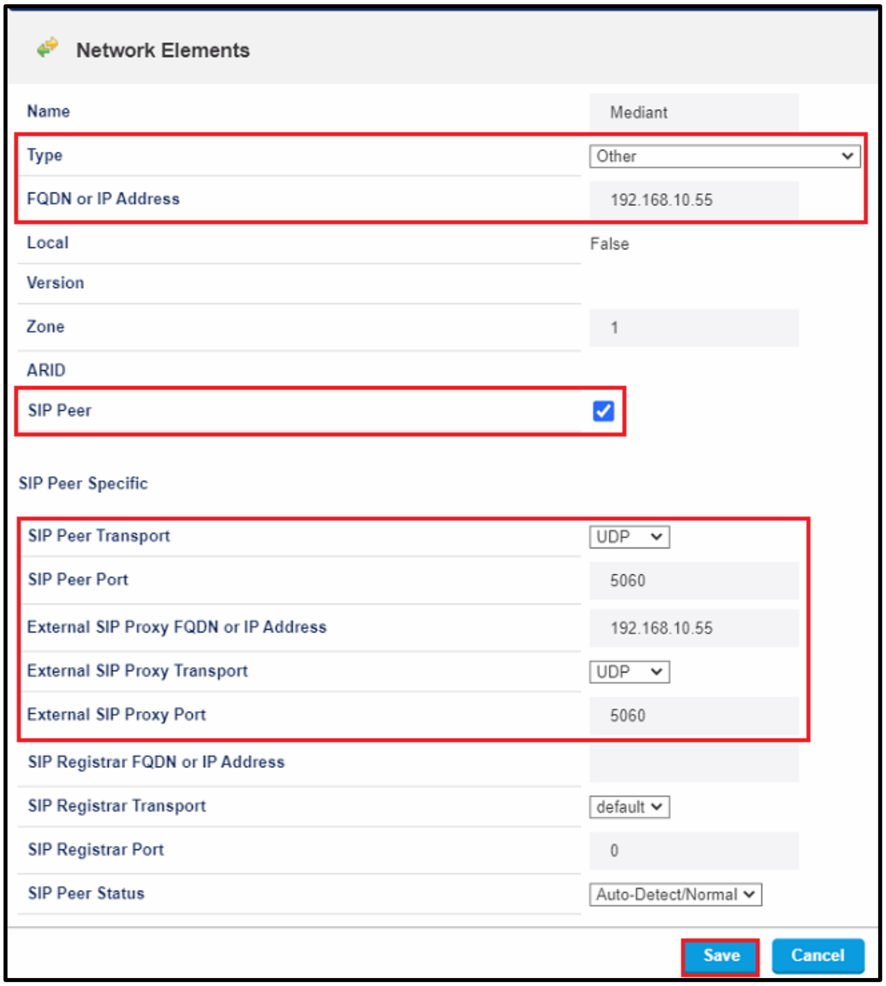


Figure 5 – Network Elements Assignment

Trunk Attributes

This is configured in the Trunk Attributes form. In this example the Trunk Attributes is defined for Trunk Service Number **7**, 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 service provider.

The example below shows configuration for incoming DID calls. The MiVB will absorb the first 0 digits of the number from AudioCodes Mediant 1000 Gateway. Please refer to the MiVB System Administration documentation for further programming information.

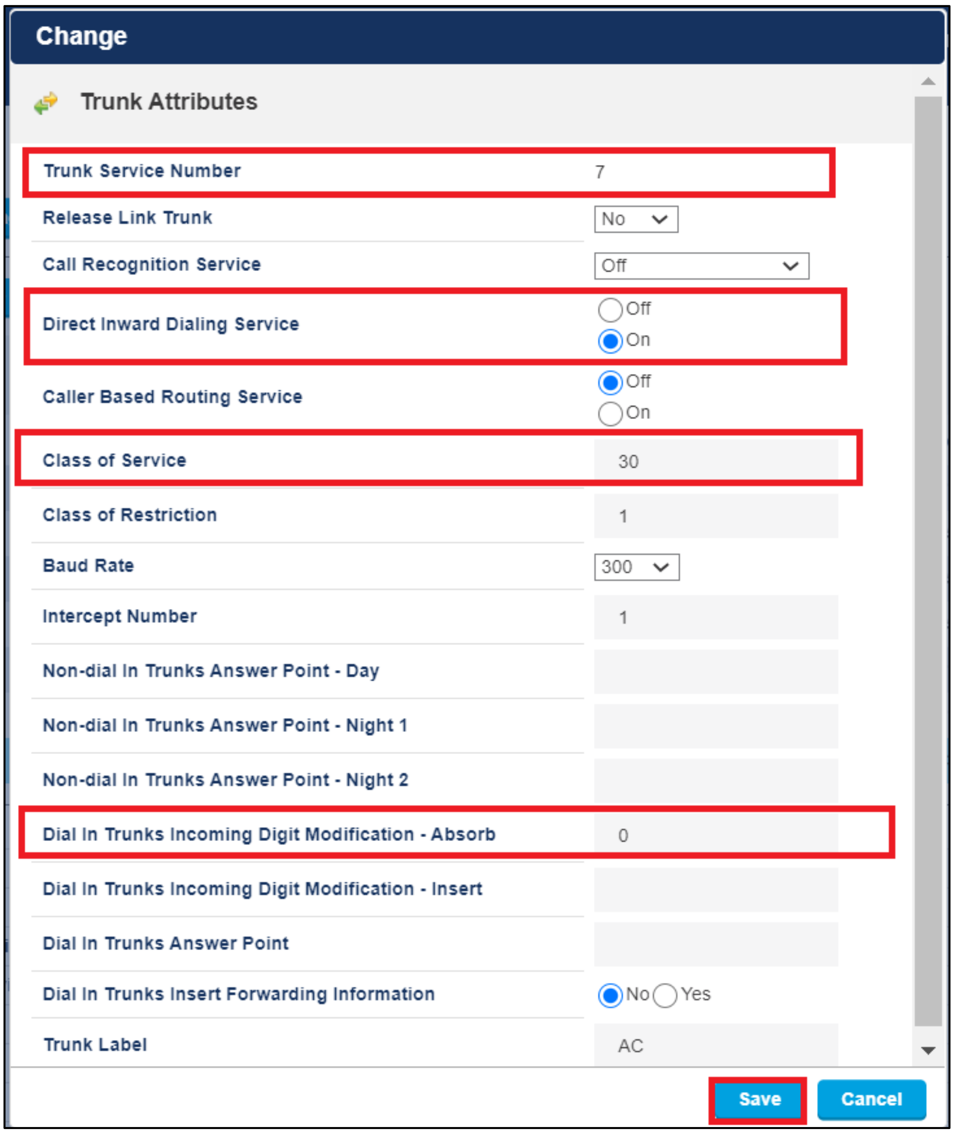


Figure 6 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base MiVoice Business 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 “Mediant” Network Element.

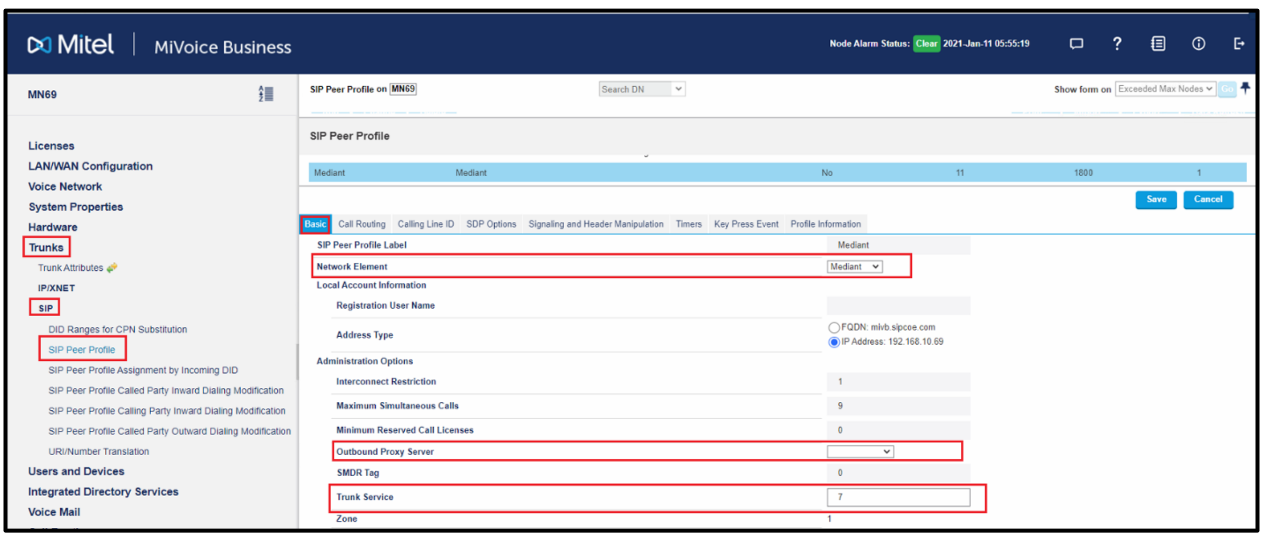
Address Type: Select IP.

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. 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 Mediant 1000 Gateway.



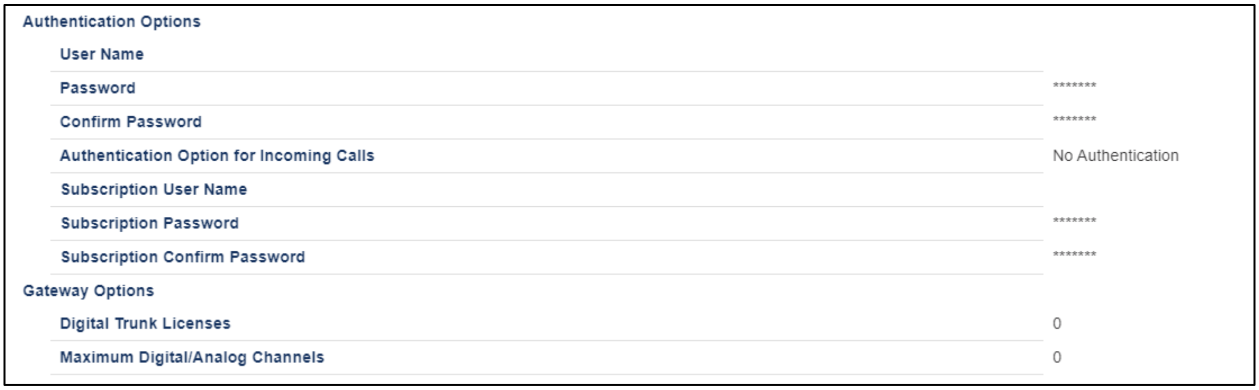


Figure 7 - SIP Peer Profile Assignment- Basic

Figure 8 - SIP Peer Profile Assignment- Call Routing

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	
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
Signal Privacy (if enabled) on Emergency Calls	No
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-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

Figure 11 – 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 12 – 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 13 – 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 14 – SIP Peer Profile Assignment – Outgoing DID Ranges

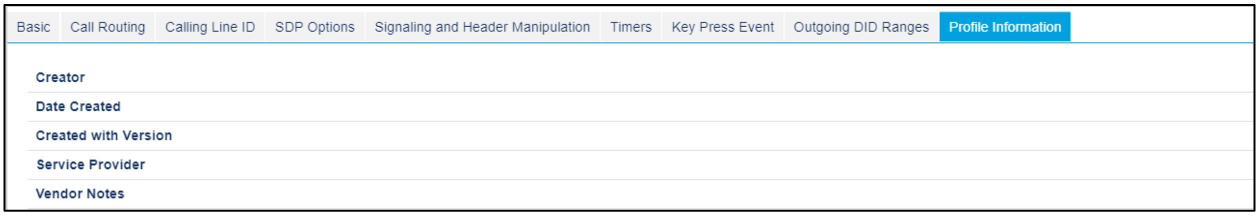


Figure 15 – SIP Peer Profile Assignment – Profile Information

SIP Peer Profile Assignment by Incoming DID

This form is used to associate DID range numbers from AudioCodes Mediant 1000 Gateway Analog trunk to a particular SIP Peer profile. The configured settings here help matching the incoming DID numbers with the SIP Peer Profile when call is arriving from anonymous caller.

Enter one or more telephone numbers. The maximum number of digits per telephone number is 26. You can enter a mix of ranges and single numbers (for example, " 008067591215"). The entire field width is limited to 60 characters.

Use a comma to separate telephone numbers and ranges. Use a dash (-) to indicate a range of telephone numbers. The first and last characters cannot be a comma or a dash. If the numbers do not fit within the 60 characters maximum, you can create a new entry for the same profile.

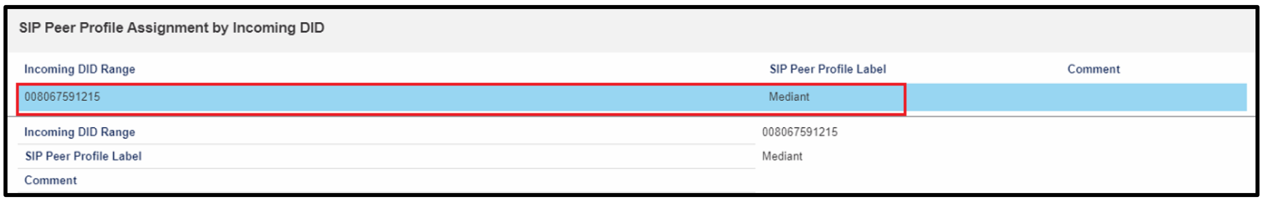


Figure 16 – SIP Peer Profile Assignment by Incoming DID

ARS Digit Modification Plans

Ensure that Digit Modification for outgoing calls on the SIP trunk to AudioCodes Mediant 1000 Gateway absorbs or inject additional digits according to your dialling plan. In this example, we will be absorbing 3 digits (in this case will be **456** to dial out).

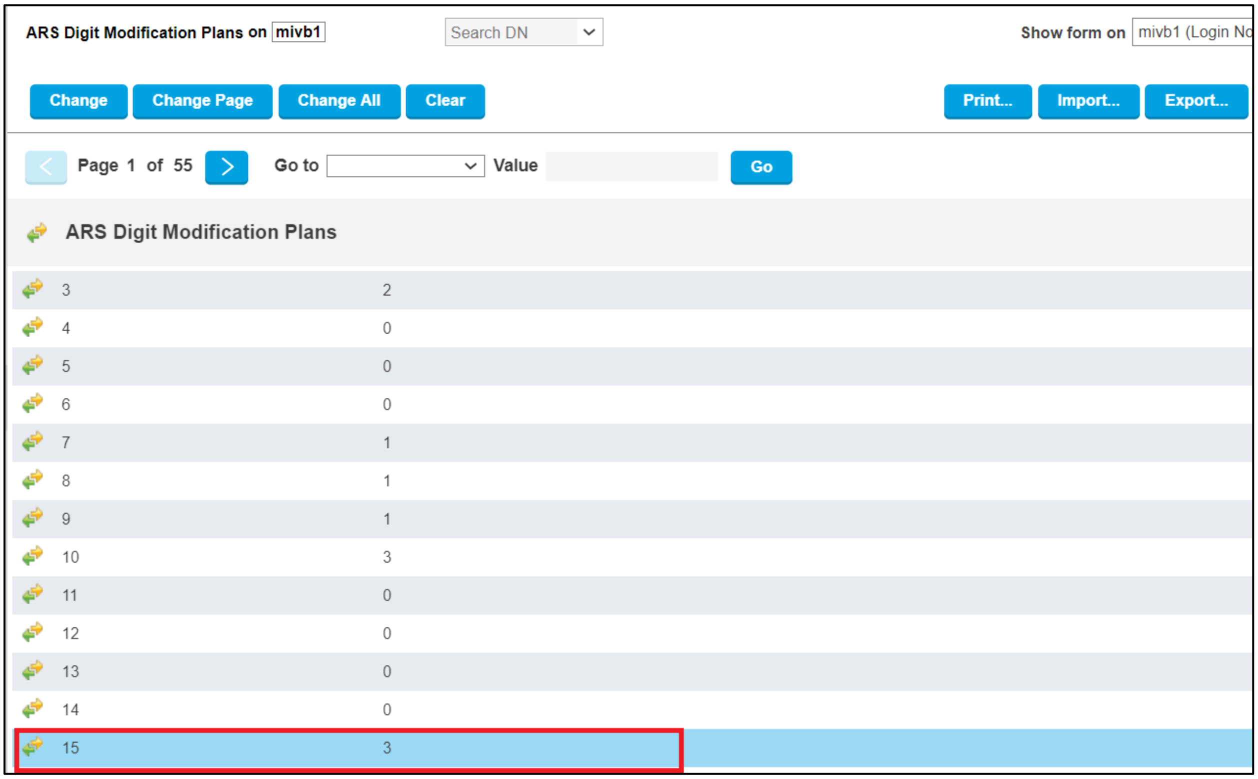


Figure 17 – Digit Modification Assignment

ARS Routes

Create a route for SIP Trunks connecting a trunk to AudioCodes Mediant 1000 Gateway. In this example, the SIP trunk is assigned to Route Number **15.** Choose SIP Trunk as a routing medium and choose the SIP Peer Profile and Digit Modification entry created earlier.

ARS Routes

Route Number	Routing Medium	Trunk Group Number	SIP Peer Profile	PBX Number / Cluster Element ID	COR Group Number	Digit Modification Number	Digits Before Outpulsing	Route Type	Compression
1	SIP Trunk		VMBA		1	3		PSTN Access Via DPNSS	Off
2	SIP Trunk		VMBB		1	3		PSTN Access Via DPNSS	Off
3	SIP Trunk		VMBA		1	4		Emergency	Off
4	SIP Trunk		AC		1	1		PSTN Access Via DPNSS	Off
5	SIP Trunk		DTAG		1	6		PSTN Access Via DPNSS	Off
6	SIP Trunk		Rev2		1	9			Off
7	SIP Trunk		Drei		1	2		PSTN Access Via DPNSS	Off
8	SIP Trunk		Revolutio		1	9			Off
9	Direct IP Route			74	65	805			Auto
10	SIP Trunk		EH		1	3		PSTN Access Via DPNSS	Off
11	SIP Trunk		AC		1	1		PSTN Access Via DPNSS	Off
12	SIP Trunk		Level3		1	12		PSTN Access Via DPNSS	Off
13	SIP Trunk		testmbg		1	8		PSTN Access Via DPNSS	Off
14	SIP Trunk		Rev2		1	9		Emergency	Off
15	SIP Trunk		Mediant		1	9		Emergency	Off

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 **456,** the call will be routed to AudioCodes Mediant 1000 Gateway (ie. **Route 15**).

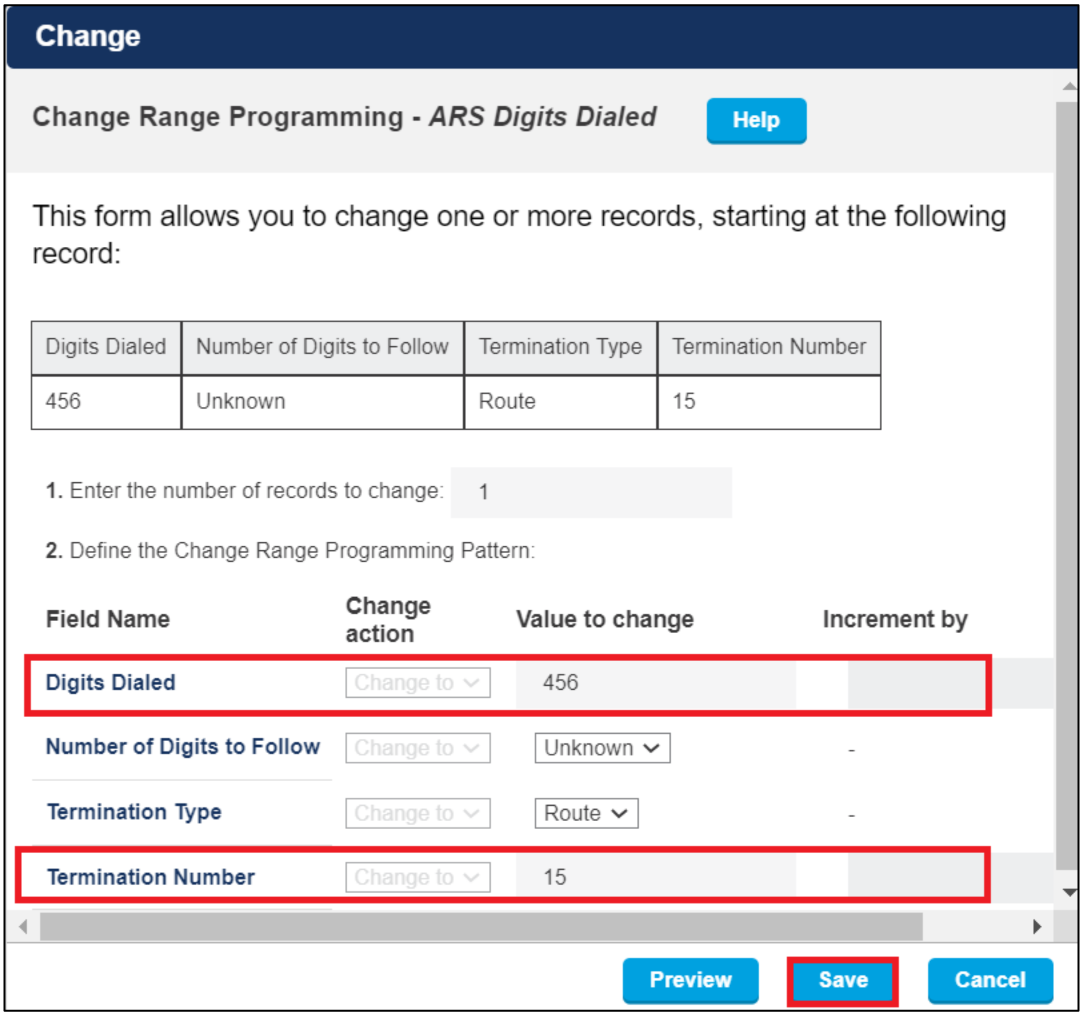


Figure 19 – ARS Digit Dialed Assignment

T.38 Fax Configuration

AudioCodes Mediant 1000 Gateway uses the inter-zone FAX profile. This form allows you to define the settings for FAX communication over the IP network. You can modify the default settings for the:

- **Inter-zone FAX profile**: defines the FAX settings between different zones in the network. There is only one Inter-zone FAX profile; it applies to all inter-zone FAX communication. It defaults to V.29, 7200bps. It defines the settings for FAX Relay (T.38) FAX communication.
- **Intra-zone FAX profile**: defines the FAX settings within each zone in the network.
  - Profile 1 defines the settings for G.711 pass through communication.
  - Profile 2 to 64 define the settings for FAX Relay (T.38) FAX communication.
  - All zones default to G.711 pass through communication (Profile 1).

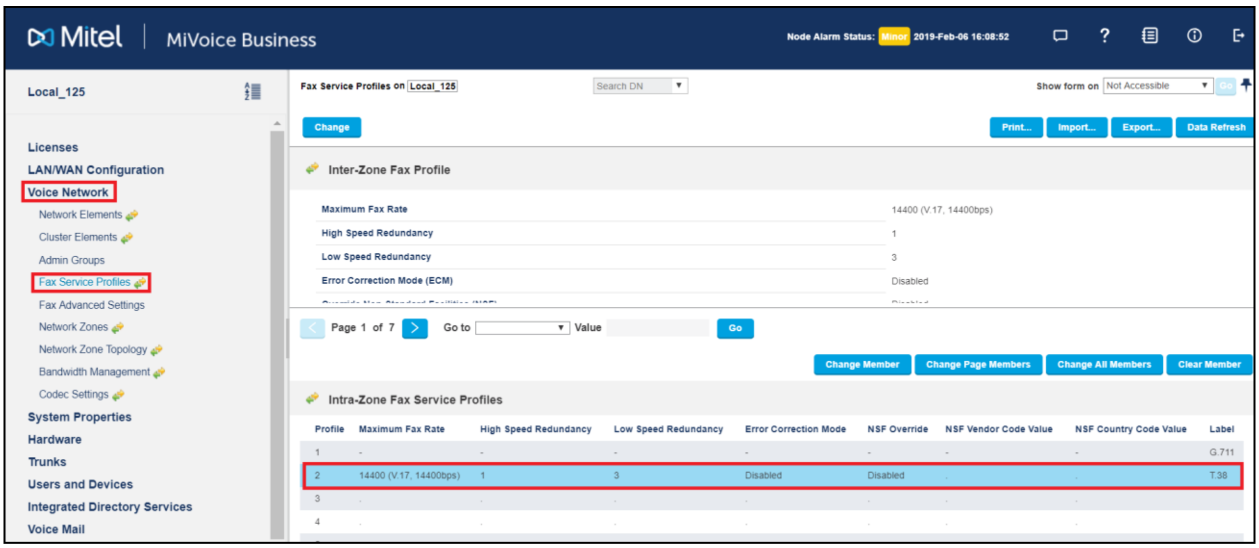


Figure 20 - Fax Configuration

Zone Assignment

By default, all zones are set to Intra-zone FAX Profile 1.

Based on your network diagram, assign the Intra-zone FAX Profiles to the Zone IDs of the zones. If audio compression is required within the same zone, set Intra-Zone Compression to “Yes”. AudioCodes Mediant 1000 Gateway uses the Intra-zone FAX Profile 2.

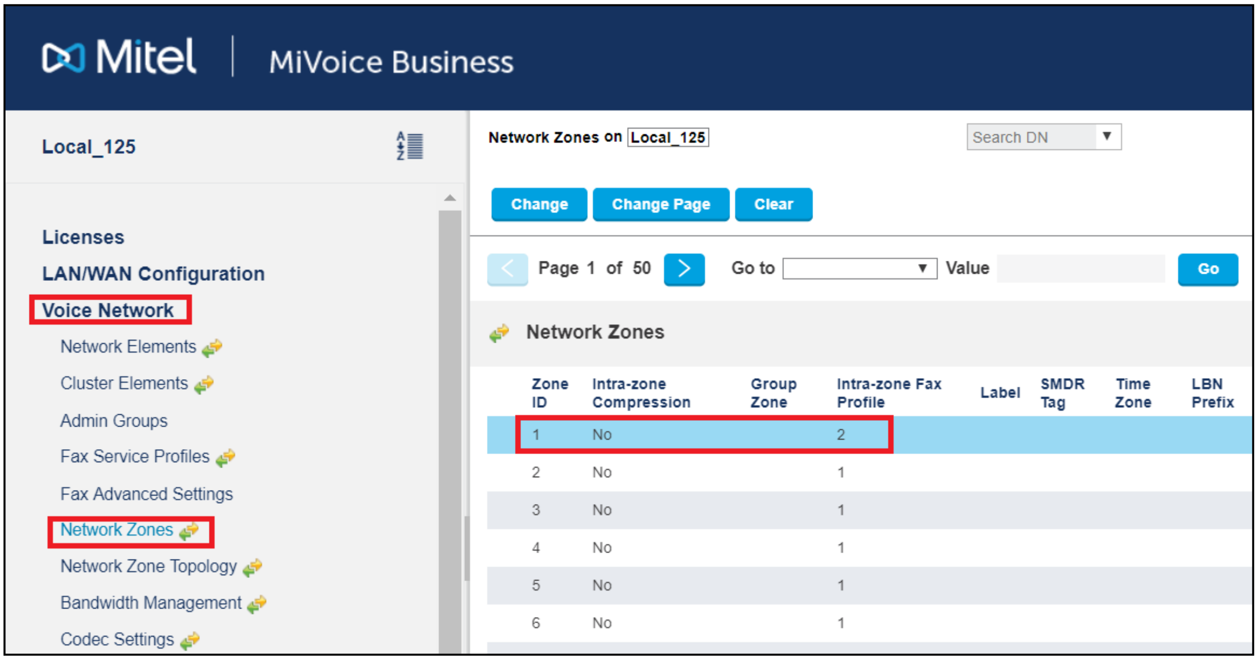


Figure 21 – Zone Assignment

Lineside Configuration

Software License – SIP Licensing

Ensure that the MiVoice Business is equipped with enough Mode licenses for the connection of SIP end points (AudioCodes Mediant 1000 Gateway). This can be verified within the Software License Feature section form.

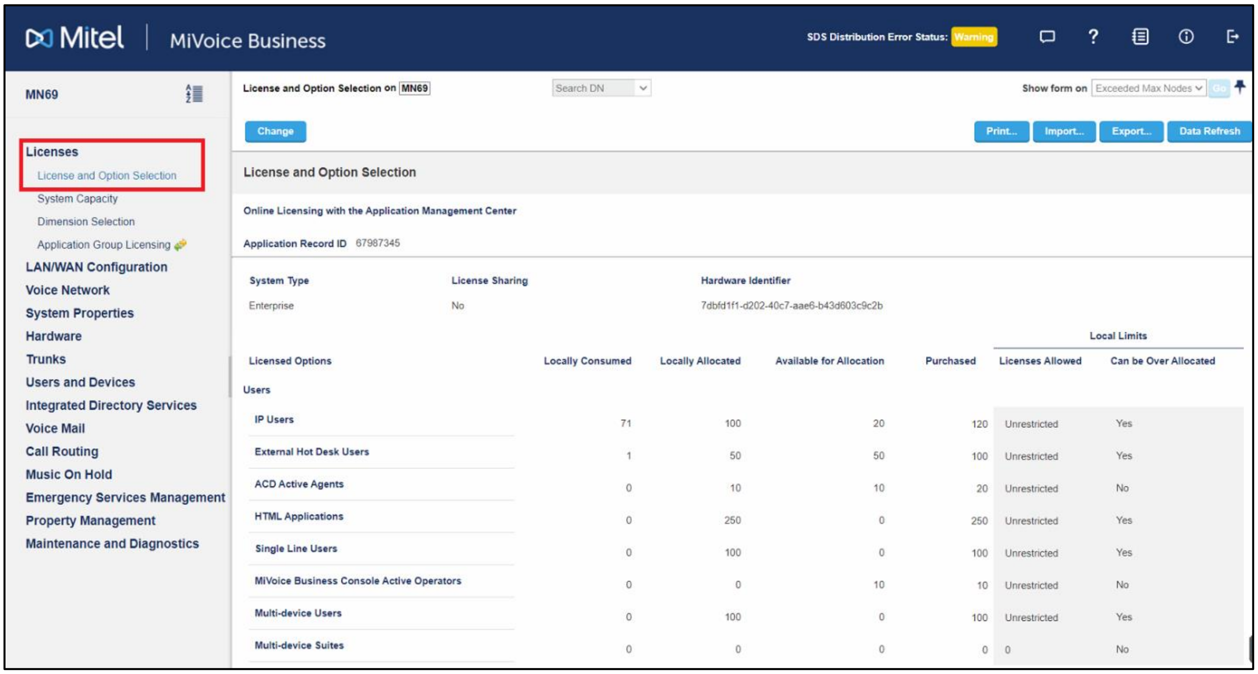


Figure 22 – Software License

Multiline IP Set Configuration

On the MiVoice Business, a SIP device can be programmed either in the User Configuration form or the Multiline IP Set Configuration form and are programmed as a “Generic SIP Phone”.

The User PIN is the SIP authentication password and the Number is the Directory Number (DN is a telephone number). All other field names should be programmed according to the site requirements or left at default.

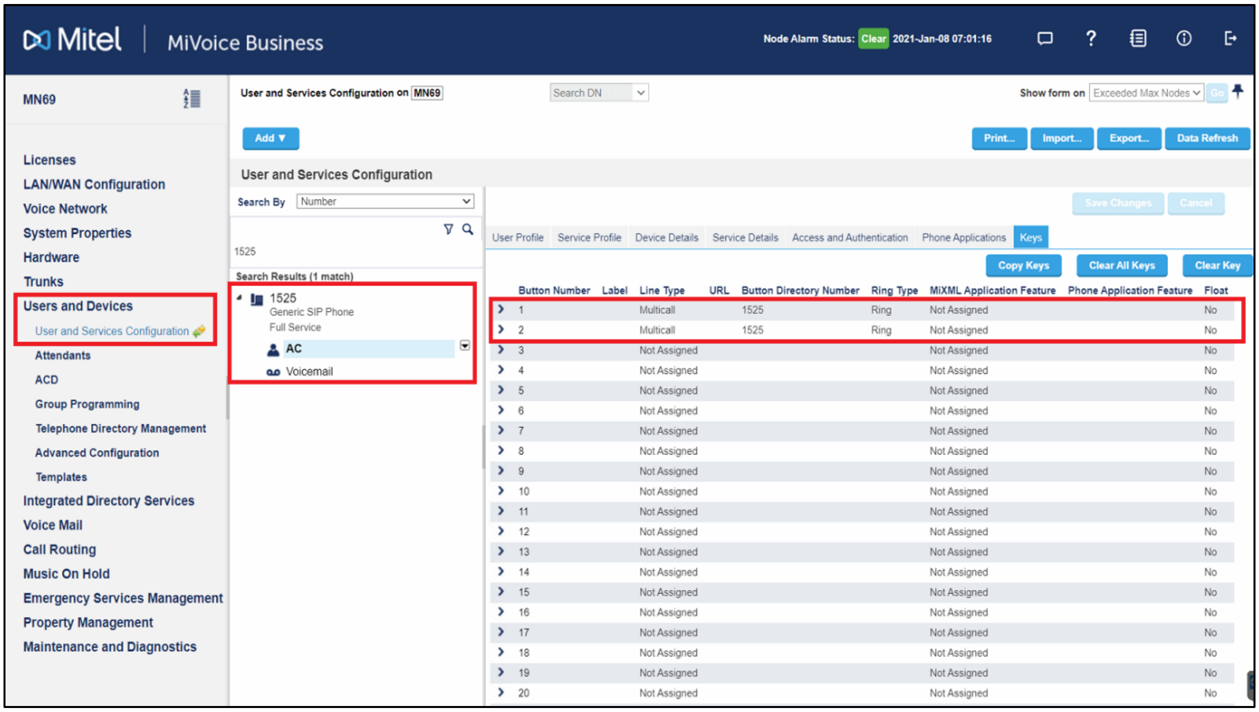
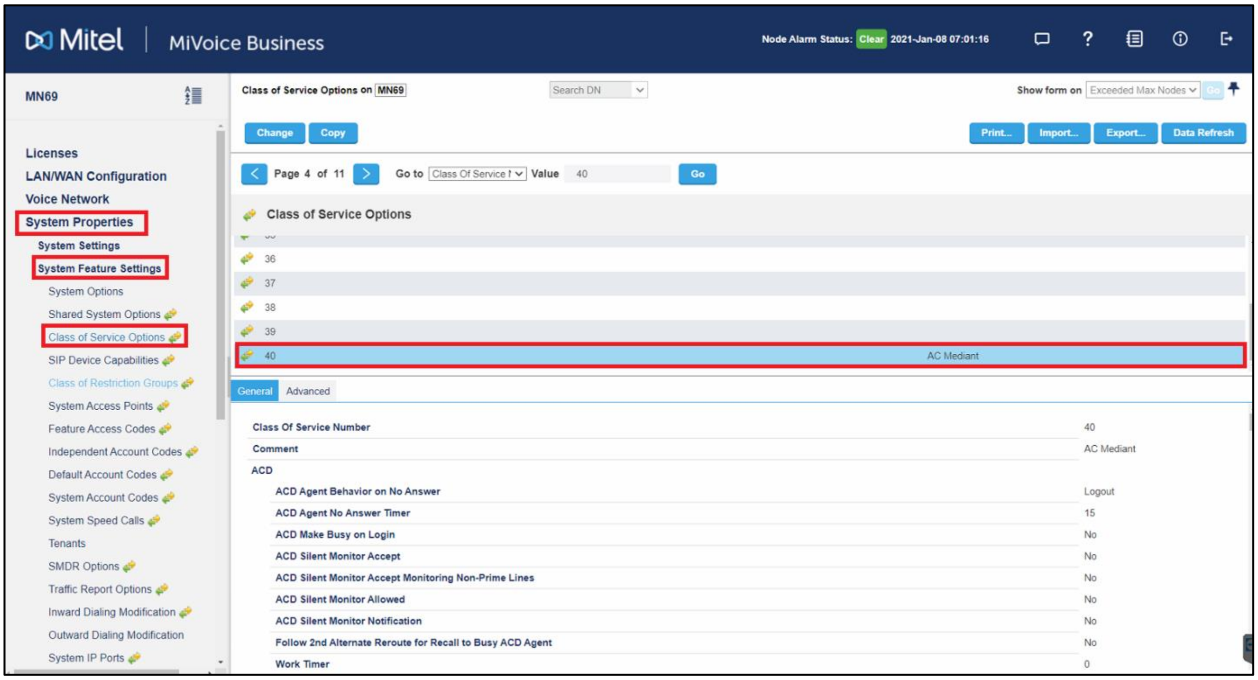


Figure 23 – Create SIP Extension

Class of Service Assignment

The Class of Service Options form is used to create or edit the Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced by the Station Attributes form for the SIP device.

Many different options may be required for your site deployment, but the options below are required to be changed from the default for a Generic SIP Device to work with the 3300 ICP.



General | Advanced

Off-Hook Voice Announce Allowed	No
Handsfree AnswerBack 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
Lockout Timer	45
Call Duration	
Call Duration	10
Call Duration Forced Cleardown Timer	0
Enable Call Duration Limit on External Calls	No
Enable Call Duration Limit on Internal Calls	No
Call Forwarding/Rerouting	
Call Forward - Delay	0
Call Forward No Answer Timer	15
Call Forward Override	No

General | Advanced

Call Forwarding (External Destination)	No
Call Forwarding (Internal Destination)	Yes
Call Forwarding Accept	Yes
Call Reroute after CFFM to Busy Destination	No
Call Forwarding Reminder Ring (CFFM and CFIAH only)	No
Disable Call Reroute Chaining On Diversion	No
Follow Reroute on Disabled Forwarding	No
Group Call Forward Follow Me Accept	No
Group Call Forward Follow Me Allow	No
Third Party Call Forward Follow Me Accept	No
Third Party Call Forward Follow Me Allow	No
Use Held Party Device for Call Re-routing	Yes

General	Advanced
Call Hold	
Call Hold	Yes
Call Hold - Retrieve with Hold Key	No
Call Hold Remote Retrieve	Yes
Call Hold Timer	30
Local Music On Hold source	No
Music on Hold on Transfer	No
Use Called Party Call Hold Timer	No
Call Park	
Call Park Timer	180
Call Park-Allowed To Park	Yes
Call Pickup	
Allow Directed Call Pickup Of Attendant Call	No
Call Pickup Dialed Accept	Yes
Call Pickup Directed Accept	Yes
Call Pickup Display	No
Call Privacy	
Call Privacy	No
Calling Party Name Substitution	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No

General	Advanced
Public Network Identity Provided	No
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes
Campon	
Auto Campon Timer	
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No
Display	
After Answer Display Time	
Calling Name Display - Internal - ONS	Yes
Calling Number Display - Internal - ONS	Yes
Display ANI/DNIS/ISDN Calling/Called Number	No
Display ANI/ISDN Calling Number Only	No
Display Caller ID on multicall/keylines	No
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	No
Display Dialed Digits during Outgoing Calls	No

General | Advanced

Display DNIS/Called Number Before Digit Modification	No
Display DNIS on Key Label	No
Display Held Call ID on Transfer	No
Display Transfer Destination on Recall	No
Hot Desk External User - Display Internal Calling ID	No
Maintain Ringing Party During Recall	No
Non-Prime Public Network Identity	No
Originator's Display Update In Call Forwarding/Rerouting	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	No
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No
Fax	
Campon Tone Security	Yes
External Trunk Standard Ringback	No
Fax Capable	Yes
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes

General | Advanced

HCI/CTI/TAPI Monitor Allowed	Yes
Hot Desk	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk Auto Logout Timer	0
Hot Desk External User - Allow DTMF Dialing	Yes
Hot Desk External User - Allow Mid-Call Features	Yes
Hot Desk External User - Answer Confirmation	Yes
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	No
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No
Miscellaneous	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Enable Device Configuration	0
Enbloc Dialing - Enabled	No
Force Device Busy If Any Line In Use	No
Handset Volume Adjustment Saved	No
Headset Switch Mute	No

General | Advanced

Headset Play In-Band Ring Burst	No
Integrated DECT High Power - Enabled	Yes
Integrated DECT Wideband - Enabled	Yes
Multi-Color LED Support - Disable	No
Phone Lock	No
Reseize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0
Paging	
Group Page Accept	No
Group Page Allow	No
Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes
Pager Access All Zones	Yes
Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10

General | Advanced

Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
Ringing	
Allow Recall after Transfer	No
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	180
SMDR	
SMDR External	No
SMDR Internal	No
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	No
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes

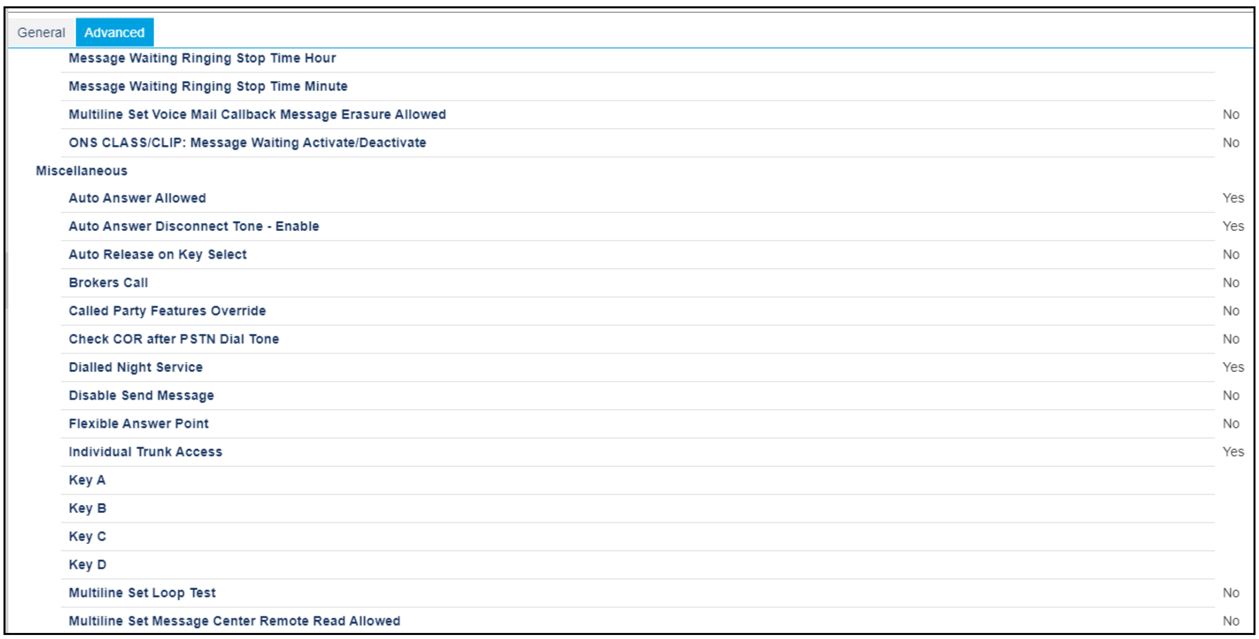
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	No
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	No
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5

General | Advanced

Account Code	
Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes
Attendant	
Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No
Call Screening	
BLF Screening Allow	No
BLF Screening Accept	No
Conference	
Conference Call	Yes
Disable Conference Join Tone	No
DND	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No

General | Advanced

Emergency	
Emergency Call - Audio Level for Set	Ringer
Emergency Call Notification - Audio	No
Emergency Call Notification - Visual	No
Group Presence	
Group Presence Control	No
Group Presence Third Party Control	No
Hotel	
Display VIP	No
Hotel Room Monitor Setup Allowed	No
Hotel Room Monitoring Allowed	No
Hotel/Motel Room Personal Wakeup Call Allowed	No
Hotel/Motel Room Remote Wakeup Call Allowed	No
Message Waiting	
Message Waiting	Yes
Message Waiting - Disable Ringing Lamp Notification	No
Message Waiting Audible Tone Notification	No
Message Waiting Deactivate On Off-Hook	Yes
Message Waiting Inquire	Yes
Message Waiting Ringing Start Time Hour	
Message Waiting Ringing Start Time Minute	



General | Advanced

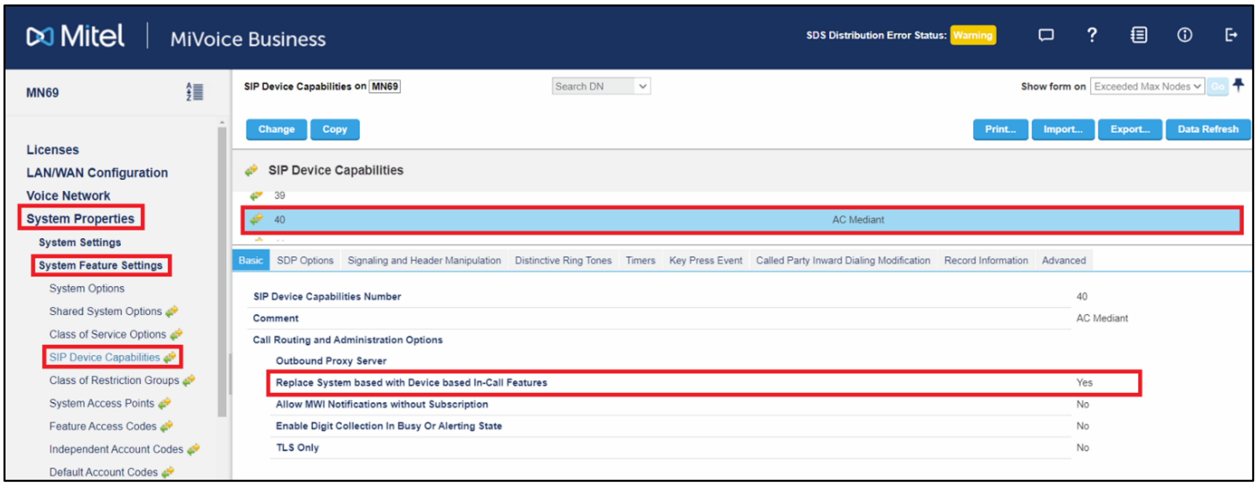
Message Waiting Ringing Stop Time Hour	
Message Waiting Ringing Stop Time Minute	
Multiline Set Voice Mail Callback Message Erasure Allowed	No
ONS CLASS/CLIP: Message Waiting Activate/Deactivate	No
Miscellaneous	
Auto Answer Allowed	Yes
Auto Answer Disconnect Tone - Enable	Yes
Auto Release on Key Select	No
Brokers Call	No
Called Party Features Override	No
Check COR after PSTN Dial Tone	No
Dialled Night Service	Yes
Disable Send Message	No
Flexible Answer Point	No
Individual Trunk Access	Yes
Key A	
Key B	
Key C	
Key D	
Multiline Set Loop Test	No
Multiline Set Message Center Remote Read Allowed	No

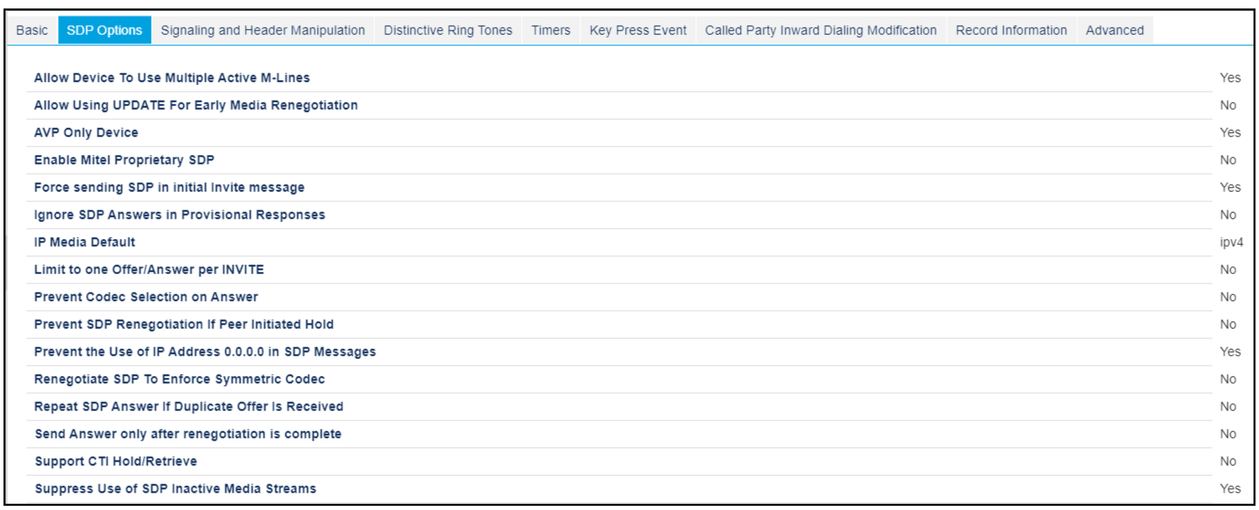
Figure 24 – Class of Service Options

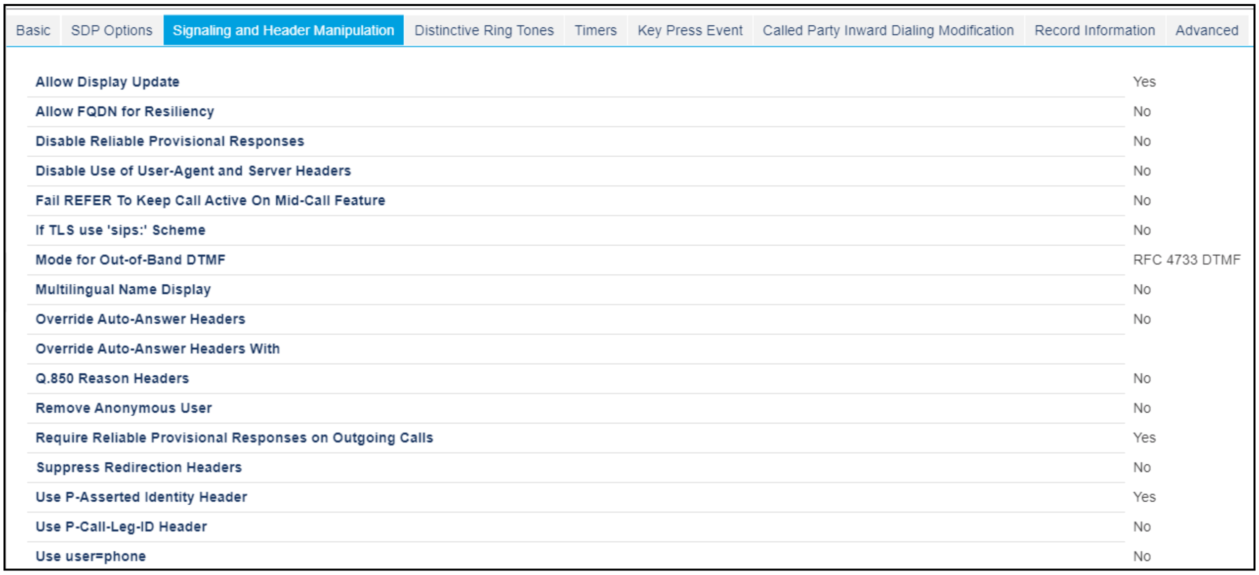
SIP Device Capabilities

This form provides configuration options that can be applied to various types of SIP devices. The association between the SIP device and the form is like how the Class of Service options work. The SIP Device Capabilities number provides a SIP profile that can be applied to SIP devices to allow for alternate capabilities as recommended through the Mitel interop process.

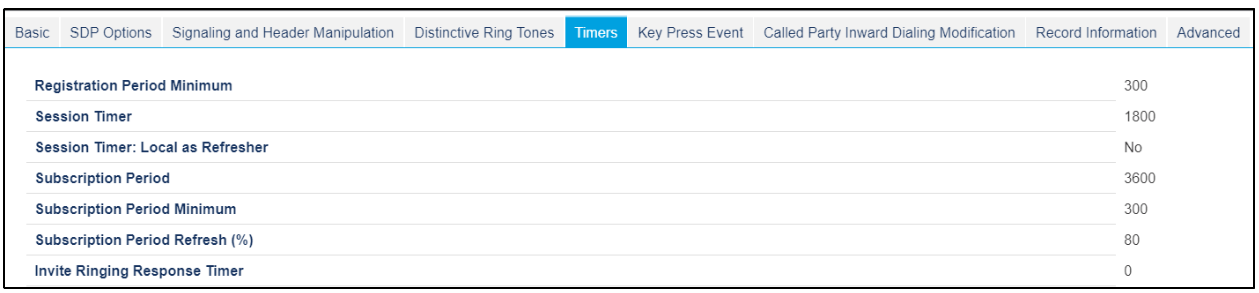
In the SIP Device Capabilities form, program a SIP Device Capabilities Number for the AudioCodes Mediant 1000 Gateway device. Ensure that “Replace System based with Device based In-Call Features” is set to ‘**Yes**’.

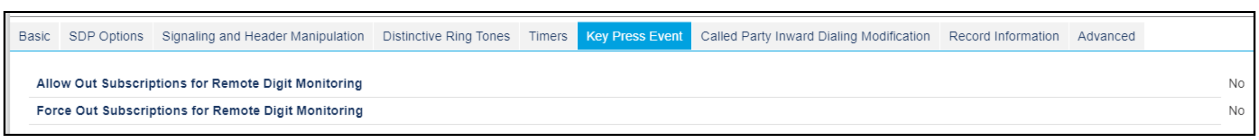





Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Allow Display Update Yes
Allow FQDN for Resiliency No
Disable Reliable Provisional Responses No
Disable Use of User-Agent and Server Headers No
Fail REFER To Keep Call Active On Mid-Call Feature No
If TLS use 'sips:' Scheme No
Mode for Out-of-Band DTMF RFC 4733 DTMF
Multilingual Name Display No
Override Auto-Answer Headers No
Override Auto-Answer Headers With
Q.850 Reason Headers No
Remove Anonymous User No
Require Reliable Provisional Responses on Outgoing Calls Yes
Suppress Redirection Headers No
Use P-Asserted Identity Header Yes
Use P-Call-Leg-ID Header No
Use user=phone No


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Enable Distinctive Ringing No
Internal Ring <http://www.notused.com>;info=alert-internal
External Ring <http://www.notused.com>;info=alert-external
Callback Ring <http://www.notused.com>;info=alert-community1


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Registration Period Minimum 300
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


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Allow Out Subscriptions for Remote Digit Monitoring No
Force Out Subscriptions for Remote Digit Monitoring No


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Update
Index
Digits to Match
Digit Length Operator
Digit Length
Number of Digits to Absorb
Digits to be Inserted

Figure 25 – SIP Device Capabilities

Station Attributes

Use the Station Attributes form to assign the previously configured Class of Service and SIP Device Capability number to each of the AudioCodes Mediant 1000 Gateway in the MiVoice Business. This form utilizes Range Programming.

Select the AudioCodes Mediant 1000 Gateway device number then select Change. Enter the previously configured SIP Device Capability number (**40**) and Class of Service for Day, Night 1 & Night 2 (**40**).
See an example in **Figure 26** below.

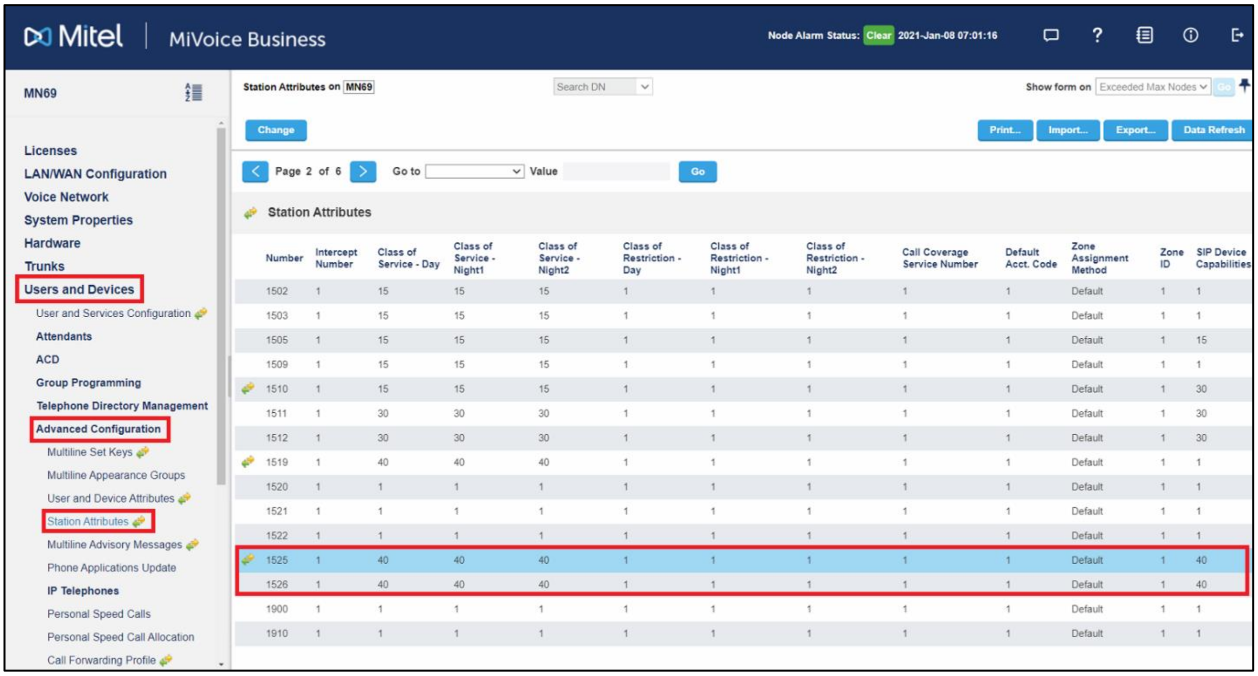


Figure 26 – Station Attributes

AudioCodes Mediant 1000 Gateway Configuration Notes

AudioCodes Setup

Basic configuration notes for configuring the AudioCodes Mediant 1000 Gateway with MiVoice Business.

All AudioCodes Mediant 1000 Gateway configuration can be done via web browser access to the AudioCodes Mediant 1000 Gateway IP address.

Note: The default IP address is 192.168.0.2 and default login credentials are Admin/Admin. You must configure Mediant 1000 IP address as per your network configuration.

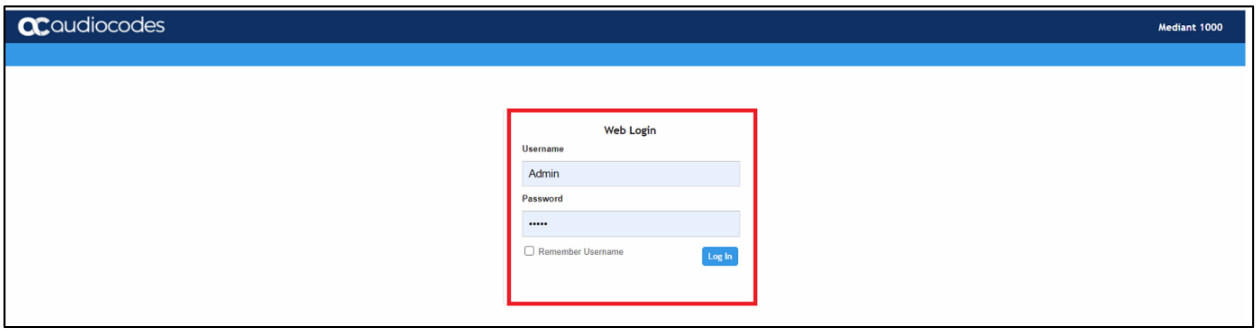


Figure 27 – Web Login

AudioCodes Home Screen

The home screen displays all the AudioCodes Mediant 1000 Gateway general information, GW and SBC.

Note: Prior to the interop the AudioCodes Mediant 1000 Gateway was upgraded to the latest GA load (v.7.20A.258.271).

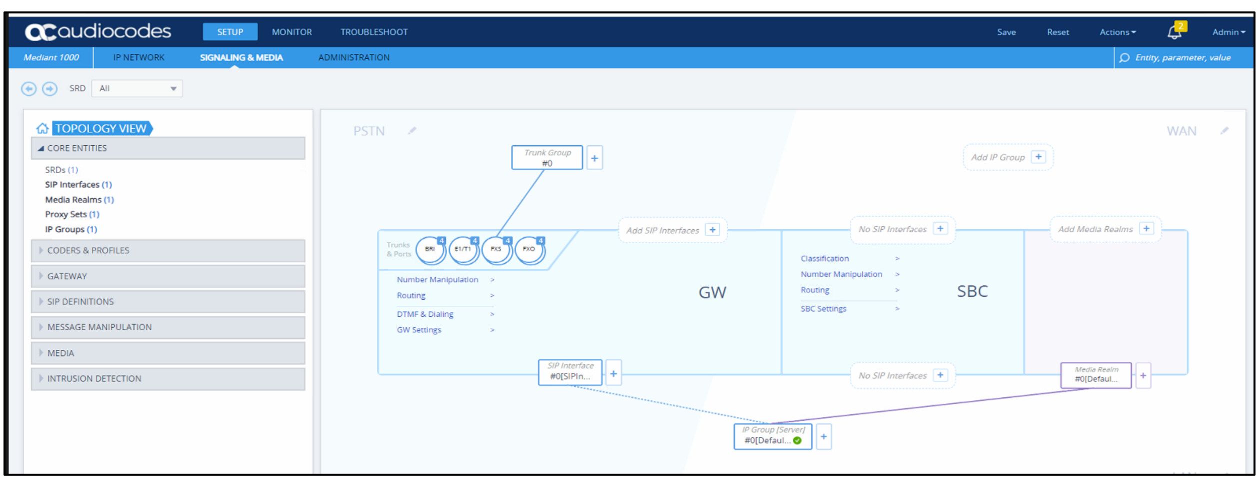


Figure 28 – Home Screen

Network Settings

Select: Setup > IP Network > IP Interfaces in the left pane. Note: You must then press the APPLY and SAVE button for any changes to take effect.

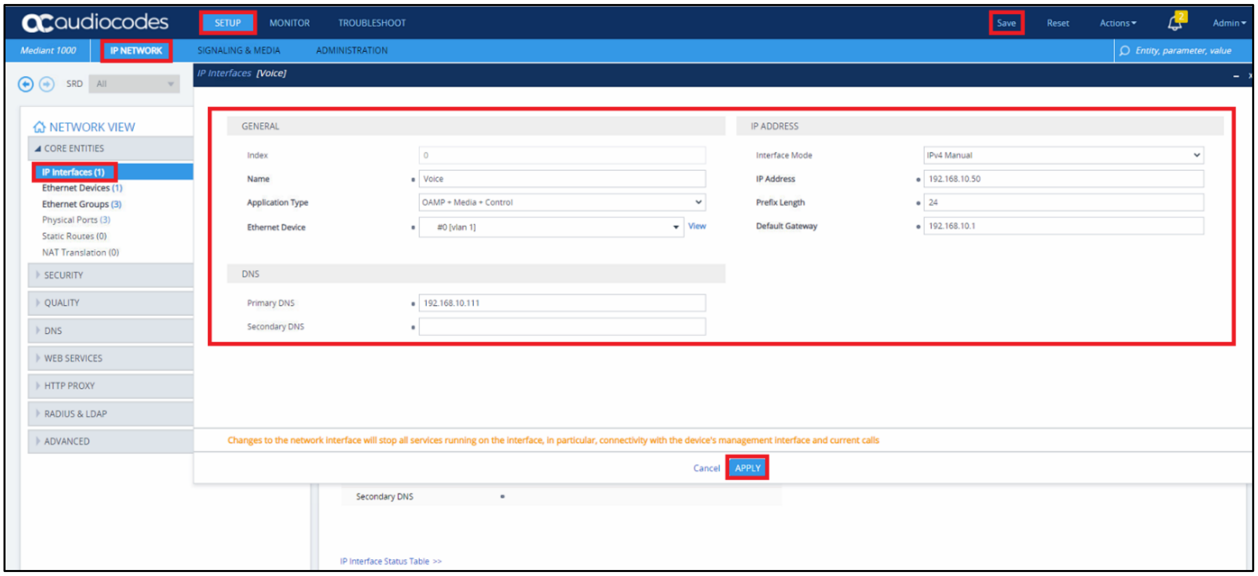


Figure 29 – Network Settings

Coder Groups

Select: Setup > Signaling & Media > Coders and Profiles > Coder Groups. Note: Configure in order or preference from most preferred codec to the least.

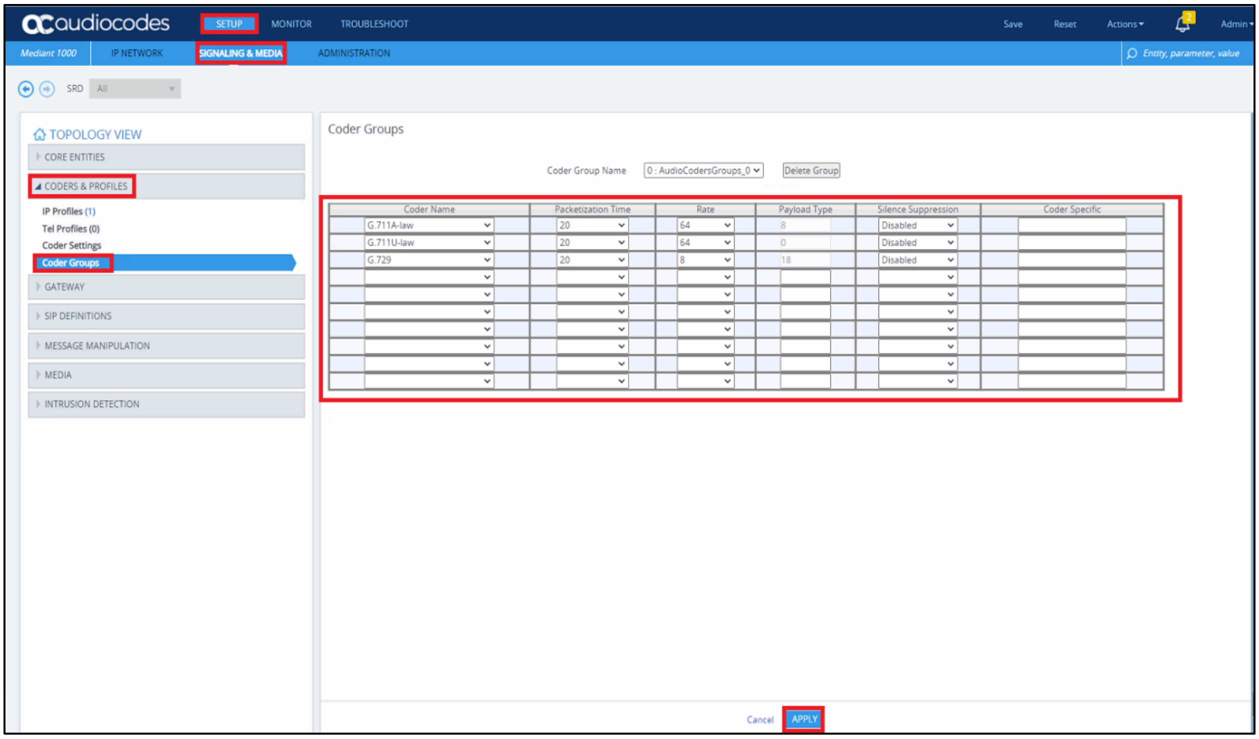


Figure 30 – Coder Groups

Media Settings

Select: Setup > Signaling & Media > Media > Voice Settings in the left-hand pane.

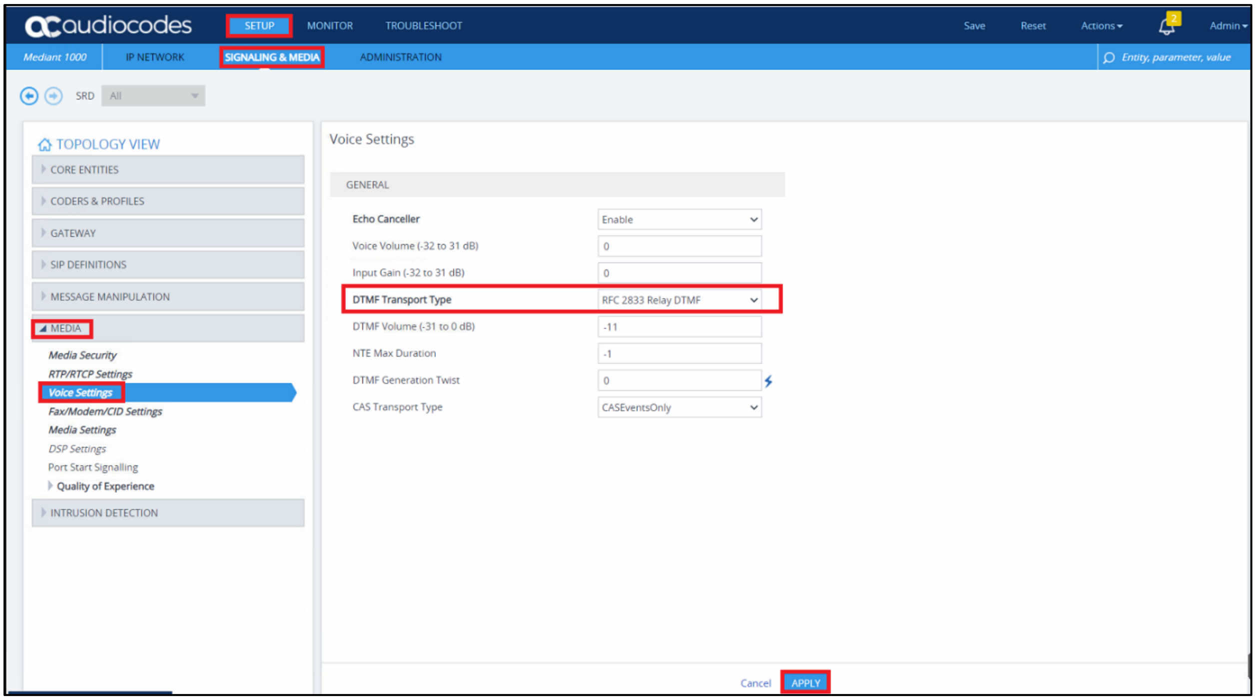


Figure 31 – Media Settings

DTMF Settings

Select: Setup > Signaling & Media > Gateway > DTMF and Supplementary > DTMF & Dialing in the left pane.

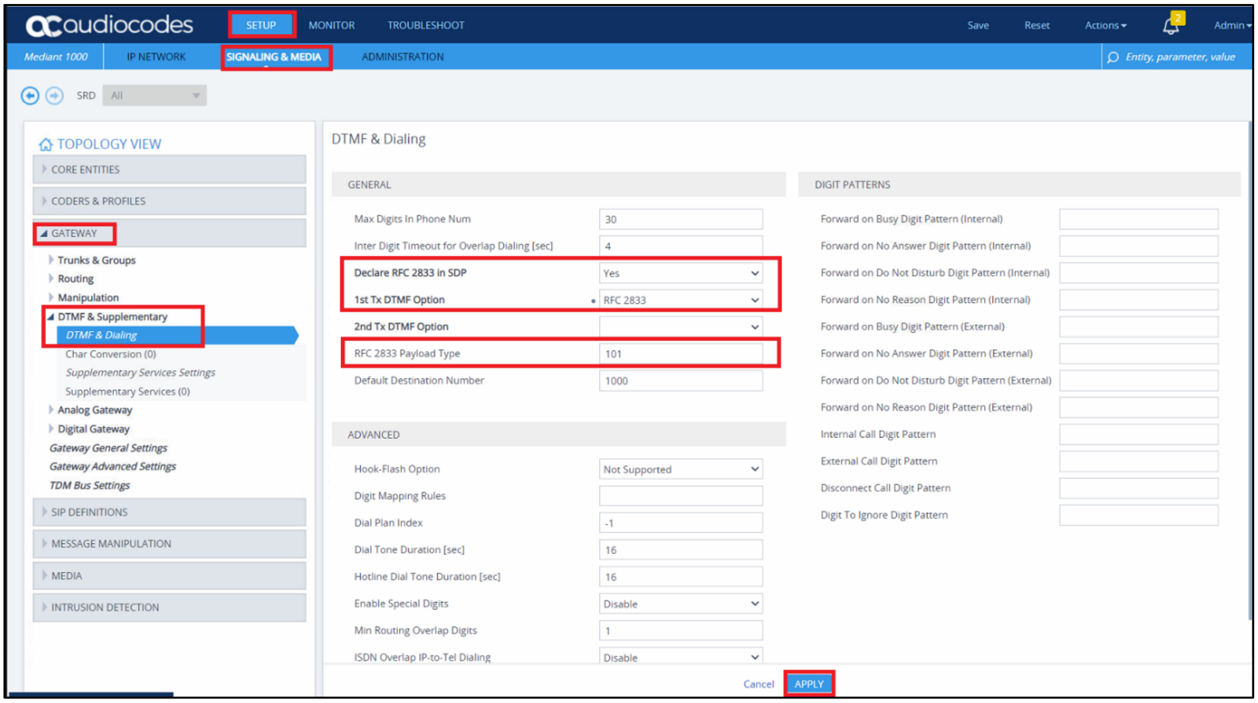


Figure 32 – DTMF Settings

DTMF Supplementary Services Settings

Select: Setup > Signaling & Media > Gateway > DTMF and Supplementary > Supplementary Services Settings in the left pane.

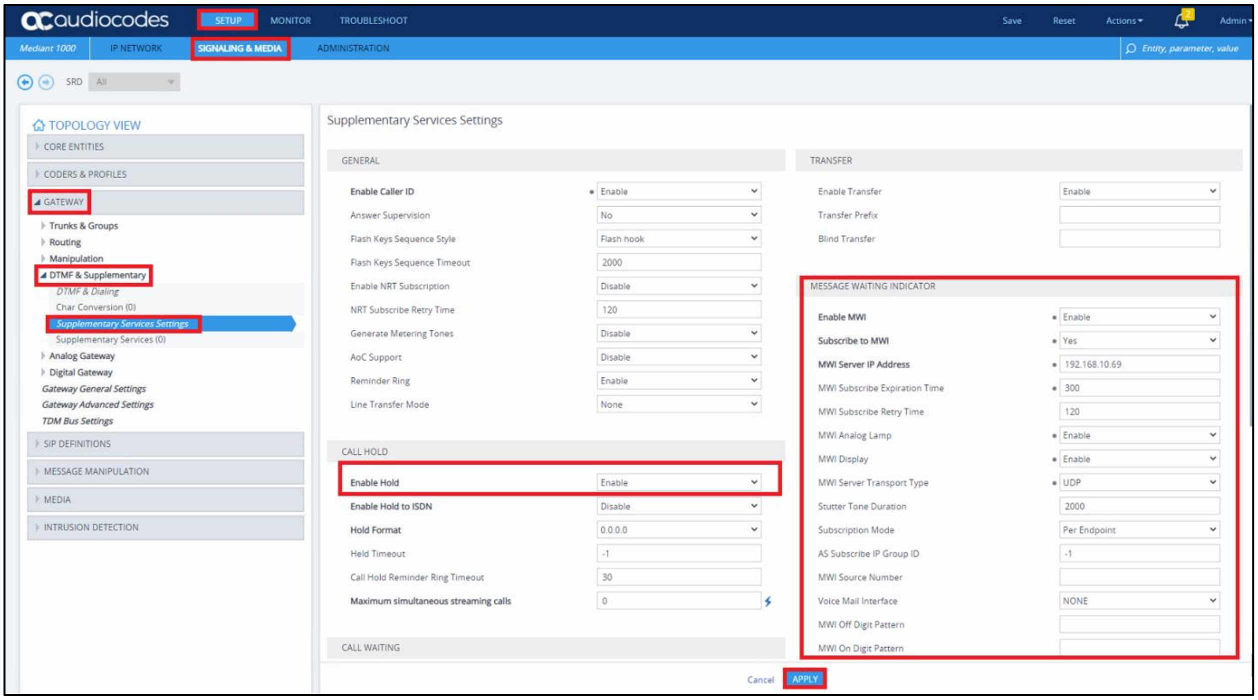


Figure 33 – DTMF Supplementary Service Settings

SIP Definitions General Settings

Select: Setup > Signaling & Media > SIP Definitions > SIP Definitions General Settings in the left-hand pane.

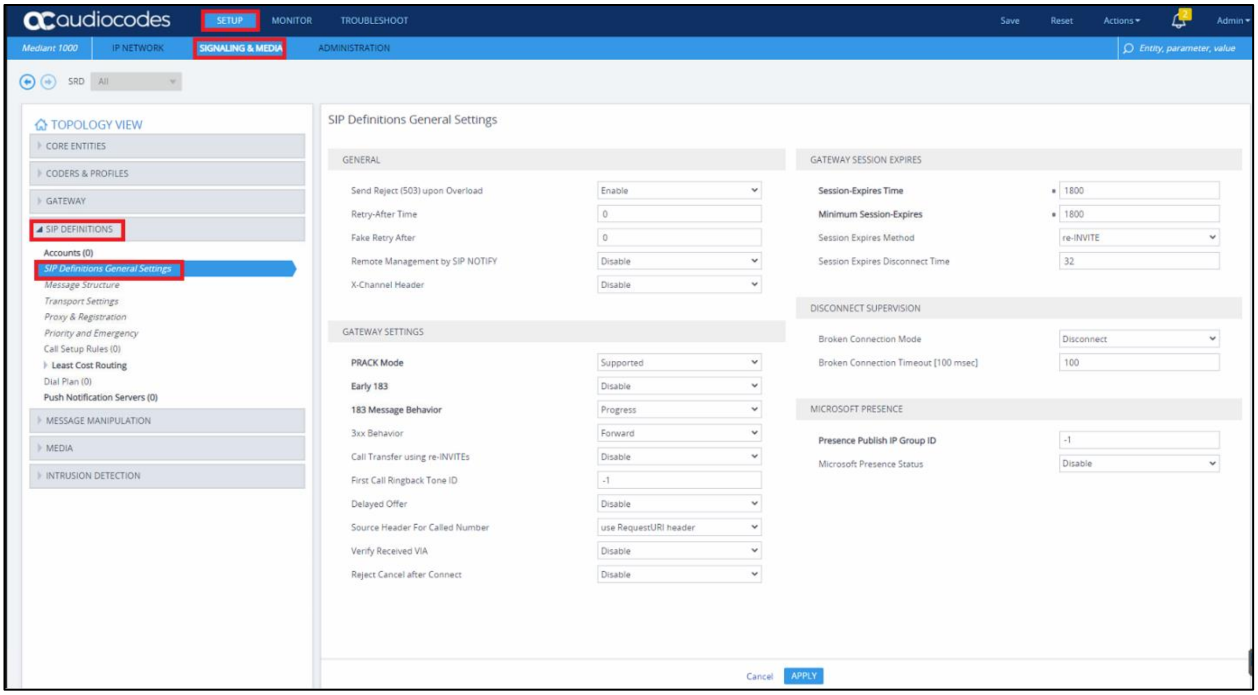


Figure 34 – General Settings

Trunk Configuration (FXO)

Proxy & Registration

Select: Setup > Signaling & Media > SIP Definitions > Proxy & Registration in the left pane.

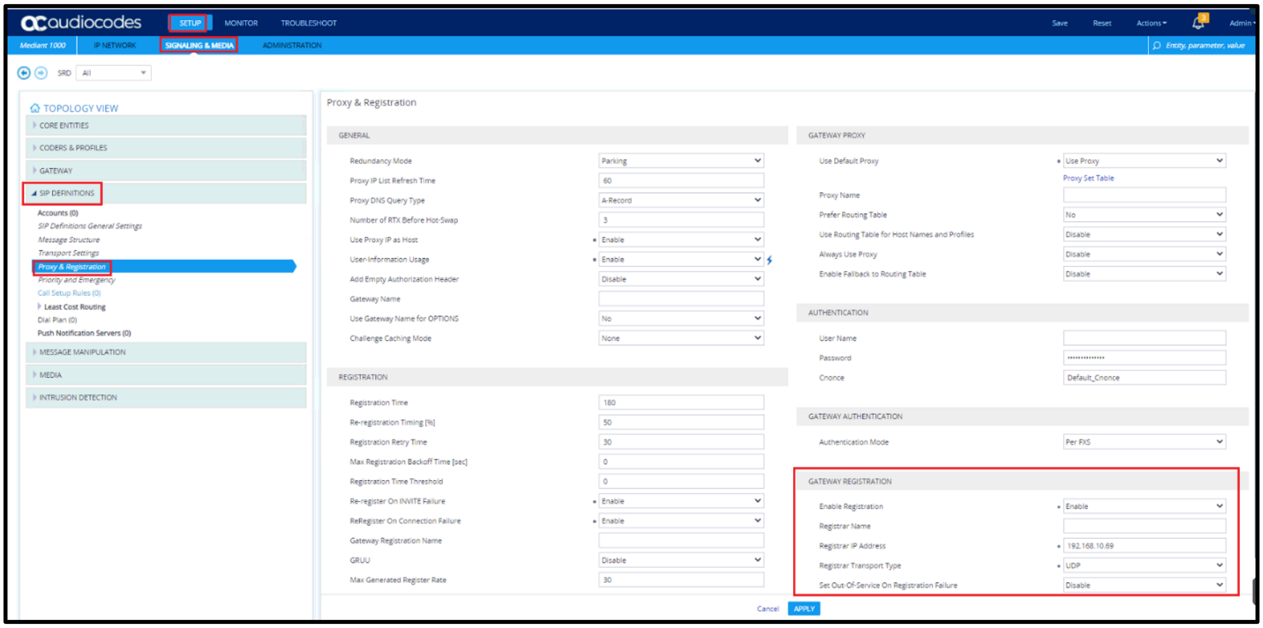


Figure 35 – Proxy & Registration

Trunk Groups

Select: Setup > Signaling & Media > Gateway > Trunks & Groups > Trunk Groups > Seclect Module to Module 3 FXS from drop down > Provide Channels (Ex: 1) > Apply.

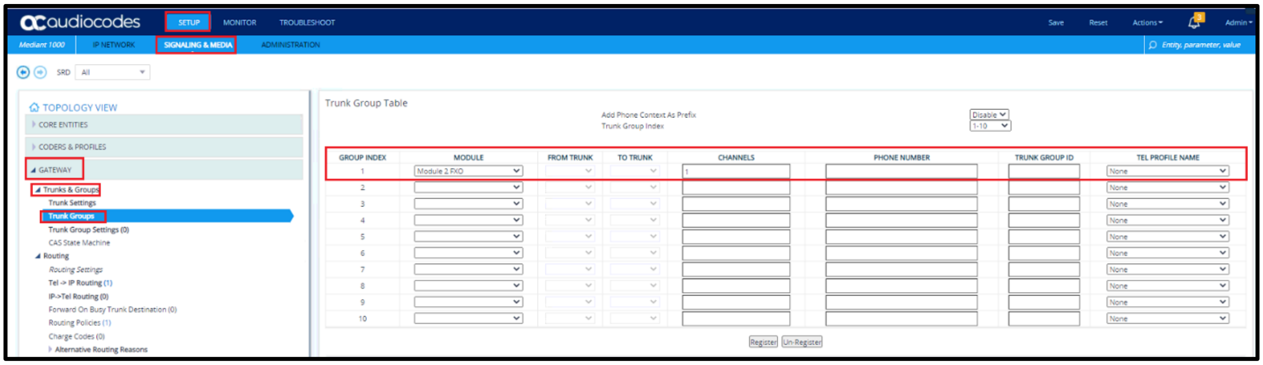


Figure 36 – Trunk Groups

Automatic Dialing

Gateway > Analog Gateway > Automatic Dialing > Port 1 FXO > Destination Phone Number to MiVB internal extension number (Ex:4000) > Submit.

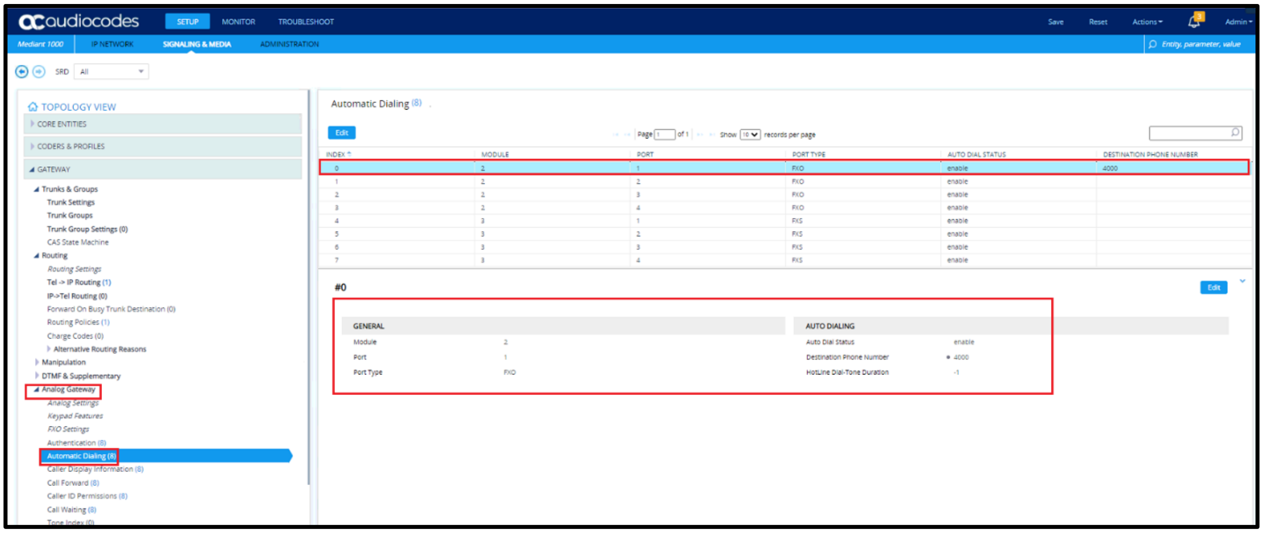


Figure 37 – Automatic Dialing

Example Call Flows for Trunkside:

Inbound PSTN Call

When making a call from PSTN to Analog DID (+91 8067591215), The call first comes to AudioCodes and then AudioCodes forwards the call to MiVB internal extension based on Automatic Dialing configuration (Ex:4000) on AudioCodes Device and then MiVB internal extension will ring.

Outbound PSTN Call

When making a call from MiVB internal extension to PSTN number (ARS followed by 0 and mobile number for example: 45608123347168), The call comes to AudioCodes and AudioCodes routes the call to PSTN over FXO port and then PSTN number will ring.

Lineside Configuration (FXS)

Proxy & Registration

Select: Setup > Signaling & Media > SIP Definitions > Proxy & Registration in the left pane.

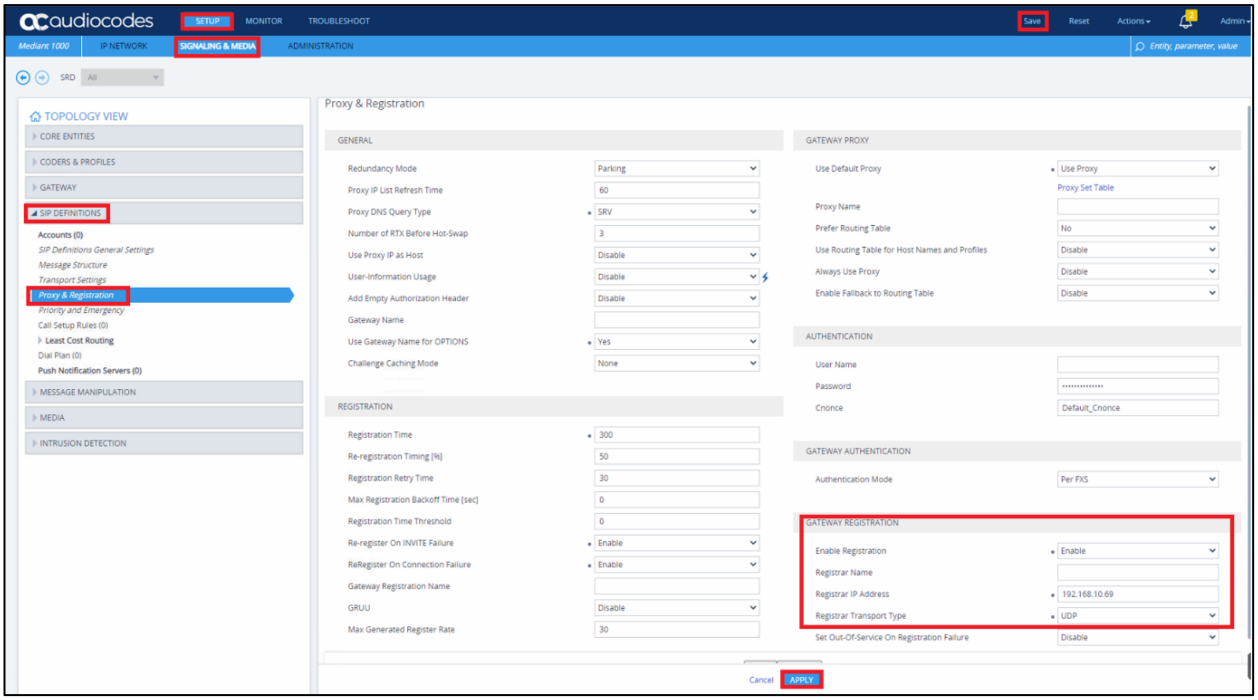


Figure 38 – Proxy & Registration

Trunk Groups

Select: Setup > Signaling & Media > Gateway > Trunks & Groups > Trunk Groups > Seclect Module to Module 3 FXS from drop down > Provide Channels (Ex: 1) > Provide Phone Number (Ex: 1525) > Apply.

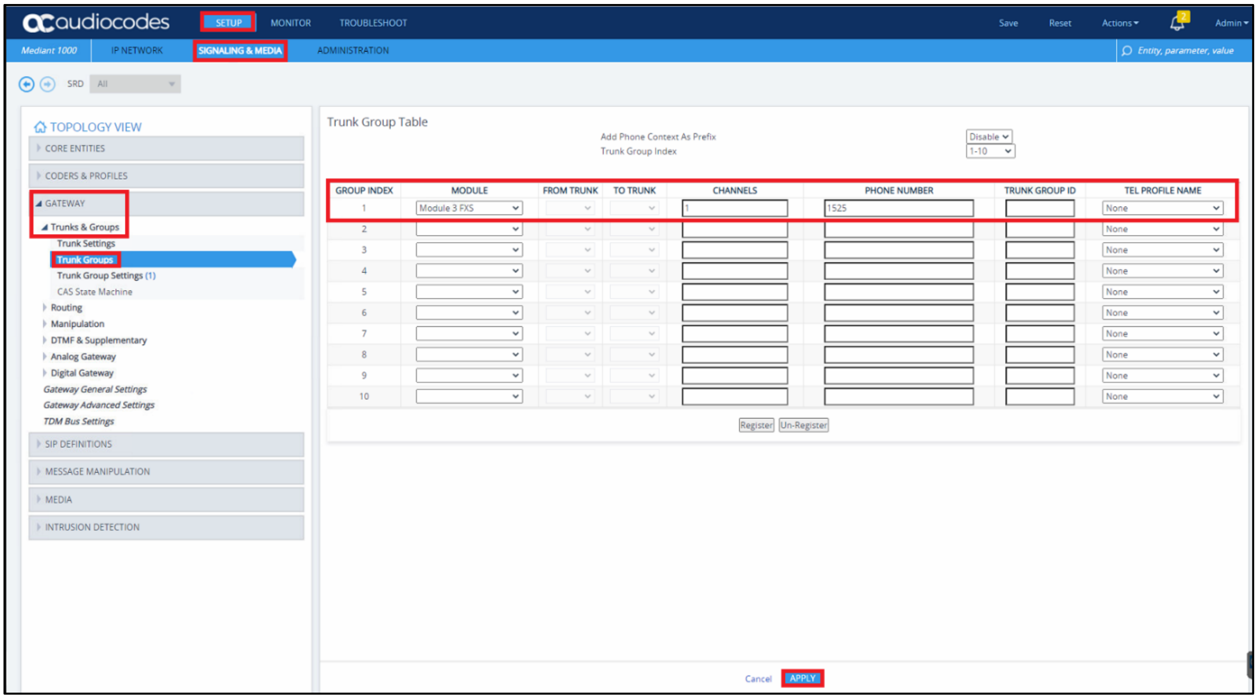


Figure 39 – Trunk Groups

Authentication Settings

Select: Setup > Signaling & Media > Gateway > Analog Gateway > Authentication.

Enter the user name and password for authentication as shown in the figure below.

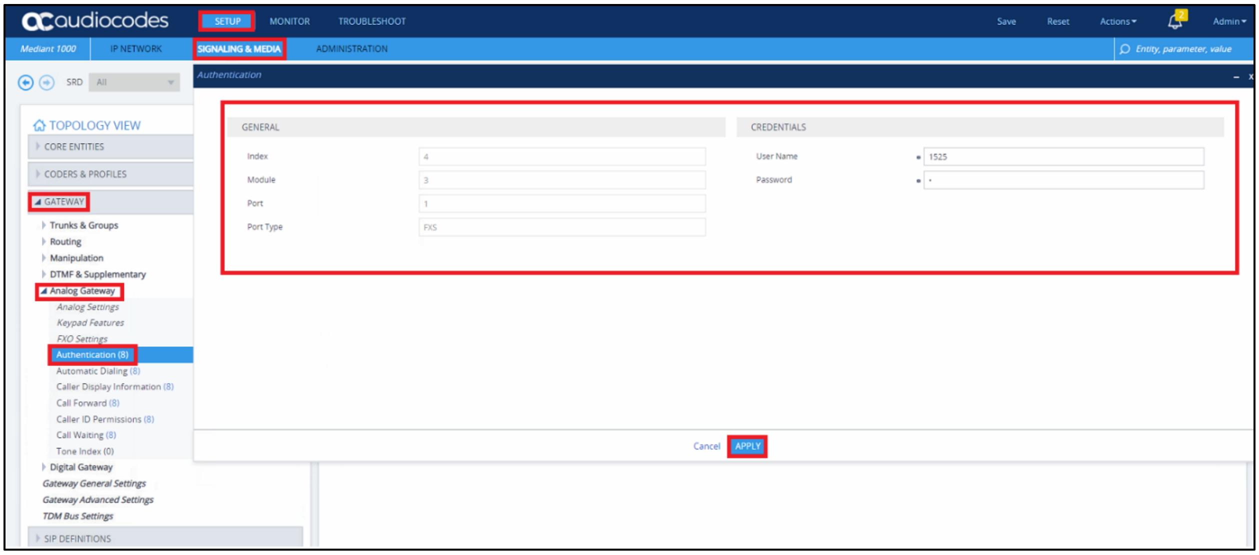


Figure 40 – Authentication Settings

Message Manipulation

Select: Setup > Signaling & Media > Message Manipulation > Message Manipulations > Apply.

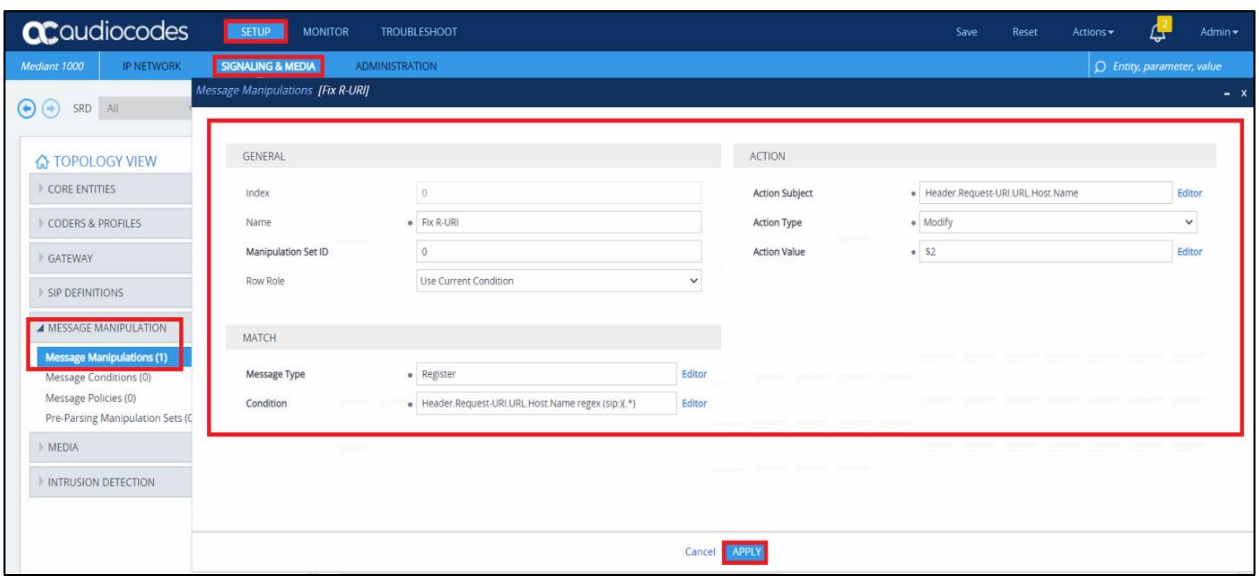


Figure 41 – Message manipulation

Caller Display Information Settings

Select: Setup > Signaling & Media > Gateway > Analog Gateway > Caller Display Information.

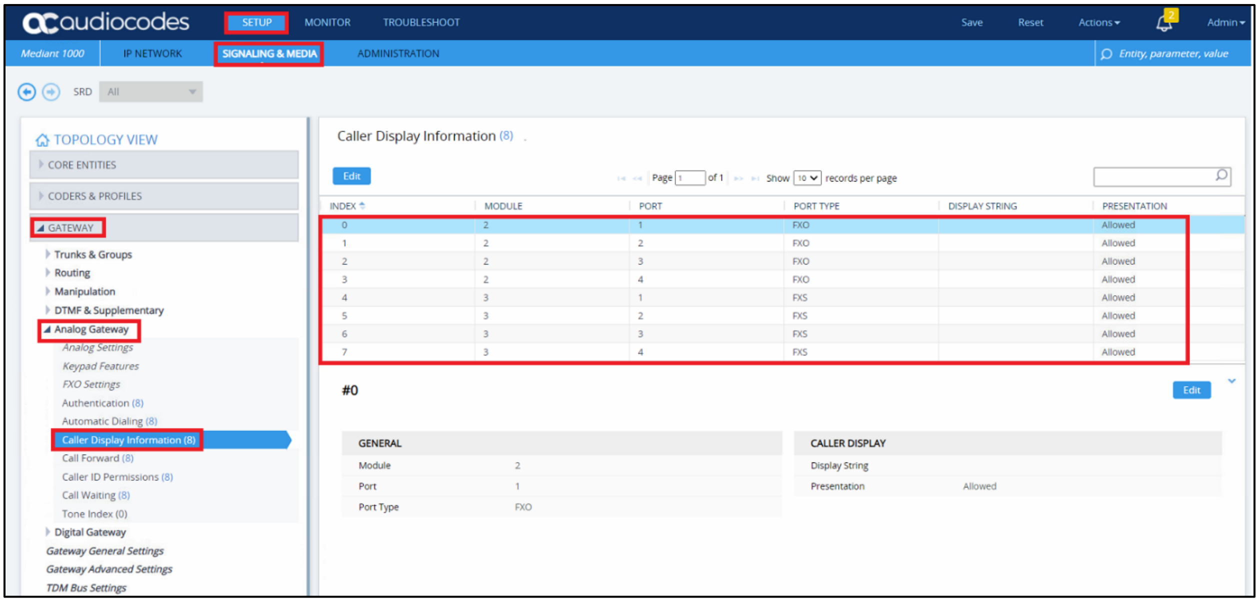


Figure 42 – Called Display information Settings

Caller ID Settings

Select: Setup > Signaling & Media > Gateway > Analog Gateway > Caller ID Permissions.

Enable Caller ID for FXS or FXO based on your scenario and requirement.

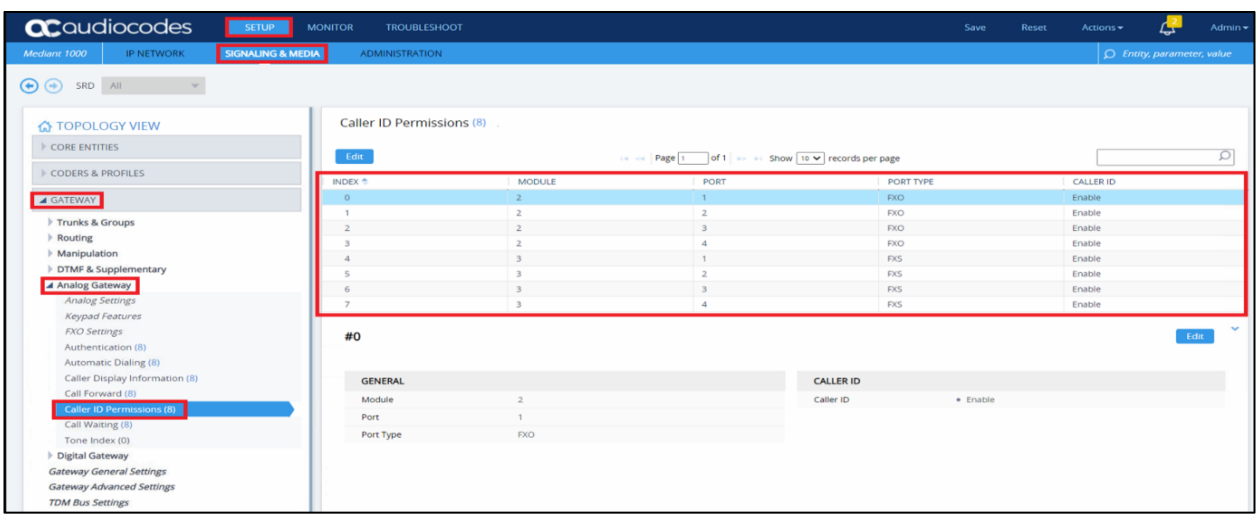
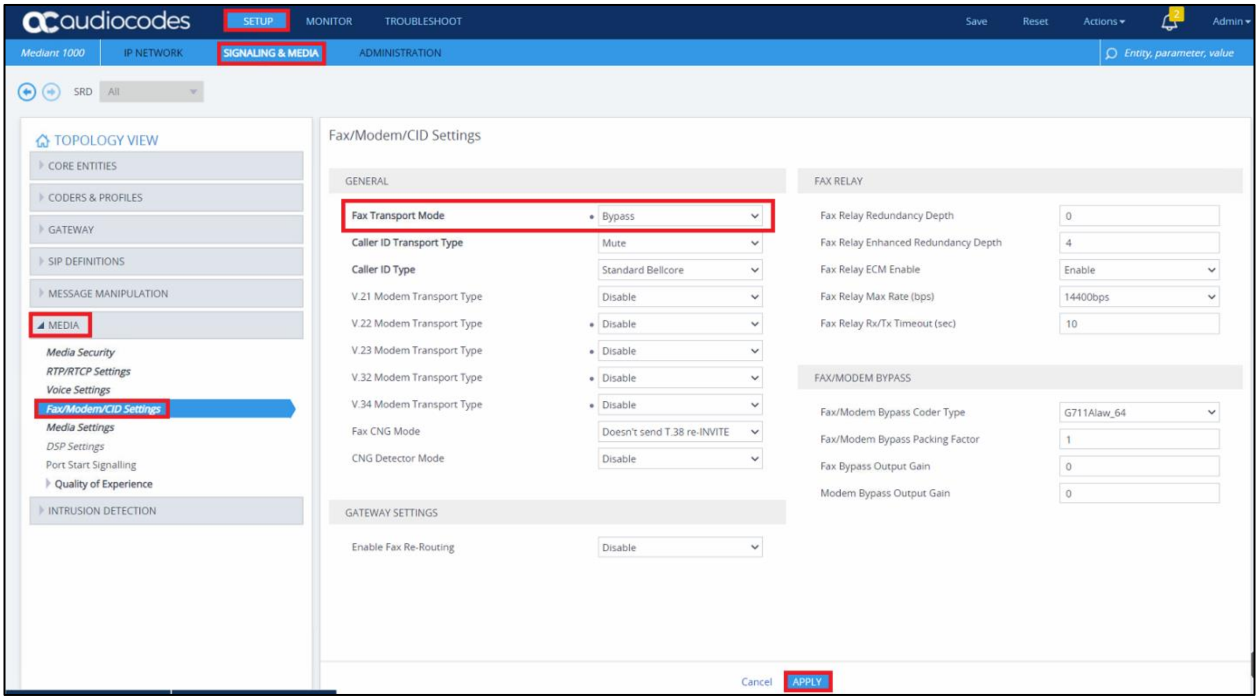


Figure 43 – Called ID Settings

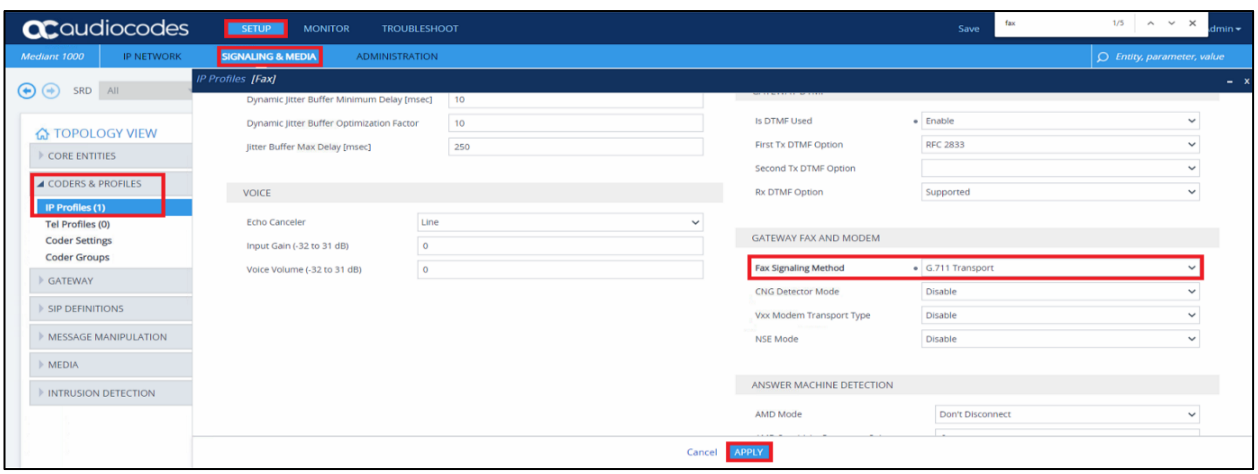
G.711 FAX Settings

For all FAX scenarios using G711, use the following configuration in Mediant 1000 and MiVB.

Select: Setup > Signaling & Media > Media > Fax/Modem/CID Settings > General > Fax Transport Mode set to Bypass > Apply.



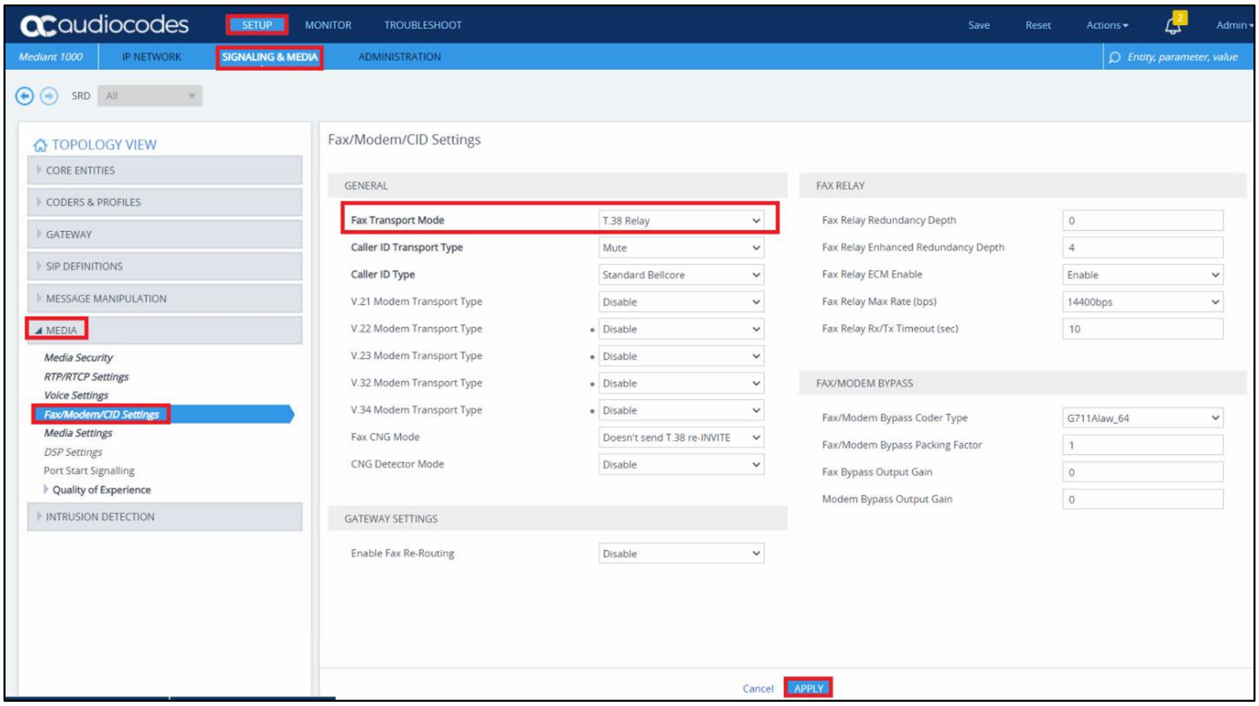
After that go to Coders and Profiles > IP Profiles > Fax Signaling Method set to G711 Transport > Apply.



T.38 FAX Mode Settings

To use T.38 for FAX, use the following configuration in Mediant 1000 and MiVB.

Select: Setup > Signaling & Media > Media > Fax/Modem/CID Settings > General > Fax Transport Mode set to T.38 Relay > Apply.



After that go to Coders and Profiles > IP Profiles > Fax Signaling Method set to T.38 Relay > Apply.

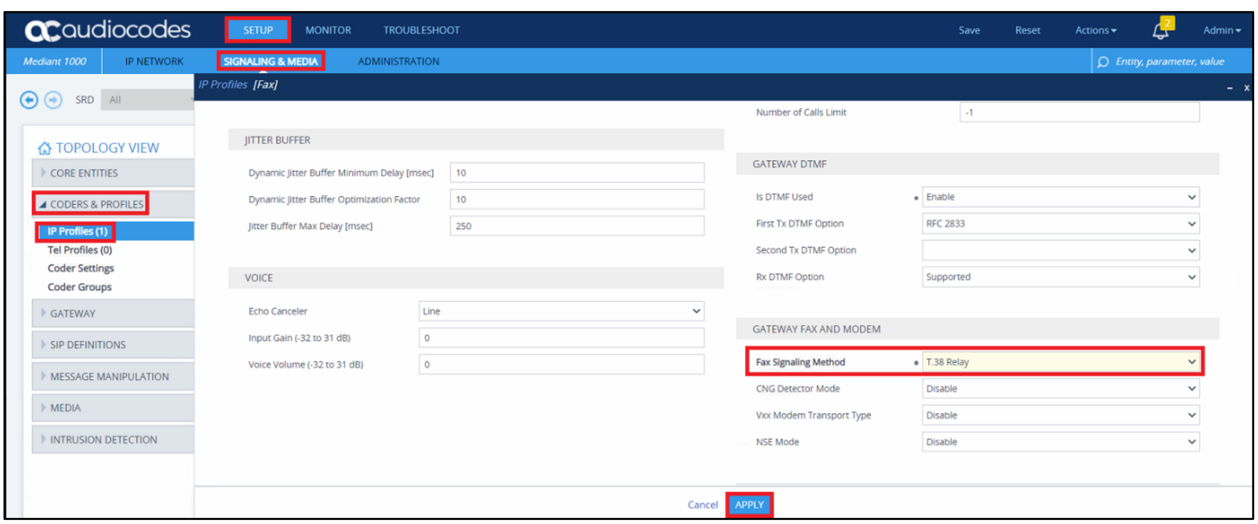


Figure 44 – Fax Settings

MiVoice Border Gateway Setup Notes (Optional)

The following steps show how to program the MiVoice Border Gateway (MBG) server to allow connections between AudioCodes Mediant 1000 Gateway and the MiVoice Business for teleworking.

Network Requirements

- Please refer to the Multi-Protocol Border Gateway Engineering guidelines for further information.

Assumptions for MBG Configuration

- MiVB configuration completed as per instructions in previous section.
- The SIP signaling connection between the MiVoice Business (MiVB) and MBG server uses UDP on Port 5060.
- MBG server installed and configured for SIP clients' support.

MiVoice Business

Select **Network > ICPs** and click **+ (Add an ICP)**

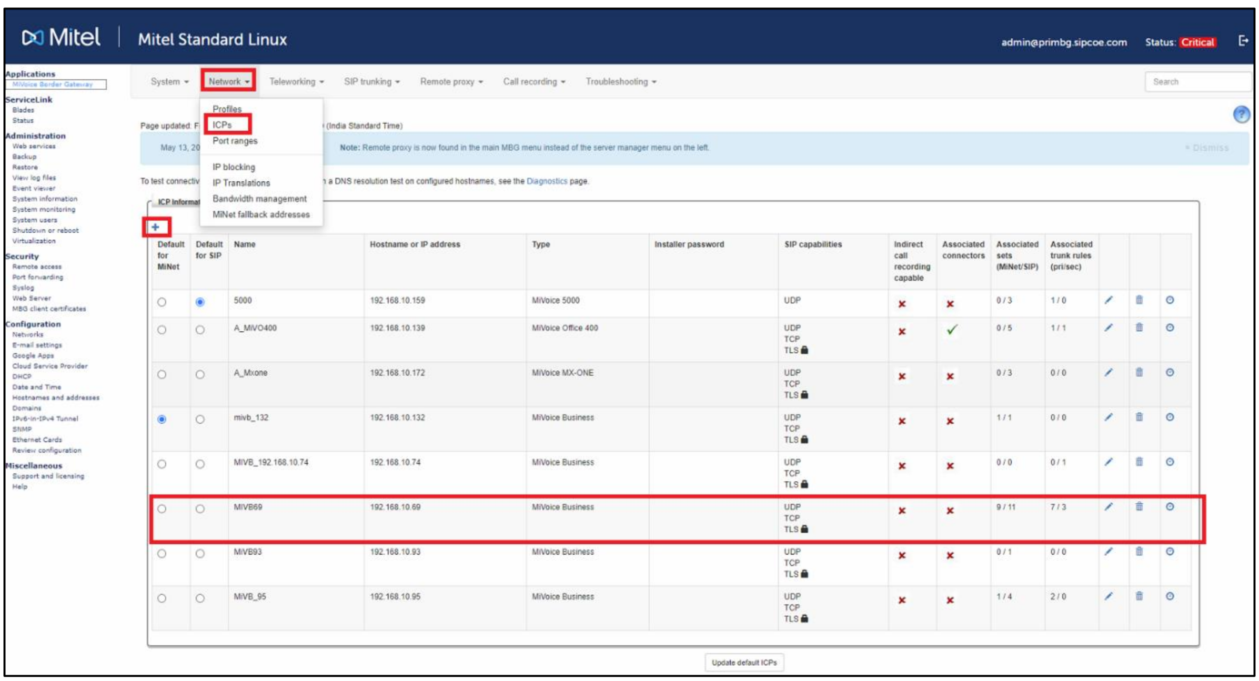


Figure 45 – Setting up Default ICP

Enter ICP information (name, IP, type) and select **Save.**

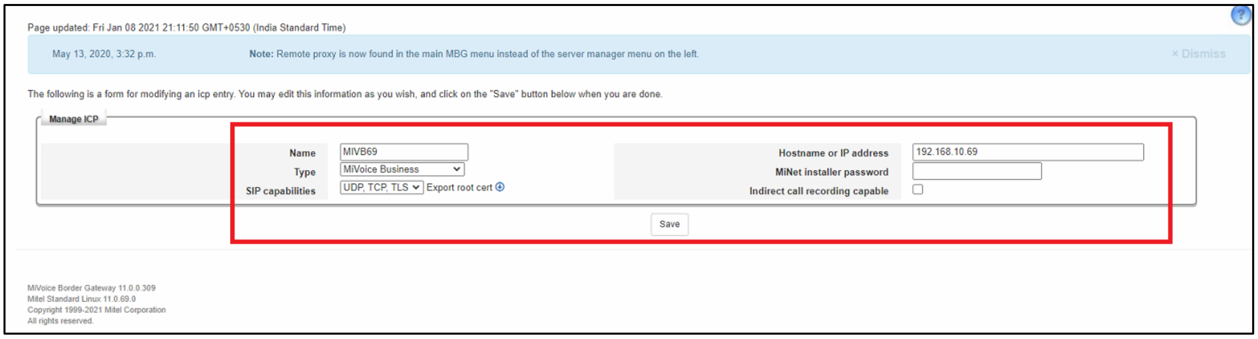


Figure 46 – Setting up Default ICP

Adding SIP devices

Navigate to **Teleworking > SIP**. Click **+ (Add)** a SIP Device as shown below.

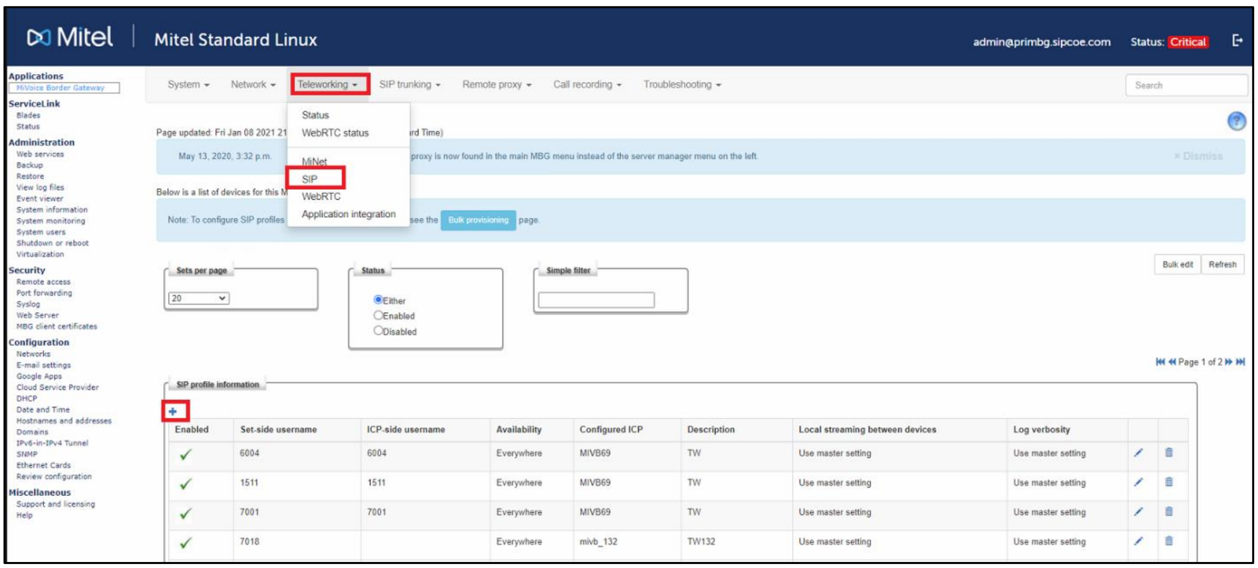


Figure 47 – Creating SIP User

In the opened form, enter the data to create the new SIP device in MBG.

Enter all required information. Set side credentials must match username and password provisioned on the phone. ICP side credentials must match Login PIN and Number provisioned on the MiVB. Since PRACK is disabled on master setting, if PRACK is enabled on MiVB, then you must enable it. Click Save when you are done.

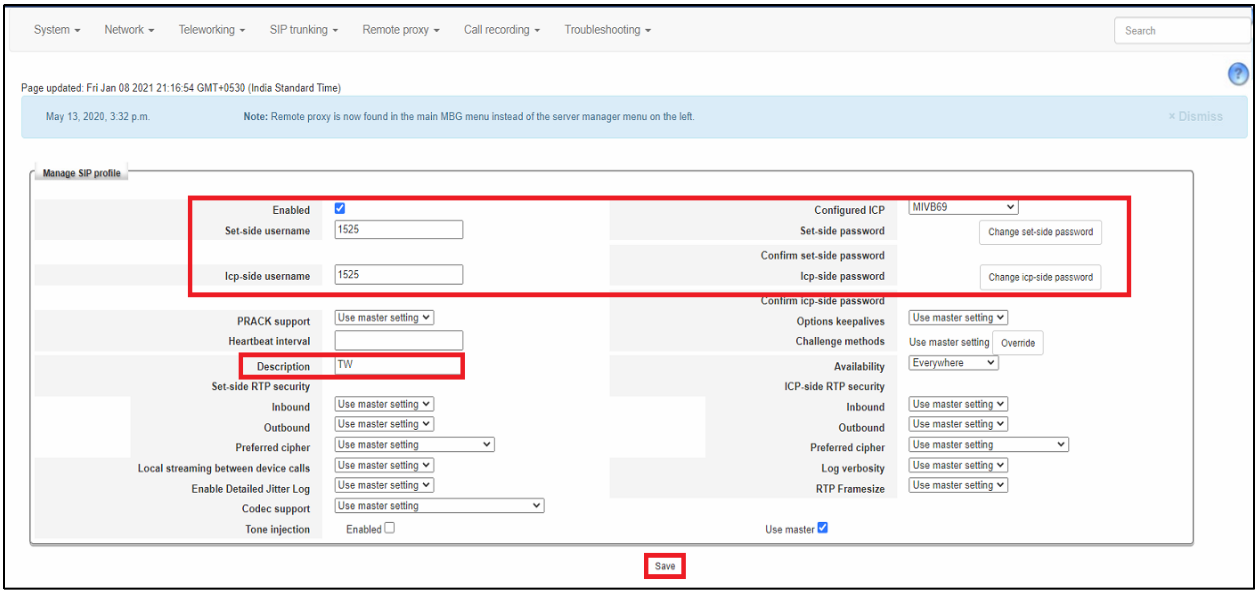
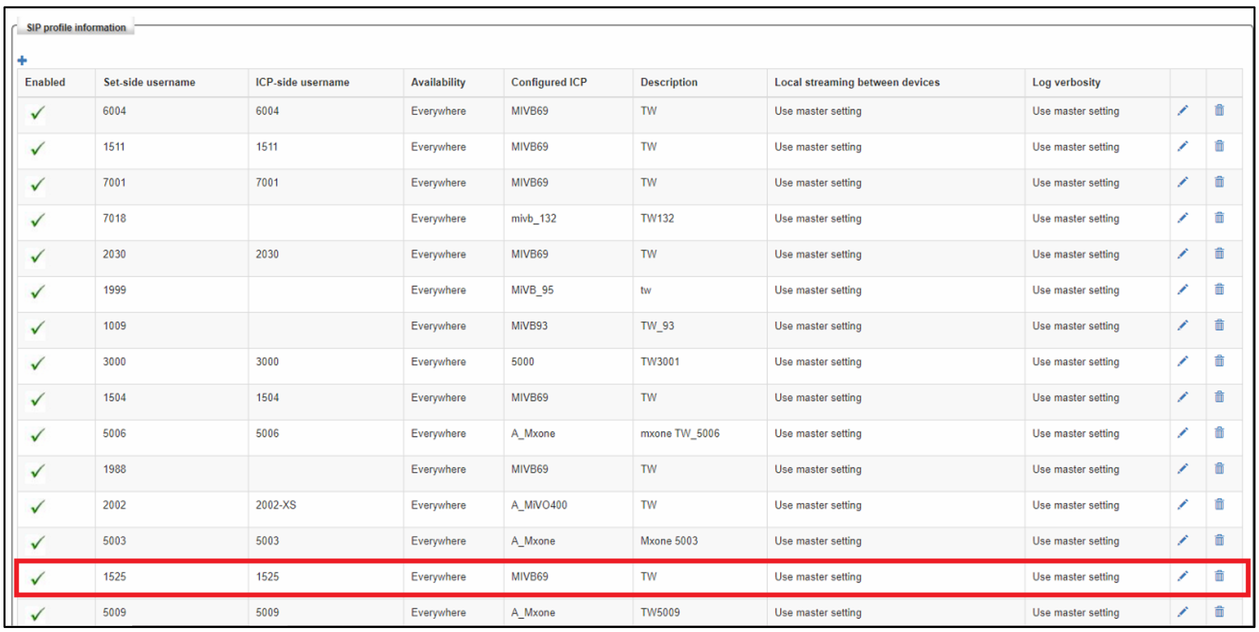


Figure 48 – Entering SIP Device Details


SIP profile information

Enabled	Set-side username	ICP-side username	Availability	Configured ICP	Description	Local streaming between devices	Log verbosity
✓	6004	6004	Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	1511	1511	Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	7001	7001	Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	7018		Everywhere	mivb_132	TW132	Use master setting	Use master setting
✓	2030	2030	Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	1999		Everywhere	MIVB_95	tw	Use master setting	Use master setting
✓	1009		Everywhere	MIVB93	TW_93	Use master setting	Use master setting
✓	3000	3000	Everywhere	5000	TW3001	Use master setting	Use master setting
✓	1504	1504	Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	5006	5006	Everywhere	A_Mxone	mxone TW_5006	Use master setting	Use master setting
✓	1988		Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	2002	2002-XS	Everywhere	A_MiVO400	TW	Use master setting	Use master setting
✓	5003	5003	Everywhere	A_Mxone	Mxone 5003	Use master setting	Use master setting
✓	1525	1525	Everywhere	MIVB69	TW	Use master setting	Use master setting
✓	5009	5009	Everywhere	A_Mxone	TW5009	Use master setting	Use master setting

Figure 49 –SIP device Information

SIP Settings

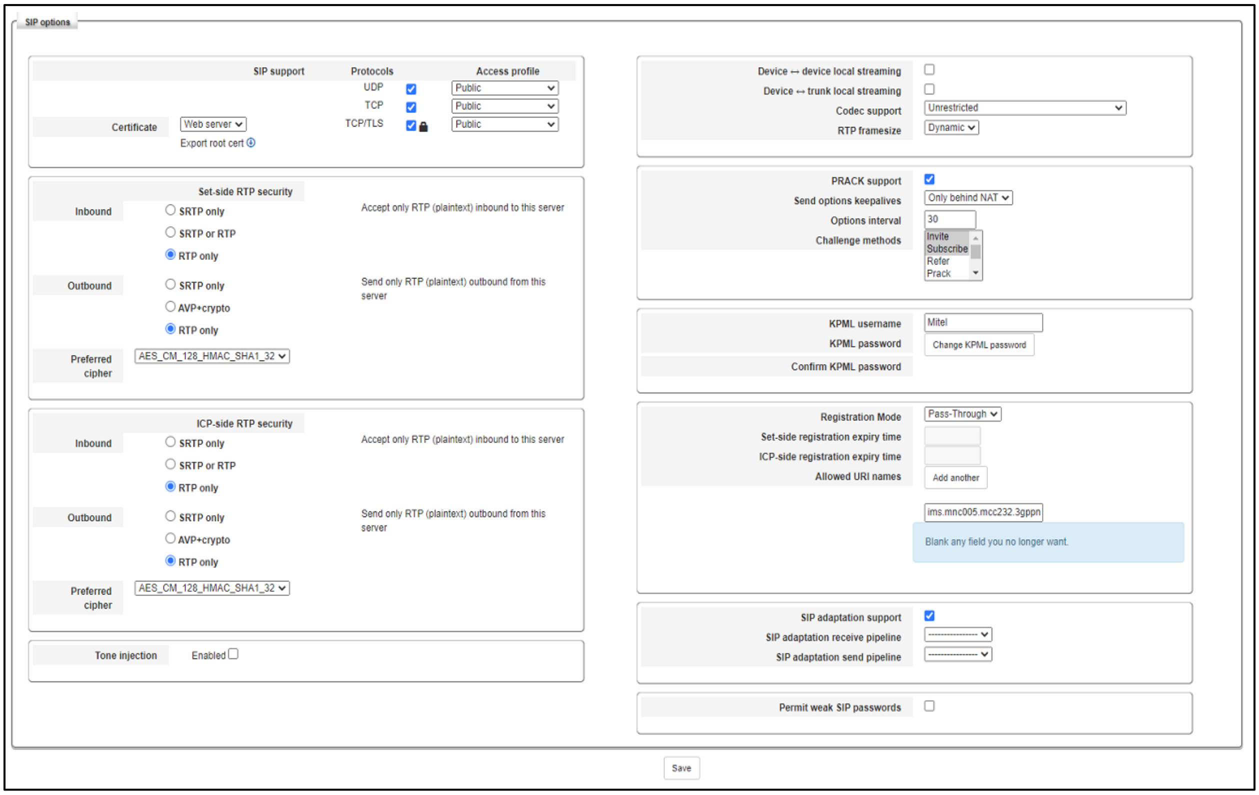


Figure 50 –SIP Settings

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
AudioCodes	AC