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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)

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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:

| | |
|---|---|
|  | 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. | <input checked="" type="checkbox"/> |
| Nu-Point Voicemail | Terminating calls to a NuPoint voicemail boxes and DTMF detection. | <input type="checkbox"/> |
| Automatic Call Distribution | Making calls to an ACD environment with RAD treatments, Interflow and Overflow call scenarios and DTMF detection. | <input checked="" type="checkbox"/> |
| Packetization | Forcing the MiVoice Business to stream RTP packets through its E2T card at different intervals, from 10ms to 30ms | <input checked="" type="checkbox"/> |
| 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. | <input checked="" type="checkbox"/> |
| Teleworker | Making and receiving a call through the AudioCodes Mediant 1000 Gateway to and from Teleworker extensions. | <input checked="" type="checkbox"/> |
| Video | Making and receiving a call through the AudioCodes Mediant 1000 Gateway with video capable devices. | <input type="checkbox"/> |
| Fax | G711 & T.38 Fax Calls. | <input checked="" type="checkbox"/> |
| TLS/SRTP | Basic incoming/outgoing calls. | <input checked="" type="checkbox"/> |
| Resiliency | Device able to handle resiliency when primary MiVB goes down. | <input checked="" type="checkbox"/> |

- No issues found - 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. | <input checked="" type="checkbox"/> |
| DTMF Signal | Sending DTMF after call setup (i.e. mailbox password) | <input checked="" type="checkbox"/> |
| Registration/Authentication | Device registration w/o authentication | <input checked="" type="checkbox"/> |
| Call Hold | Putting a call on hold | <input checked="" type="checkbox"/> |
| Call Transfer | Transferring a call to another destination | <input checked="" type="checkbox"/> |
| Call Forward | Forwarding a call to another destination | <input checked="" type="checkbox"/> |
| Conference | Conferencing multiple calls together | <input checked="" type="checkbox"/> |
| Redial | Last Number Redial | <input checked="" type="checkbox"/> |
| Personal Ring Group | Multiple sets ringing when one number dialed | <input checked="" type="checkbox"/> |
| Video | Video Capabilities | <input checked="" type="checkbox"/> |
| TLS/SRTP | Basic incoming/outgoing call. Teleworker incoming/outgoing call. | <input checked="" type="checkbox"/> |
| Resiliency | Device able to handle resiliency when primary MiVB or MBG goes down. | <input checked="" type="checkbox"/> |
| MWI | Message Waiting Indication | <input checked="" type="checkbox"/> |
| G.711 and T.38 Fax | Fax Messages | <input checked="" type="checkbox"/> |

- No issues found - 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 | <p>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.</p> <p>Recommendation: Please contact Audiocodes support for further details.</p> |
| Call when other party is Busy | <p>AudioCodes Mediant 1000 Gateway doesn't send busy tone to the caller.</p> <p>Recommendation: Contact AudioCodes support for more details.</p> |
| TLS/SRTP | <p>Force SDP and AVP only should be disabled under SIP peer profile in order to use TLS/SRTP with MiVB.</p> <p>Recommendation: Contact Mitel Support for more information.</p> |
| Resiliency | <p>Service provider resiliency feature is not tested in case of trunking environment. So, we have tested only MiVB resiliency.</p> <p>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.</p> |

Lineside:

| Feature | Problem Description |
|-------------------|--|
| TLS/SRTP | <p>Force SDP and AVP only should be disabled under SIP Device Capability in order to use TLS/SRTP with MiVB.</p> <p>Recommendation: Contact Mitel Support for more information.</p> |
| Basic Call | <p>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.</p> <p>Recommendation: India Re-order tones should be uploaded on AudioCodes Mediant 1000 Gateway or Contact AudioCodes support for further clarifications on this.</p> |
| Resiliency | <p>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.</p> <p>Workaround – AudioCodes support has provided workaround with help of message maunipulation on Meidant 1000 in order to fix this issue.</p> <p>Issue ID - SBC-25918</p> <p>Recommendation: Contact AudioCodes support for further clarifications on this.</p> |

Network Topology

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

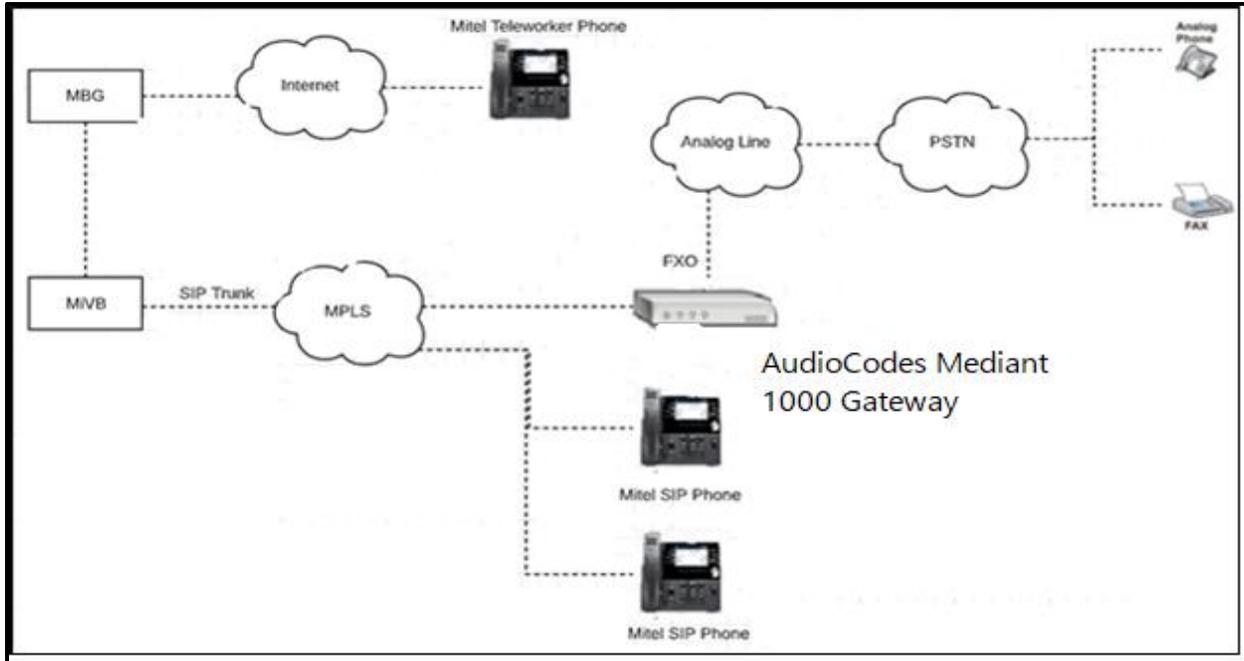


Figure 1 – Network Topology for Trunkside

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

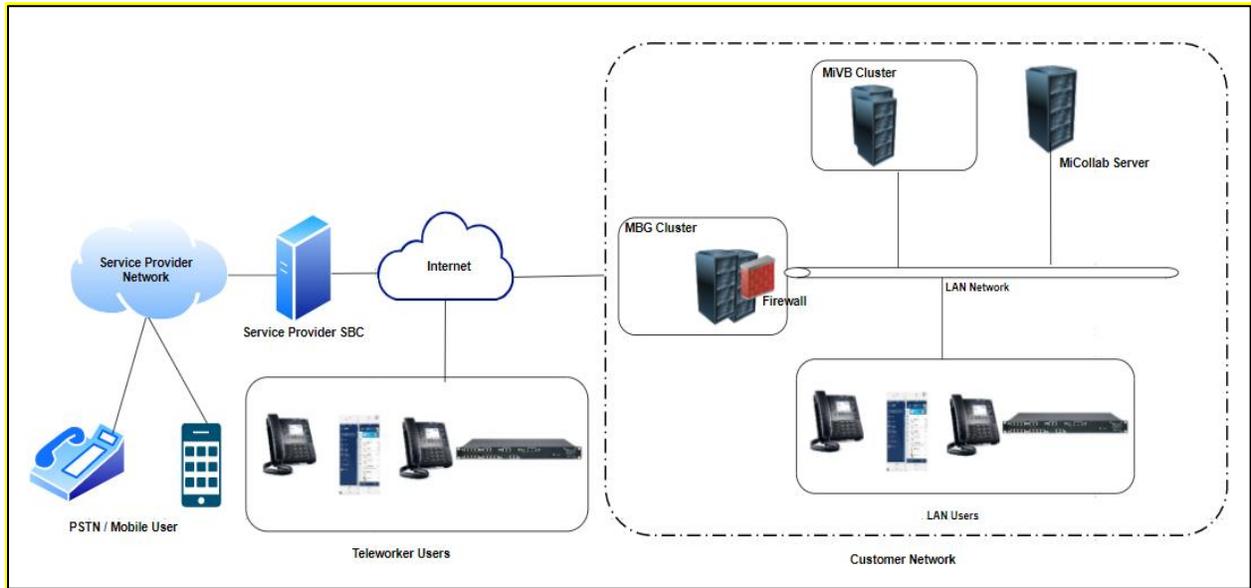


Figure 2 – Network Topology for Lineside

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.

The screenshot shows the Mitel MiVoice Business administration interface. The left sidebar contains a navigation menu with categories like Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, Users and Devices, Integrated Directory Services, Voice Mail, Call Routing, Music On Hold, Emergency Services Management, Property Management, and Maintenance and Diagnostics. The 'Licenses' category is selected and highlighted with a red box. The main content area is titled 'License and Option Selection on [mivb1]' and features a 'Change' button and action buttons for 'Print...', 'Import...', 'Export...', and 'Data...'. Below this is a table of license and option selections.

| License/Option | Current | Limit | Used | Available | Restrictions | Options |
|---|---------|-------|------|-----------|--------------|---------|
| ACD Active Agents | 0 | 20 | 0 | 20 | Unrestricted | No |
| HTML Applications | 0 | 1000 | 0 | 1000 | Unrestricted | Yes |
| Single Line Users | 0 | 200 | 0 | 200 | 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 | 30 | 30 | 0 | 20 | Unrestricted | Yes |
| Embedded Voice Mail PMS | 0 | No | 1 | 0 | Unrestricted | Yes |
| Trunking / Networking | | | | | | |
| Digital Links | 0 | 0 | 2 | 0 | Unrestricted | Yes |
| Compression | | 256 | 0 | 256 | Unrestricted | Yes |
| FAX Over IP (T.38) | | 4 | 0 | 4 | Unrestricted | Yes |
| SIP Trunks | 0 | 2000 | 0 | 2000 | Unrestricted | Yes |

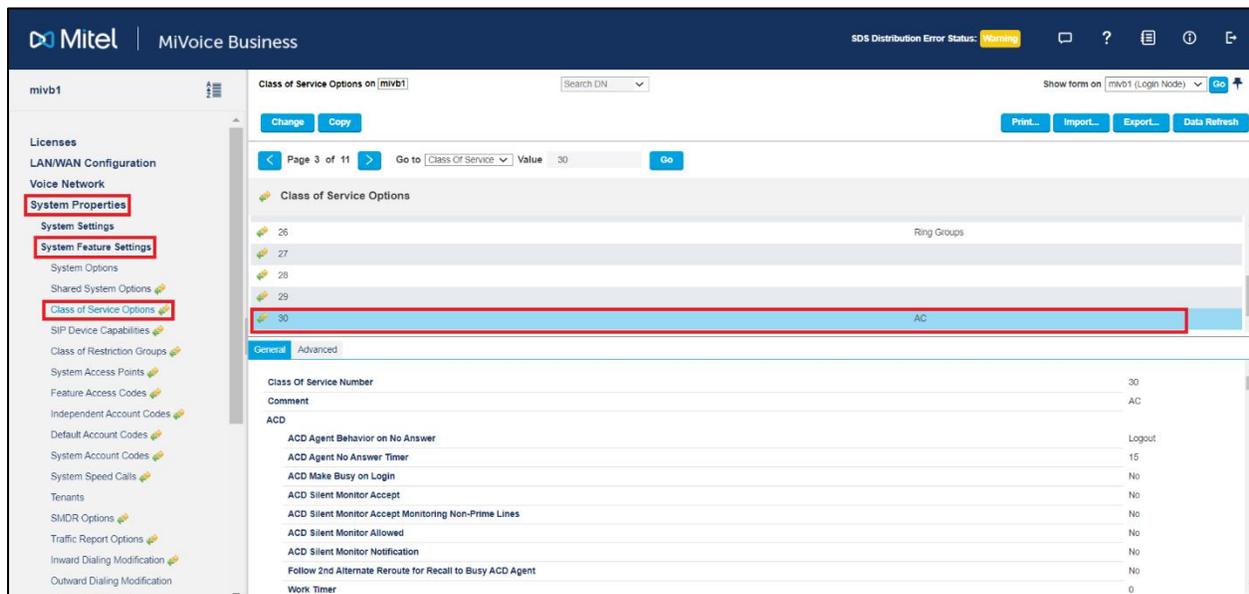
Figure 3 – License and Option Selection

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



The screenshot shows the Mitel MiVoice Business interface. The left sidebar contains a navigation menu with the following items: Licenses, LAN/WAN Configuration, Voice Network, System Properties (highlighted with a red box), System Settings, System Feature Settings (highlighted with a red box), System Options, Shared System Options, Class of Service Options (highlighted with a red box), SIP Device Capabilities, Class of Restriction Groups, System Access Points, Feature Access Codes, Independent Account Codes, Default Account Codes, System Account Codes, Tenants, SMDR Options, Traffic Report Options, Inward Dialing Modification, and Outward Dialing Modification. The main content area is titled "Class of Service Options on [mivb1]". It shows a list of Class of Service Options with columns for Class of Service Number and Comment. Option 30 is highlighted with a red box and has a comment of "AC". Below the list, the configuration details for option 30 are shown under the "General" tab. The details include: Class of Service Number: 30, Comment: AC, 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 Reroute for Recall to Busy ACD Agent: No, and Work Timer: 0.

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.

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

| | |
|---------------------------------------|-------------------------------------|
| Name | Mediant |
| Type | Other |
| FQDN or IP Address | 192.168.10.55 |
| 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 | 192.168.10.55 |
| External SIP Proxy Transport | UDP |
| External SIP Proxy Port | 5060 |
| SIP Registrar FQDN or IP Address | |
| SIP Registrar Transport | default |
| SIP Registrar Port | 0 |
| SIP Peer Status | Auto-Detect/Normal |

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

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.

| Change | |
|---|---|
| Trunk Attributes | |
| Trunk Service Number | 7 |
| Release Link Trunk | No |
| Call Recognition Service | Off |
| Direct Inward Dialing Service | <input checked="" type="radio"/> On |
| Caller Based Routing Service | <input checked="" type="radio"/> Off |
| Class of Service | 30 |
| 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 | AC |
| <input type="button" value="Save"/> <input type="button" value="Cancel"/> | |

Figure 6 – Trunk Attributes

SIP Peer Profile

The recommended connectivity via SIP Trunking does not require additional physical interfaces. IP/Ethernet connectivity is part of the base 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 Peer 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.

The screenshot shows the Mitel MiVoice Business configuration interface for a SIP Peer Profile. The interface is divided into a left sidebar and a main configuration area. The sidebar contains a navigation menu with categories: Licenses, LAN/WAN Configuration, Voice Network, System Properties, Hardware, Trunks, Trunk Attributes, IPXNET, SIP, DID Ranges for CPN Substitution, SIP Peer Profile, SIP Peer Profile Assignment by Incoming DID, SIP Peer Profile Called Party Inward Dialing Modification, SIP Peer Profile Calling Party Inward Dialing Modification, SIP Peer Profile Called Party Outward Dialing Modification, URI/Number Translation, Users and Devices, Integrated Directory Services, and Voice Mail. The 'SIP' and 'SIP Peer Profile' items are highlighted with red boxes. The main configuration area is titled 'SIP Peer Profile on MN69' and includes a search bar and a 'Show form on Exceeded Max Nodes' button. Below the title is a table with columns for Mediant, Mediant, No, 11, 1000, and 1. The configuration area is divided into tabs: Basic, Call Routing, Calling Line ID, SDP Options, Signaling and Header Manipulation, Timers, Key Press Event, and Profile Information. The 'Basic' tab is active and contains the following fields: SIP Peer Profile Label (Mediant), Network Element (Mediant), Local Account Information, Registration User Name, Address Type (IP Address: 192.168.10.69), Administration Options, Interconnect Restriction (1), Maximum Simultaneous Calls (9), Minimum Reserved Call Licenses (0), Outbound Proxy Server, SMDR Tag (0), Trunk Service (7), and Zone (1). The 'Outbound Proxy Server' and 'Trunk Service' fields are highlighted with red boxes.

| Authentication Options | |
|--|-------------------|
| User Name | |
| Password | ***** |
| Confirm Password | ***** |
| Authentication Option for Incoming Calls | No Authentication |
| Subscription User Name | |
| Subscription Password | ***** |
| Subscription Confirm Password | ***** |
| Gateway Options | |
| Digital Trunk Licenses | 0 |
| Maximum Digital/Analog Channels | 0 |

Figure 7 - SIP Peer Profile Assignment- Basic

| Basic | | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information |
|---|--|--------------|-----------------|-------------|-----------------------------------|--------|-----------------|---------------------|---------------------|
| Alternate Destination Domain Enabled | | | | | | | | | No |
| Alternate Destination Domain FQDN or IP Address | | | | | | | | | No |
| 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 8 - SIP Peer Profile Assignment- Call Routing

| Basic | | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information |
|--|--|--------------|-----------------|-------------|-----------------------------------|--------|-----------------|---------------------|---------------------|
| Default CPN | | | | | | | | | |
| Default CPN Name | | | | | | | | | |
| CPN Restriction | | | | | | | | | No |
| Override From Header with Default CPN | | | | | | | | | No |
| Public Calling Party Number Passthrough | | | | | | | | | No |
| Strip PNI | | | | | | | | | No |
| Use Diverting Party Number as Calling Party Number | | | | | | | | | No |
| Use Original Calling Party Number If Available | | | | | | | | | No |

Figure 9 - SIP Peer Profile Assignment- Calling Line ID

| Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information |
|-------|--------------|-----------------|---|-----------------------------------|--------|-----------------|---------------------|---------------------|
| | | | Allow Peer To Use Multiple Active M-Lines | | | | | Yes |
| | | | Allow Using UPDATE For Early Media Renegotiation | | | | | No |
| | | | Avoid Signaling Hold to the Peer | | | | | Yes |
| | | | AVP Only Peer | | | | | Yes |
| | | | Enable Mitel Proprietary SDP | | | | | No |
| | | | Force sending SDP in initial Invite message | | | | | Yes |
| | | | Force sending SDP in Initial Invite - Early Answer | | | | | No |
| | | | Ignore SDP Answers in Provisional Responses | | | | | No |
| | | | IP Media Default | | | | | ipv4 |
| | | | Limit to one Offer/Answer per INVITE | | | | | Yes |
| | | | NAT Keepalive | | | | | Yes |
| | | | Prevent Codec Selection on Answer | | | | | No |
| | | | Prevent the Use of IP Address 0.0.0.0 in SDP Messages | | | | | Yes |
| | | | Reject Call without telephone-event payload | | | | | No |
| | | | Renegotiate SDP To Enforce Symmetric Codec | | | | | No |
| | | | Repeat SDP Answer If Duplicate Offer is Received | | | | | No |
| | | | Restrict Audio Codec | | | | | No Restriction |
| | | | RTP Packetization Rate Override | | | | | No |
| | | | RTP Packetization Rate | | | | | 20ms |
| | | | Special handling of Offers in 2XX responses (INVITE) | | | | | No |
| | | | Suppress Use of SDP Inactive Media Streams | | | | | Yes |

Figure 10 - SIP Peer Profile Assignment- SDP Options

| Basic | Call Routing | Calling Line ID | SDP Options | Signaling and Header Manipulation | Timers | Key Press Event | Outgoing DID Ranges | Profile Information |
|-------|--------------|-----------------|---|-----------------------------------|--------|-----------------|---------------------|---------------------|
| | | | Trunk Group Label | | | | | |
| | | | Allow Display Update | | | | | No |
| | | | Build Contact Using Request URI Address | | | | | No |
| | | | De-register Using Contact Address not * | | | | | Yes |
| | | | Disable Reliable Provisional Responses | | | | | No |
| | | | Disable Use of User-Agent and Server Headers | | | | | No |
| | | | Discard Received P-Asserted-Identity Headers | | | | | No |
| | | | Domain for Trunk Context | | | | | |
| | | | 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 |
| <hr/> | | | | | | | | |
| 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 |
| <hr/> | | | | | | | | |
| 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 |
| <hr/> | | | | | | | | |
| Update | | | | | | | | |
| <hr/> | | | | | | | | |
| Index | DID Range | | CPN Substitution | | | | | |

Figure 14 – 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 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.

| SIP Peer Profile Assignment by Incoming DID | | |
|---|------------------------|---------|
| Incoming DID Range | SIP Peer Profile Label | Comment |
| 008067591215 | Mediant | |
| Incoming DID Range | 008067591215 | |
| SIP Peer Profile Label | Mediant | |
| Comment | | |

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).

| ARS Digit Modification Plans on mivb1 | | Search DN | Show form on mivb1 (Login No | | | |
|--|-----------------------------|----------------------------|-------------------------------------|--------------------------|---------------------------|---------------------------|
| Change | Change Page | Change All | Clear | Print... | Import... | Export... |
| < | Page 1 of 55 | > | Go to | Value | Go | |
| ARS Digit Modification Plans | | | | | | |
| | 3 | 2 | | | | |
| | 4 | 0 | | | | |
| | 5 | 0 | | | | |
| | 6 | 0 | | | | |
| | 7 | 1 | | | | |
| | 8 | 1 | | | | |
| | 9 | 1 | | | | |
| | 10 | 3 | | | | |
| | 11 | 0 | | | | |
| | 12 | 0 | | | | |
| | 13 | 0 | | | | |
| | 14 | 0 | | | | |
| | 15 | 3 | | | | |

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**).

Change

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 |
|---------------|----------------------------|------------------|--------------------|
| 456 | Unknown | Route | 15 |

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="456"/> | <input type="text" value=""/> |
| Number of Digits to Follow | <input type="text" value="Change to"/> | <input type="text" value="Unknown"/> | <input type="text" value="-"/> |
| Termination Type | <input type="text" value="Change to"/> | <input type="text" value="Route"/> | <input type="text" value="-"/> |
| Termination Number | <input type="text" value="Change to"/> | <input type="text" value="15"/> | <input type="text" value=""/> |

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).

The screenshot displays the Mitel MiVoice Business configuration interface for 'Local_125'. The left sidebar shows the navigation menu with 'Voice Network' and 'Fax Service Profiles' highlighted. The main content area is titled 'Fax Service Profiles on Local_125' and includes a search bar and a 'Change' button. Below this, the 'Inter-Zone Fax Profile' section shows the following settings:

- Maximum Fax Rate: 14400 (V.17, 14400bps)
- High Speed Redundancy: 1
- Low Speed Redundancy: 3
- Error Correction Mode (ECM): Disabled

Below the inter-zone profile is a table for 'Intra-Zone Fax Service Profiles' with the following data:

| Profile | Maximum Fax Rate | High Speed Redundancy | Low Speed Redundancy | Error Correction Mode | NSF Override | NSF Vendor Code Value | NSF Country Code Value | Label |
|---------|------------------------|-----------------------|----------------------|-----------------------|--------------|-----------------------|------------------------|-------|
| 1 | - | - | - | - | - | - | - | G.711 |
| 2 | 14400 (V.17, 14400bps) | 1 | 3 | Disabled | Disabled | - | - | T.38 |
| 3 | - | - | - | - | - | - | - | - |
| 4 | - | - | - | - | - | - | - | - |

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.

The screenshot displays the Mitel MiVoice Business configuration interface for 'Local_125'. The left sidebar shows a navigation menu with 'Voice Network' and 'Network Zones' highlighted in red. The main content area shows 'Network Zones on Local_125' with a search dropdown and buttons for 'Change', 'Change Page', and 'Clear'. Below this is a pagination control showing 'Page 1 of 50' and a 'Go to' field. The 'Network Zones' table is displayed with the following data:

| Zone ID | Intra-zone Compression | Group Zone | Intra-zone Fax Profile | Label | SMDR Tag | Time Zone | LBN Prefix |
|---------|------------------------|------------|------------------------|-------|----------|-----------|------------|
| 1 | No | | 2 | | | | |
| 2 | No | | 1 | | | | |
| 3 | No | | 1 | | | | |
| 4 | No | | 1 | | | | |
| 5 | No | | 1 | | | | |
| 6 | No | | 1 | | | | |

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.

The screenshot shows the Mitel MiVoice Business interface for system MN69. The 'Licenses' section is highlighted in the sidebar. The main content area is titled 'License and Option Selection' and includes a 'Change' button. Below this, there is a section for 'Online Licensing with the Application Management Center' showing the 'Application Record ID' as 67987345. A table displays the system type as 'Enterprise' with 'License Sharing' set to 'No' and a 'Hardware Identifier'. The primary table lists 'Licensed Options' with columns for 'Locally Consumed', 'Locally Allocated', 'Available for Allocation', and 'Purchased'. A sub-table under 'Local Limits' shows 'Licenses Allowed' and 'Can be Over Allocated' for each option.

| Licensed Options | Locally Consumed | Locally Allocated | Available for Allocation | Purchased | Local Limits | |
|---|------------------|-------------------|--------------------------|-----------|------------------|-----------------------|
| | | | | | Licenses Allowed | Can be Over Allocated |
| Users | | | | | | |
| IP Users | 71 | 100 | 20 | 120 | Unrestricted | Yes |
| External Hot Desk Users | 1 | 50 | 50 | 100 | Unrestricted | Yes |
| ACD Active Agents | 0 | 10 | 10 | 20 | Unrestricted | No |
| HTML Applications | 0 | 250 | 0 | 250 | Unrestricted | Yes |
| Single Line Users | 0 | 100 | 0 | 100 | Unrestricted | Yes |
| MiVoice Business Console Active Operators | 0 | 0 | 10 | 10 | Unrestricted | No |
| Multi-device Users | 0 | 100 | 0 | 100 | Unrestricted | Yes |
| Multi-device Suites | 0 | 0 | 0 | 0 | 0 | No |

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.

The screenshot displays the Mitel MiVoice Business configuration interface. The left sidebar contains a navigation menu with 'Users and Devices' highlighted. The main area shows 'User and Services Configuration on MN69' with a search for '1525' yielding one result: 'Generic SIP Phone Full Service'. Below this, a table lists 20 button configurations. The first two rows are highlighted with a red box, showing Button Number 1 and 2, both Multicall, with Directory Number 1525 and Ring Type Ring.

| Button Number | Label | Line Type | URL | Button Directory Number | Ring Type | MIXML Application Feature | Phone Application Feature | Float |
|---------------|-------|--------------|-----|-------------------------|-----------|---------------------------|---------------------------|-------|
| 1 | | Multicall | | 1525 | Ring | Not Assigned | | No |
| 2 | | Multicall | | 1525 | Ring | Not Assigned | | No |
| 3 | | Not Assigned | | | | Not Assigned | | No |
| 4 | | Not Assigned | | | | Not Assigned | | No |
| 5 | | Not Assigned | | | | Not Assigned | | No |
| 6 | | Not Assigned | | | | Not Assigned | | No |
| 7 | | Not Assigned | | | | Not Assigned | | No |
| 8 | | Not Assigned | | | | Not Assigned | | No |
| 9 | | Not Assigned | | | | Not Assigned | | No |
| 10 | | Not Assigned | | | | Not Assigned | | No |
| 11 | | Not Assigned | | | | Not Assigned | | No |
| 12 | | Not Assigned | | | | Not Assigned | | No |
| 13 | | Not Assigned | | | | Not Assigned | | No |
| 14 | | Not Assigned | | | | Not Assigned | | No |
| 15 | | Not Assigned | | | | Not Assigned | | No |
| 16 | | Not Assigned | | | | Not Assigned | | No |
| 17 | | Not Assigned | | | | Not Assigned | | No |
| 18 | | Not Assigned | | | | Not Assigned | | No |
| 19 | | Not Assigned | | | | Not Assigned | | No |
| 20 | | Not Assigned | | | | Not Assigned | | No |

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.

The screenshot displays the Mitel MiVoice Business web interface. The left sidebar shows the navigation menu with 'System Properties' and 'System Feature Settings' highlighted. The main content area is titled 'Class of Service Options on MN69'. It shows a list of Class of Service Options, with '40' selected and highlighted in blue. The '40' option is labeled 'AC Mediant'. Below the list, the 'General' tab is active, showing the following configuration details:

| Field | Value |
|---|------------|
| Class Of Service Number | 40 |
| Comment | AC Mediant |
| 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 Reroute for Recall to Busy ACD Agent | No |
| Work Timer | 0 |

| 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/SDN Calling/Called Number | No |
| Display ANI/SDN 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 |
| Ringling | | |
| | Allow Recall after Transfer | No |
| | Delay Ring Timer | 10 |
| | No Answer Recall Timer | 17 |
| | Ringling Line Select | No |
| | Ringling 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 |

| General | Advanced | |
|----------------------|--|-----|
| | Multiline Set Music | No |
| | Multiline Set On-hook Dialing | Yes |
| | Multiline Set Phonebook Allowed | Yes |
| | Non DID Extension | No |
| | ONS CLASS/CLIP: Set | No |
| | ONS/OPS Internal Ring Cadence for External Callers | No |
| | Override Interconnect Restriction on Transfer | No |
| | Recall If Transferred to Original Call Destination | No |
| | Redial Facilities | Yes |
| | Use Default Billable Number For Trunk Calls | No |
| | Voice Dial Preferred | No |
| | Voice Mail Softkey | No |
| Phonebook | | |
| | Phonebook Lookup - Default to User Location | No |
| | Phonebook Lookup - Display User Location | No |
| Record A Call | | |
| | Record-A-Call - Save Recording on Hang-up | No |
| | Record-A-Call - Start Automatic Incoming Call Recording | No |
| | Record-A-Call - Start Automatic Outgoing External Call Recording | No |
| | Record-A-Call Active | 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’.

The screenshot shows the Mitel MiVoice Business configuration interface. The left sidebar contains navigation options: Licenses, LAN/WAN Configuration, Voice Network (with sub-items System Properties and System Settings), and System Settings (with sub-items System Options, Shared System Options, Class of Service Options, SIP Device Capabilities, Class of Restriction Groups, System Access Points, Feature Access Codes, Independent Account Codes, and Default Account Codes). The main area displays the SIP Device Capabilities form for device 40. The form includes fields for SIP Device Capabilities Number (40) and Comment (AC Mediant). Under the Outbound Proxy Server section, the option 'Replace System based with Device based In-Call Features' is set to 'Yes' and is highlighted with a red box. Other options include 'Allow MWI Notifications without Subscription' (No), 'Enable Digit Collection in Busy Or Alerting State' (No), and 'TLS Only' (No).

| Basic | SDP Options | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press Event | Called Party Inward Dialing Modification | Record Information | Advanced |
|-------|---|-----------------------------------|------------------------|--------|-----------------|--|--------------------|----------|
| | Allow Device To Use Multiple Active M-Lines | | | | | | | Yes |
| | Allow Using UPDATE For Early Media Renegotiation | | | | | | | No |
| | AVP Only Device | | | | | | | Yes |
| | Enable Mitel Proprietary SDP | | | | | | | No |
| | Force sending SDP in initial Invite message | | | | | | | Yes |
| | Ignore SDP Answers in Provisional Responses | | | | | | | No |
| | IP Media Default | | | | | | | ipv4 |
| | Limit to one Offer/Answer per INVITE | | | | | | | No |
| | Prevent Codec Selection on Answer | | | | | | | No |
| | Prevent SDP Renegotiation If Peer Initiated Hold | | | | | | | No |
| | Prevent the Use of IP Address 0.0.0.0 in SDP Messages | | | | | | | Yes |
| | Renegotiate SDP To Enforce Symmetric Codec | | | | | | | No |
| | Repeat SDP Answer If Duplicate Offer is Received | | | | | | | No |
| | Send Answer only after renegotiation is complete | | | | | | | No |
| | Support CTI Hold/Retrieve | | | | | | | No |
| | Suppress Use of SDP Inactive Media Streams | | | | | | | Yes |

| | | | | | | | | |
|--|-------------|-----------------------------------|------------------------|--------|-----------------|--|--------------------|----------|
| Basic | SDP Options | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press Event | Called Party Inward Dialing Modification | Record Information | Advanced |
| Creator Date Created Created with Version SIP Device Vendor Notes | | | | | | | | |

| | | | | | | | | |
|------------------|-------------|-----------------------------------|------------------------|--------|-----------------|--|--------------------|----------|
| Basic | SDP Options | Signaling and Header Manipulation | Distinctive Ring Tones | Timers | Key Press Event | Called Party Inward Dialing Modification | Record Information | Advanced |
| Dial Plan | | | | | | | | |

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.

| Number | Intercept Number | Class of Service - Day | Class of Service - Night1 | Class of Service - Night2 | Class of Restriction - Day | Class of Restriction - Night1 | Class of Restriction - Night2 | Call Coverage Service Number | Default Acct. Code | Zone Assignment Method | Zone ID | SIP Device Capabilities |
|--------|------------------|------------------------|---------------------------|---------------------------|----------------------------|-------------------------------|-------------------------------|------------------------------|--------------------|------------------------|---------|-------------------------|
| 1502 | 1 | 15 | 15 | 15 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1503 | 1 | 15 | 15 | 15 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1505 | 1 | 15 | 15 | 15 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 15 |
| 1509 | 1 | 15 | 15 | 15 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1510 | 1 | 15 | 15 | 15 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 30 |
| 1511 | 1 | 30 | 30 | 30 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 30 |
| 1512 | 1 | 30 | 30 | 30 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 30 |
| 1519 | 1 | 40 | 40 | 40 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1520 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1521 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1522 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1525 | 1 | 40 | 40 | 40 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 40 |
| 1526 | 1 | 40 | 40 | 40 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 40 |
| 1900 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |
| 1910 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | 1 | Default | 1 | 1 |

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.

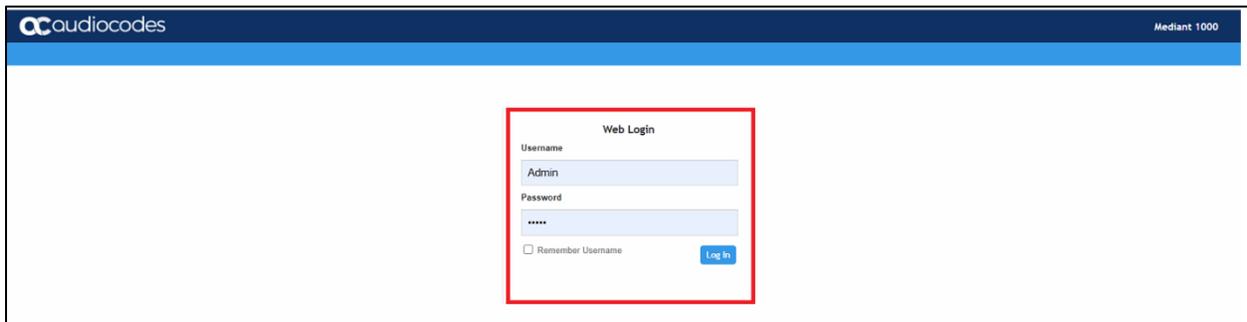


Figure 27 – Web Login

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).

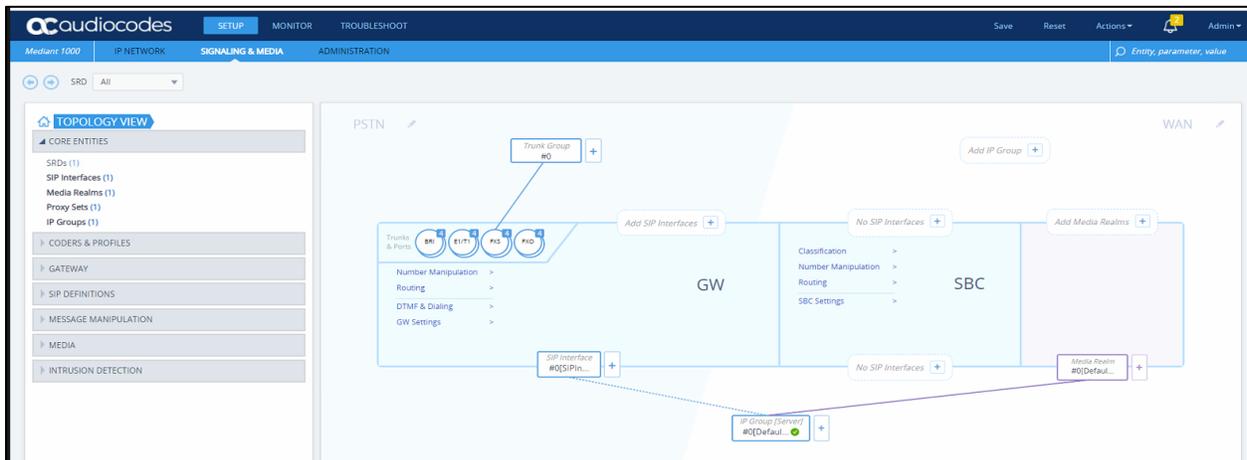


Figure 28 – Home Screen

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.

The screenshot shows the AudioCodes Mediant 1000 web interface. The top navigation bar includes 'SETUP', 'MONITOR', and 'TROUBLESHOOT'. The left sidebar shows 'IP NETWORK' selected, with 'IP Interfaces (Voice)' highlighted. The main content area is titled 'IP Interfaces (Voice)' and contains a configuration form with the following sections:

- GENERAL:** Index (0), Name (Voice), Application Type (OAMP + Media + Control), Ethernet Device (#0 [vlan 1]).
- IP ADDRESS:** Interface Mode (IPv4 Manual), IP Address (192.168.10.50), Prefix Length (24), Default Gateway (192.168.10.1).
- DNS:** Primary DNS (192.168.10.111), Secondary DNS (empty).

At the bottom of the form, there is a warning message: "Changes to the network interface will stop all services running on the interface, in particular, connectivity with the device's management interface and current calls". Below this are 'Cancel' and 'APPLY' buttons. The 'APPLY' button is highlighted with a red box. At the very bottom, there is a link for 'IP Interface Status Table >>'.

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.

The screenshot shows the 'Coder Groups' configuration page in the AudioCodes Mediant 1000 web interface. The interface includes a top navigation bar with 'SETUP', 'MONITOR', and 'TROUBLESHOOT' tabs. The left sidebar contains a 'TOPOLOGY VIEW' section with a tree structure including 'CORE ENTITIES', 'CODERS & PROFILES', 'IP Profiles (1)', 'Tel Profiles (0)', 'Coder Settings', 'Coder Groups', 'GATEWAY', 'SIP DEFINITIONS', 'MESSAGE MANIPULATION', 'MEDIA', and 'INTRUSION DETECTION'. The main content area is titled 'Coder Groups' and features a search field for 'Coder Group Name' with the value '0: AudioCodersGroups_0' and a 'Delete Group' button. Below this is a table with the following columns: Coder Name, Packetization Time, Rate, Payload Type, Silence Suppression, and Coder Specific. The table contains three rows of data:

| Coder Name | Packetization Time | Rate | Payload Type | Silence Suppression | Coder Specific |
|------------|--------------------|------|--------------|---------------------|----------------|
| G.711A-law | 20 | 64 | 8 | Disabled | |
| G.711U-law | 20 | 64 | 0 | Disabled | |
| G.729 | 20 | 8 | 18 | Disabled | |

At the bottom right of the page, there are 'Cancel' and 'APPLY' buttons. A red box highlights the table and the 'APPLY' button.

Figure 30 – Coder Groups

Media Settings

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

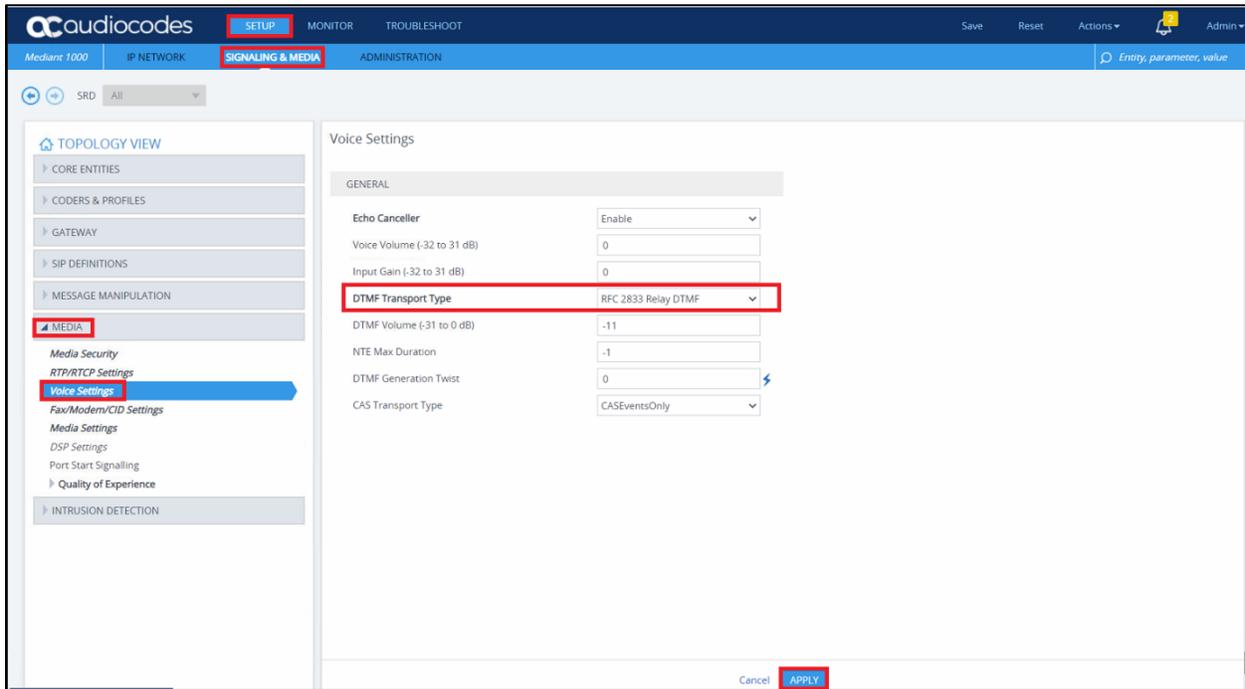


Figure 31 – Media Settings

DTMF Settings

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

The screenshot displays the 'DTMF & Dialing' configuration page in the AudioCodes Mediant 1000 Gateway interface. The left sidebar shows the navigation tree with 'DTMF & Dialing' selected. The main content area is divided into 'GENERAL' and 'ADVANCED' sections. The 'GENERAL' section includes fields for 'Max Digits In Phone Num' (30), 'Inter Digit Timeout for Overlap Dialing [sec]' (4), 'Declare RFC 2833 in SDP' (Yes), '1st Tx DTMF Option' (RFC 2833), '2nd Tx DTMF Option' (), and 'RFC 2833 Payload Type' (101). The 'ADVANCED' section includes 'Hook-Flash Option' (Not Supported), 'Digit Mapping Rules' (), 'Dial Plan Index' (-1), 'Dial Tone Duration [sec]' (16), 'Hotline Dial Tone Duration [sec]' (16), 'Enable Special Digits' (Disable), 'Min Routing Overlap Digits' (1), and 'ISDN Overlap IP-to-Tel Dialing' (Disable). The 'DIGIT PATTERNS' section contains a list of forwarding rules, all of which are currently empty.

| Section | Parameter | Value |
|--------------------------------|--|---------------|
| GENERAL | Max Digits In Phone Num | 30 |
| | Inter Digit Timeout for Overlap Dialing [sec] | 4 |
| | Declare RFC 2833 in SDP | Yes |
| | 1st Tx DTMF Option | RFC 2833 |
| | 2nd Tx DTMF Option | |
| | RFC 2833 Payload Type | 101 |
| ADVANCED | Hook-Flash Option | Not Supported |
| | Digit Mapping Rules | |
| | Dial Plan Index | -1 |
| | Dial Tone Duration [sec] | 16 |
| | Hotline Dial Tone Duration [sec] | 16 |
| | Enable Special Digits | Disable |
| | Min Routing Overlap Digits | 1 |
| ISDN Overlap IP-to-Tel Dialing | Disable | |
| DIGIT PATTERNS | Forward on Busy Digit Pattern (Internal) | |
| | Forward on No Answer Digit Pattern (Internal) | |
| | Forward on Do Not Disturb Digit Pattern (Internal) | |
| | Forward on No Reason Digit Pattern (Internal) | |
| | Forward on Busy Digit Pattern (External) | |
| | Forward on No Answer Digit Pattern (External) | |
| | Forward on Do Not Disturb Digit Pattern (External) | |
| | Forward on No Reason Digit Pattern (External) | |
| | Internal Call Digit Pattern | |
| | External Call Digit Pattern | |
| Disconnect Call Digit Pattern | | |
| Digit To Ignore Digit Pattern | | |

Figure 32 – DTMF Settings

DTMF Supplementary Services Settings

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

The screenshot displays the AudioCodes Mediant 1000 Gateway configuration interface. The left-hand navigation pane shows a tree structure with 'Supplementary Services Settings' highlighted. The main content area is titled 'Supplementary Services Settings' and is divided into several sections:

- GENERAL:** Includes settings for 'Enable Caller ID' (set to Enable), 'Answer Supervision' (No), 'Flash Keys Sequence Style' (Flash hook), 'Flash Keys Sequence Timeout' (2000), 'Enable NRT Subscription' (Disable), 'NRT Subscribe Retry Time' (120), 'Generate Metering Tones' (Disable), 'AoC Support' (Disable), 'Reminder Ring' (Enable), and 'Line Transfer Mode' (None).
- CALL HOLD:** Includes 'Enable Hold' (set to Enable), 'Enable Hold to ISDN' (Disable), 'Hold Format' (0.0.0.0), 'Hold Timeout' (-1), 'Call Hold Reminder Ring Timeout' (30), and 'Maximum simultaneous streaming calls' (0).
- TRANSFER:** Includes 'Enable Transfer' (Enable), 'Transfer Prefix', and 'Blind Transfer'.
- MESSAGE WAITING INDICATOR:** Includes 'Enable MWI' (Enable), 'Subscribe to MWI' (Yes), 'MWI Server IP Address' (192.168.10.69), 'MWI Subscribe Expiration Time' (300), 'MWI Subscribe Retry Time' (120), 'MWI Analog Lamp' (Enable), 'MWI Display' (Enable), 'MWI Server Transport Type' (UDP), 'Stutter Tone Duration' (2000), 'Subscription Mode' (Per Endpoint), 'AS Subscribe IP Group ID' (-1), 'MWI Source Number', 'Voice Mail Interface' (NONE), 'MWI Off Digit Pattern', and 'MWI On Digit Pattern'.

The 'Enable Hold' setting in the CALL HOLD section and the entire MESSAGE WAITING INDICATOR section are highlighted with red boxes. The interface also shows 'Cancel' and 'APPLY' buttons at the bottom right.

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.

The screenshot displays the 'SIP Definitions General Settings' configuration page in the AudioCodes Mediant 1000 web interface. The interface is divided into a left-hand navigation pane and a main content area. The left-hand pane shows a 'TOPOLOGY VIEW' with several categories: 'CORE ENTITIES', 'CODERS & PROFILES', 'GATEWAY', and 'SIP DEFINITIONS'. The 'SIP DEFINITIONS' category is expanded, and 'SIP Definitions General Settings' is selected. The main content area displays various settings for SIP Definitions, organized into sections: GENERAL, GATEWAY SESSION EXPIRES, GATEWAY SETTINGS, DISCONNECT SUPERVISION, and MICROSOFT PRESENCE. Each section contains a list of settings with their current values and dropdown menus for selection.

| Section | Setting | Value | |
|---------------------------------|---------------------------------|--------------------------------------|------------|
| GENERAL | Send Reject (503) upon Overload | Enable | |
| | Retry-After Time | 0 | |
| | Fake Retry After | 0 | |
| | Remote Management by SIP NOTIFY | Disable | |
| | X-Channel Header | Disable | |
| GATEWAY SESSION EXPIRES | Session-Expires Time | 1800 | |
| | Minimum Session-Expires | 1800 | |
| | Session Expires Method | re-INVITE | |
| | Session Expires Disconnect Time | 32 | |
| | DISCONNECT SUPERVISION | Broken Connection Mode | Disconnect |
| | | Broken Connection Timeout [100 msec] | 100 |
| | GATEWAY SETTINGS | PRACK Mode | Supported |
| Early 183 | | Disable | |
| 183 Message Behavior | | Progress | |
| 3xx Behavior | | Forward | |
| Call Transfer using re-INVITES | | Disable | |
| First Call Ringback Tone ID | | -.1 | |
| Delayed Offer | | Disable | |
| Source Header For Called Number | | use RequestURI header | |
| Verify Received VIA | | Disable | |
| Reject Cancel after Connect | | Disable | |
| MICROSOFT PRESENCE | Presence Publish IP Group ID | -.1 | |
| | Microsoft Presence Status | Disable | |

Figure 34 – General Settings

Trunk Configuration (FXO)

Proxy & Registration

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

The screenshot displays the AudioCodes Mediant 1000 configuration interface. The left sidebar shows a navigation tree with 'SIP DEFINITIONS' selected and 'Proxy & Registration' highlighted. The main content area is titled 'Proxy & Registration' and is divided into several sections:

- GENERAL:** Includes settings for Redundancy Mode (Parking), Proxy IP List Refresh Time (60), Proxy DNS Query Type (A-Record), Number of RTT Before Hot Swap (3), Use Proxy IP as Host (Enable), User Information Usage (Enable), Add Empty Authorization Header (Disable), Gateway Name, Use Gateway Name for OPTIONS (No), and Challenge Caching Mode (None).
- REGISTRATION:** Includes Registration Time (180), Re-registration Timing (N) (90), Registration Retry Time (30), Max Registration Backoff Time (sec) (0), Registration Time Threshold (0), Re-register On INVITE Failure (Enable), Re-register On Connection Failure (Enable), Gateway Registration Name, GRUU (Disable), and Max Generated Register Rate (30).
- GATEWAY PROXY:** Includes Use Default Proxy (Use Proxy), Proxy Set Table, Proxy Name, Prefer Routing Table (No), Use Routing Table for Host Names and Profiles (Disable), Always Use Proxy (Disable), and Enable Fallback to Routing Table (Disable).
- AUTHENTICATION:** Includes User Name, Password, and Cnonce (Default_Cnonce).
- GATEWAY AUTHENTICATION:** Includes Authentication Mode (Per FQDN).
- GATEWAY REGISTRATION:** Includes Enable Registration (Enable), Registrar Name, Registrar IP Address (192.168.10.69), Registrar Transport Type (UDP), and Set Out-Of-Service On Registration Failure (Disable).

The 'GATEWAY REGISTRATION' section is highlighted with a red box in the original image. At the bottom right, there are 'Cancel' and 'APPLY' buttons.

Figure 35 – Proxy & Registration

Trunk Groups

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

The screenshot shows the AudioCodes Mediant 1000 Gateway configuration interface. The top navigation bar includes 'Mediant 1000', 'IP NETWORK', 'SIGNALING & MEDIA', and 'ADMINISTRATION'. The 'SIGNALING & MEDIA' tab is active. The left sidebar shows a 'TOPOLOGY VIEW' with 'CORE ENTITIES' and 'CODERS & PROFILES'. Under 'CODERS & PROFILES', 'GATEWAY' is expanded, and 'Trunks & Groups' is selected. The main area displays the 'Trunk Group Table' with the following columns: GROUP INDEX, MODULE, FROM TRUNK, TO TRUNK, CHANNELS, PHONE NUMBER, TRUNK GROUP ID, and TEL PROFILE NAME. The table has 10 rows. The first row is highlighted with a red border. The 'PHONE NUMBER' column has a 'Disable' button and a '1:10' dropdown menu. The 'TRUNK GROUP ID' and 'TEL PROFILE NAME' columns have dropdown menus set to 'None'. The 'FROM TRUNK' and 'TO TRUNK' columns have dropdown menus. The 'CHANNELS' column has a text input field with the value '1'. The 'MODULE' column has a dropdown menu set to 'Module 3 FXS'. The 'GROUP INDEX' column has values from 1 to 10. The 'PHONE NUMBER' column has a 'Disable' button and a '1:10' dropdown menu. The 'TRUNK GROUP ID' and 'TEL PROFILE NAME' columns have dropdown menus set to 'None'. The 'FROM TRUNK' and 'TO TRUNK' columns have dropdown menus. The 'CHANNELS' column has a text input field with the value '1'. The 'MODULE' column has a dropdown menu set to 'Module 3 FXS'. The 'GROUP INDEX' column has values from 1 to 10.

| GROUP INDEX | MODULE | FROM TRUNK | TO TRUNK | CHANNELS | PHONE NUMBER | TRUNK GROUP ID | TEL PROFILE NAME |
|-------------|--------------|------------|----------|----------|--------------|----------------|------------------|
| 1 | Module 3 FXS | | | 1 | | | None |
| 2 | | | | | | | None |
| 3 | | | | | | | None |
| 4 | | | | | | | None |
| 5 | | | | | | | None |
| 6 | | | | | | | None |
| 7 | | | | | | | None |
| 8 | | | | | | | None |
| 9 | | | | | | | None |
| 10 | | | | | | | None |

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.

The screenshot shows the AudioCodes Mediant 1000 configuration interface. The main area displays the 'Automatic Dialing' configuration. A table lists several dialing rules, with the first rule (index 0) highlighted. Below the table, a detailed view of the selected rule is shown, including 'GENERAL' and 'AUTO DIALING' sections.

| INDEX | MODULE | PORT | PORT TYPE | AUTO DIAL STATUS | DESTINATION PHONE NUMBER |
|-------|--------|------|-----------|------------------|--------------------------|
| 0 | 2 | 1 | FXO | enable | 4000 |
| 1 | 2 | 2 | FXO | enable | |
| 2 | 2 | 3 | FXO | enable | |
| 3 | 2 | 4 | FXO | enable | |
| 4 | 3 | 1 | FXS | enable | |
| 5 | 3 | 2 | FXS | enable | |
| 6 | 3 | 3 | FXS | enable | |
| 7 | 3 | 4 | FXS | enable | |

| GENERAL | | AUTO DIALING | |
|-----------|-----|-----------------------------|--------|
| Module | 2 | Auto Dial Status | enable |
| Port | 1 | Destination Phone Number | 4000 |
| Port Type | FXO | HoldTime Dial-Tone Duration | -1 |

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.

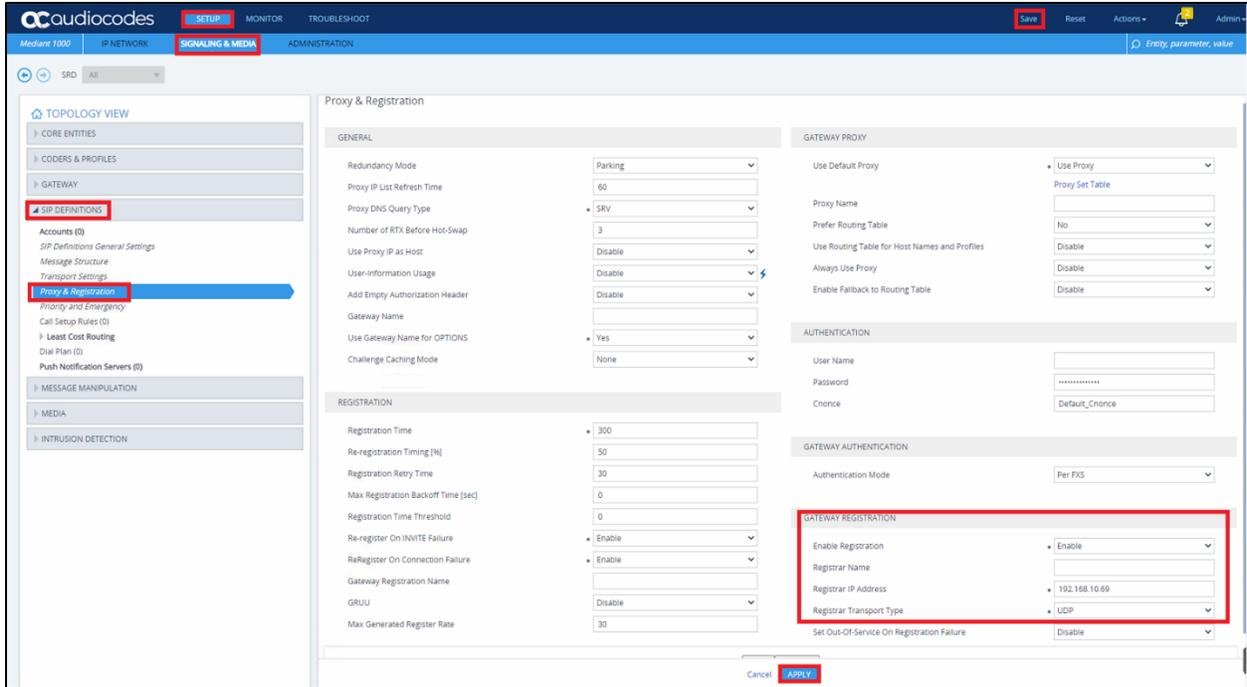


Figure 38 – Proxy & Registration

Trunk Groups

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

The screenshot displays the Audiocodes Mediant 1000 configuration interface. The main area is titled "Trunk Group Table" and contains a table with the following columns: GROUP INDEX, MODULE, FROM TRUNK, TO TRUNK, CHANNELS, PHONE NUMBER, TRUNK GROUP ID, and TEL PROFILE NAME. The first row of the table is highlighted with a red border and contains the following data: GROUP INDEX: 1, MODULE: Module 3 FXS, FROM TRUNK: (empty), TO TRUNK: (empty), CHANNELS: 1, PHONE NUMBER: 1525, TRUNK GROUP ID: (empty), and TEL PROFILE NAME: None. The interface also includes a left sidebar with navigation options such as "GATEWAY", "Trunk Settings", and "Trunk Groups". The top navigation bar shows "SETUP", "MONITOR", and "TROUBLESHOOT" tabs. The "Trunk Groups" option in the sidebar is highlighted in blue. The "Trunk Group Table" title is also highlighted in blue. The "Trunk Group Table" title is also highlighted in blue. The "Trunk Group Table" title is also highlighted in blue.

| GROUP INDEX | MODULE | FROM TRUNK | TO TRUNK | CHANNELS | PHONE NUMBER | TRUNK GROUP ID | TEL PROFILE NAME |
|-------------|--------------|------------|----------|----------|--------------|----------------|------------------|
| 1 | Module 3 FXS | | | 1 | 1525 | | None |
| 2 | | | | | | | None |
| 3 | | | | | | | None |
| 4 | | | | | | | None |
| 5 | | | | | | | None |
| 6 | | | | | | | None |
| 7 | | | | | | | None |
| 8 | | | | | | | None |
| 9 | | | | | | | None |
| 10 | | | | | | | None |

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.

The screenshot shows the Audiocodes Mediant 1000 web interface. The navigation menu on the left is expanded to 'GATEWAY' > 'Analog Gateway' > 'Authentication (8)'. The main configuration area is titled 'Authentication' and contains two tabs: 'GENERAL' and 'CREDENTIALS'. The 'CREDENTIALS' tab is active, showing the following fields:

| Field | Value |
|-----------|-------|
| Index | 4 |
| Module | 3 |
| Port | 1 |
| Port Type | FXS |
| User Name | 1525 |
| Password | |

At the bottom of the configuration area, there are 'Cancel' and 'APPLY' buttons.

Figure 40 – Authentication Settings

Message Manipulation

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

The screenshot shows the Audiocodes Mediant 1000 web interface. The navigation menu on the left is expanded to 'MESSAGE MANIPULATION' > 'Message Manipulations (1)'. The main configuration area is titled 'Message Manipulations [Fix R-URI]' and contains three tabs: 'GENERAL', 'ACTION', and 'MATCH'. The 'ACTION' tab is active, showing the following fields:

| Field | Value |
|---------------------|----------------------------------|
| Index | 0 |
| Name | Fix R-URI |
| Manipulation Set ID | 0 |
| Row Role | Use Current Condition |
| Action Subject | Header Request-URI URL Host Name |
| Action Type | Modify |
| Action Value | \$2 |

The 'MATCH' tab shows the following fields:

| Field | Value |
|--------------|--|
| Message Type | Register |
| Condition | Header Request-URI URL Host Name regex (sip;x.*) |

At the bottom of the configuration area, there are 'Cancel' and 'APPLY' buttons.

Figure 41 – Message manipulation

Caller Display Information Settings

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

Caller Display Information (8)

| INDEX | MODULE | PORT | PORT TYPE | DISPLAY STRING | PRESENTATION |
|-------|--------|------|-----------|----------------|--------------|
| 0 | 2 | 1 | FXO | | Allowed |
| 1 | 2 | 2 | FXO | | Allowed |
| 2 | 2 | 3 | FXO | | Allowed |
| 3 | 2 | 4 | FXO | | Allowed |
| 4 | 3 | 1 | FXS | | Allowed |
| 5 | 3 | 2 | FXS | | Allowed |
| 6 | 3 | 3 | FXS | | Allowed |
| 7 | 3 | 4 | FXS | | Allowed |

#0

| GENERAL | | CALLER DISPLAY | |
|-----------|-----|----------------|---------|
| Module | 2 | Display String | |
| Port | 1 | Presentation | Allowed |
| Port Type | FXO | | |

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.

Caller ID Permissions (8)

| INDEX | MODULE | PORT | PORT TYPE | CALLER ID |
|-------|--------|------|-----------|-----------|
| 0 | 2 | 1 | FXO | Enable |
| 1 | 2 | 2 | FXO | Enable |
| 2 | 2 | 3 | FXO | Enable |
| 3 | 2 | 4 | FXO | Enable |
| 4 | 3 | 1 | FXS | Enable |
| 5 | 3 | 2 | FXS | Enable |
| 6 | 3 | 3 | FXS | Enable |
| 7 | 3 | 4 | FXS | Enable |

#0

| GENERAL | | CALLER ID | |
|-----------|-----|-----------|--------|
| Module | 2 | Caller ID | Enable |
| Port | 1 | | |
| Port Type | FXO | | |

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.

The screenshot shows the Audiocodes Mediant 1000 web interface. The navigation menu on the left includes 'TOPLOGY VIEW', 'CORE ENTITIES', 'CODERS & PROFILES', 'GATEWAY', 'SIP DEFINITIONS', 'MESSAGE MANIPULATION', 'MEDIA', and 'INTRUSION DETECTION'. The 'MEDIA' section is expanded, and 'Fax/Modem/CID Settings' is selected. The main content area displays the 'Fax/Modem/CID Settings' configuration page. The 'GENERAL' section has 'Fax Transport Mode' set to 'Bypass'. Other settings include 'Caller ID Transport Type' (Mute), 'Caller ID Type' (Standard Bellcore), and various Modem Transport Types (V.21, V.22, V.23, V.32, V.34) all set to 'Disable'. The 'FAX RELAY' section includes 'Fax Relay Redundancy Depth' (0), 'Fax Relay Enhanced Redundancy Depth' (4), 'Fax Relay ECM Enable' (Enable), 'Fax Relay Max Rate (bps)' (14400bps), and 'Fax Relay Rx/Tx Timeout (sec)' (10). The 'FAX/MODEM BYPASS' section includes 'Fax/Modem Bypass Coder Type' (G711Alaw_64), 'Fax/Modem Bypass Packing Factor' (1), 'Fax Bypass Output Gain' (0), and 'Modem Bypass Output Gain' (0). The 'GATEWAY SETTINGS' section has 'Enable Fax Re-Routing' set to 'Disable'. The 'APPLY' button is highlighted in red.

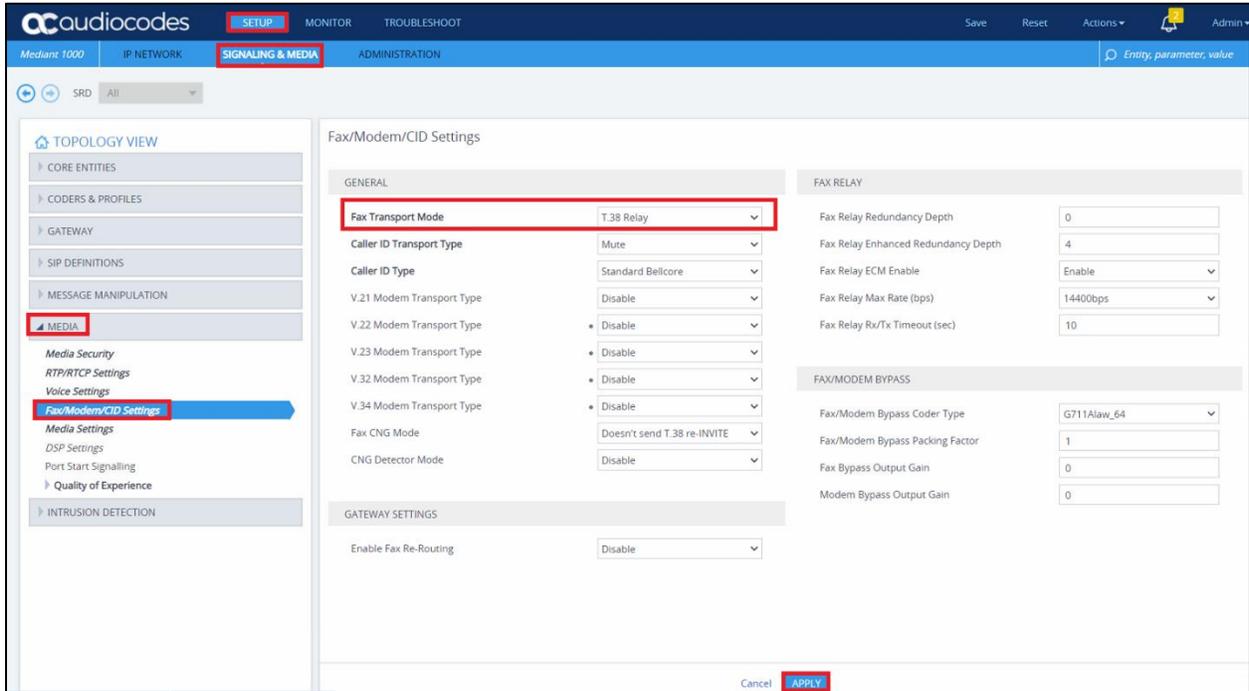
After that go to Coders and Profiles > IP Profiles > Fax Signaling Method set to G711 Transport > Apply.

The screenshot shows the Audiocodes Mediant 1000 web interface. The navigation menu on the left includes 'TOPLOGY VIEW', 'CORE ENTITIES', 'CODERS & PROFILES', 'GATEWAY', 'SIP DEFINITIONS', 'MESSAGE MANIPULATION', 'MEDIA', and 'INTRUSION DETECTION'. The 'CODERS & PROFILES' section is expanded, and 'IP Profiles (1)' is selected. The main content area displays the 'IP Profiles [Fax]' configuration page. The 'VOICE' section includes 'Dynamic Jitter Buffer Minimum Delay [msec]' (10), 'Dynamic Jitter Buffer Optimization Factor' (10), 'Jitter Buffer Max Delay [msec]' (250), 'Echo Canceler' (Line), 'Input Gain (-32 to 31 dB)' (0), and 'Voice Volume (-32 to 31 dB)' (0). The 'GATEWAY FAX AND MODEM' section has 'Fax Signaling Method' set to 'G.711 Transport'. Other settings include 'Is DTMF Used' (Enable), 'First Tx DTMF Option' (RFC 2833), 'Second Tx DTMF Option' (Supported), 'Rx DTMF Option' (Supported), 'CNG Detector Mode' (Disable), 'Vxx Modem Transport Type' (Disable), and 'NSE Mode' (Disable). The 'ANSWER MACHINE DETECTION' section has 'AMD Mode' set to 'Don't Disconnect'. The 'APPLY' button is highlighted in red.

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.

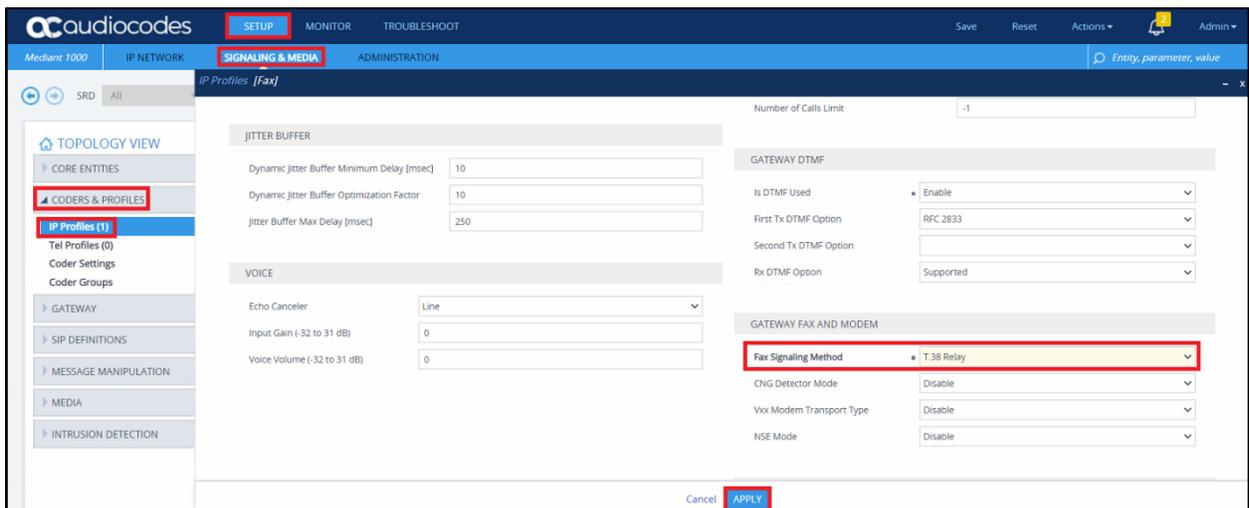


Figure 44 – Fax Settings

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)

The screenshot shows the Mitel Standard Linux web interface. The 'Network' menu is open, and 'ICPs' is selected. The main content area displays a table of ICPs. The 'MVB99' entry is highlighted with a red border. The table has the following columns: Default for MNet, Default for SIP, Name, Hostname or IP address, Type, Installer password, SIP capabilities, Indirect call recording capable, Associated connectors, Associated sets (MNet/SIP), and Associated trunk rules (privsec).

| Default for MNet | Default for SIP | Name | Hostname or IP address | Type | Installer password | SIP capabilities | Indirect call recording capable | Associated connectors | Associated sets (MNet/SIP) | Associated trunk rules (privsec) | | | |
|----------------------------------|----------------------------------|-------------------|------------------------|--------------------|--------------------|-------------------|---------------------------------|-----------------------|----------------------------|----------------------------------|--|--|--|
| <input type="radio"/> | <input checked="" type="radio"/> | 5000 | 192.168.10.159 | MiVoice 5000 | | UDP | ✗ | ✗ | 0/3 | 1/0 | | | |
| <input type="radio"/> | <input type="radio"/> | A_MVO400 | 192.168.10.139 | MiVoice Office 400 | | UDP TCP TLS | ✗ | ✓ | 0/5 | 1/1 | | | |
| <input type="radio"/> | <input type="radio"/> | A_Mtone | 192.168.10.172 | MiVoice MX-ONE | | UDP TCP TLS | ✗ | ✗ | 0/3 | 0/0 | | | |
| <input checked="" type="radio"/> | <input type="radio"/> | mivb_132 | 192.168.10.132 | MiVoice Business | | UDP TCP TLS | ✗ | ✗ | 1/1 | 0/0 | | | |
| <input type="radio"/> | <input type="radio"/> | MVB_192.168.10.74 | 192.168.10.74 | MiVoice Business | | UDP TCP TLS | ✗ | ✗ | 0/0 | 0/1 | | | |
| <input type="radio"/> | <input type="radio"/> | MVB99 | 192.168.10.60 | MiVoice Business | | UDP TCP TLS | ✗ | ✗ | 9/11 | 7/3 | | | |
| <input type="radio"/> | <input type="radio"/> | MVB93 | 192.168.10.93 | MiVoice Business | | UDP TCP TLS | ✗ | ✗ | 0/1 | 0/0 | | | |
| <input type="radio"/> | <input type="radio"/> | MVB_95 | 192.168.10.95 | MiVoice Business | | UDP TCP TLS | ✗ | ✗ | 1/4 | 2/0 | | | |

Figure 45 – Setting up Default ICP

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

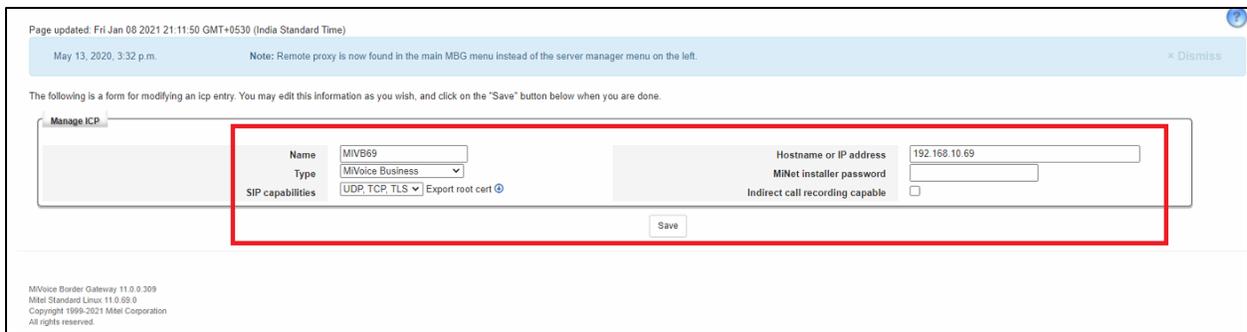


Figure 46 – Setting up Default ICP

Adding SIP devices

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

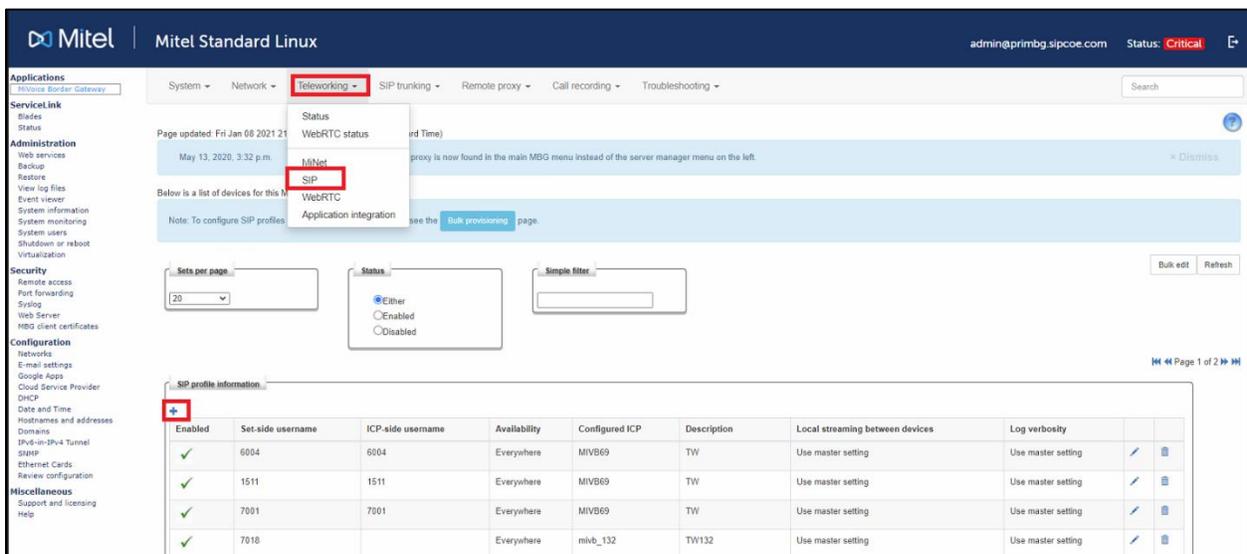


Figure 47 – Creating SIP User

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.

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

Page updated: Fri Jan 08 2021 21:16:54 GMT+0530 (India Standard Time)

May 13, 2020, 3:32 p.m. Note: Remote proxy is now found in the main MBG menu instead of the server manager menu on the left. [Dismiss](#)

Manage SIP profile

Enabled

Set-side username

ICP-side username

Configured ICP

Set-side password [Change set-side password](#)

Confirm set-side password

ICP-side password [Change icp-side password](#)

Confirm icp-side password

PRACK support

Heartbeat interval

Description

Set-side RTP security

Inbound

Outbound

Preferred cipher

Local streaming between device calls

Enable Detailed Jitter Log

Codec support

Tone injection

Options keepalives

Challenge methods [Override](#)

Availability

ICP-side RTP security

Inbound

Outbound

Preferred cipher

Log verbosity

RTP Framesize

Use master

[Save](#)

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 | | |
|-------------------------------------|-------------------|-------------------|--------------|----------------|---------------|---------------------------------|--------------------|-------------------|-------------------|
| <input checked="" type="checkbox"/> | 6004 | 6004 | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 1511 | 1511 | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 7001 | 7001 | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 7018 | | Everywhere | mivb_132 | TW132 | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 2030 | 2030 | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 1999 | | Everywhere | MIVB_95 | tw | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 1009 | | Everywhere | MIVB93 | TW_93 | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 3000 | 3000 | Everywhere | 5000 | TW3001 | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 1504 | 1504 | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 5006 | 5006 | Everywhere | A_Mxone | mxone TW_5006 | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 1988 | | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 2002 | 2002-XS | Everywhere | A_MIVO400 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 5003 | 5003 | Everywhere | A_Mxone | Mxone 5003 | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 1525 | 1525 | Everywhere | MIVB69 | TW | Use master setting | Use master setting | ✎ | 🗑 |
| <input checked="" type="checkbox"/> | 5009 | 5009 | Everywhere | A_Mxone | TW5009 | Use master setting | Use master setting | ✎ | 🗑 |

Figure 49 –SIP device Information

SIP Settings

SIP options

| SIP support | Protocols | Access profile |
|--------------------------|---|----------------|
| <input type="checkbox"/> | UDP <input checked="" type="checkbox"/> | Public |
| <input type="checkbox"/> | TCP <input checked="" type="checkbox"/> | Public |
| <input type="checkbox"/> | TCP/TLS <input checked="" type="checkbox"/> | Public |

Certificate: [Export root cert](#)

Set-side RTP security

Inbound

SRTP only Accept only RTP (plaintext) inbound to this server

SRTP or RTP

RTP only

Outbound

SRTP only Send only RTP (plaintext) outbound from this server

AVP+crypto

RTP only

Preferred cipher:

ICP-side RTP security

Inbound

SRTP only Accept only RTP (plaintext) inbound to this server

SRTP or RTP

RTP only

Outbound

SRTP only Send only RTP (plaintext) outbound from this server

AVP+crypto

RTP only

Preferred cipher:

Tone injection Enabled

Device → device local streaming

Device → trunk local streaming

Codec support:

RTP framesize:

PRACK support

Send options keepalives:

Options interval:

Challenge methods:

KPML username:

KPML password:

Confirm KPML password:

Registration Mode:

Set-side registration expiry time:

ICP-side registration expiry time:

Allowed URI names:

[Add another](#)

Blank any field you no longer want.

SIP adaptation support

SIP adaptation receive pipeline:

SIP adaptation send pipeline:

Permit weak SIP passwords

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 |