

Security Guidelines

Mediant™ Series of Session Border Controllers (SBC) & Media Gateways

Version 7.0

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Notice

This document describes the recommended security guidelines for AudioCodes Mediant Series devices.

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Each abbreviation, unless widely used, is spelled out in full when first used.

Throughout this manual, unless otherwise specified, the term *device* refers to AudioCodes products.

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Documentation Feedback

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1 Introduction

This document provides recommended security guidelines for safeguarding your network and your AudioCodes device against malicious attacks.

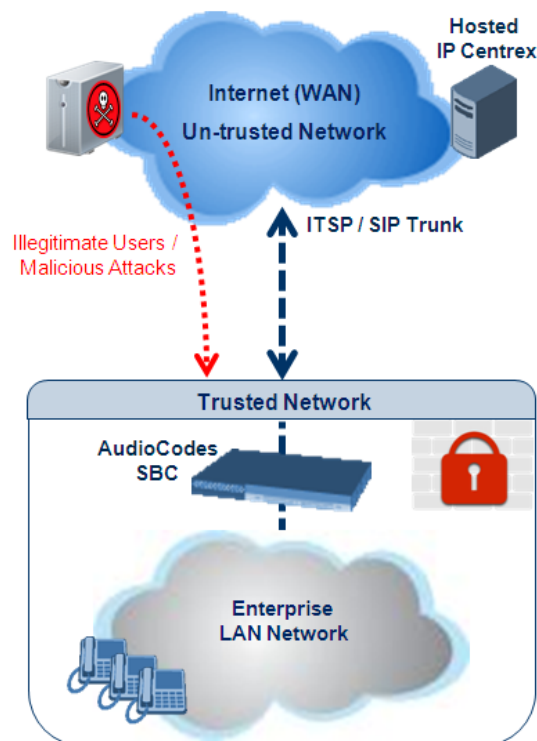
**Notes:**

- This document provides only recommended security guidelines; your network architecture may require additional and/or different security measures.
- This document includes partial configuration; for detailed configuration, refer to the device's *User's Manual*.

1.1 Security Threats

AudioCodes devices are commonly located at the demarcation point between safe (*trusted*) and unsafe (*un-trusted*) networks. A typical example of an un-trusted network would be a SIP trunk connected to an Internet Telephony Service Provider (ITSP) network; the trusted network would be the internal LAN. The figure below illustrates this basic concept of trusted and un-trusted networks.

Figure 1: Trusted and Un-trusted Networks



Attacks on your network from the un-trusted network may include the following:

- **Denial of Service (DoS) attacks:** Malicious attacks designed to cripple your VoIP network by overloading it with calls or service requests.
- **Overload events:** In addition to purposeful DoS attacks, non-malicious periods of intense activity can also cause an increase in call signaling rates that exceed what your infrastructure can support, resulting in network conditions that are similar in effect to DoS attacks. Successful attacks resulting in contact center downtime can result in lost revenue and diminished customer satisfaction.

- **Network abuse and fraud:** Malicious intrusion or service theft may take the form of an unauthorized user gaining access to your VoIP network by mimicking an authorized user or seizing control of a SIP proxy and initiating outbound calls to the PSTN for free. Another possibility is using a compromised endpoint to redirect or forward calls for eavesdropping.
- **Viruses and malware:** Computer viruses, worms, Trojan horses, and other malware can infect user agent phones and SIP-based ACD infrastructure - just as they can computers and servers - and degrade performance or completely disrupt service. As devices become more sophisticated with distinct operating systems, malware also serves as a way to subjugate devices and launch DoS attacks that piggyback encrypted links.
- **Identity theft:** Phishing and "man-in-the-middle" can be used to acquire caller identification information to gain unauthorized access to services and information. Theft by phone (or service theft), whereby access to your corporate phone system is attempted by users posing as legitimate ones can sky-rocket your corporation's phone bill.
- **Eavesdropping:** The ability to listen to or record calls is easier on VoIP networks than on PSTN. This is a concern not only because of personal privacy violations, but also because sensitive information can be compromised and exploited.
- **Spam over Internet Telephony (SPIT):** The delivery of unsolicited calls or voicemails can inundate networks, annoy subscribers, and diminish the usefulness of VoIP networks.

These threats can exist, for example, at the following main IP network border points:

- **Interconnect:** SIP trunks to ITSPs, using SIP signaling for inbound and outbound calls.
- **Trusted access:** Private, managed IP networks that connect service providers' residential, enterprise, or mobile subscribers (as part of an emerging federation of trusted networks).
- **Un-trusted access:** Unmanaged Internet for connections to work-at-home agents or inbound callers.

1.2 AudioCodes Security Solution

The AudioCodes device provides a comprehensive package of security features that handles the following two main security areas:

- **Securing the Service:** Secures the call services it provides by implementing separation and defense of different network entities (e.g., SIP Trunk, softswitch, and users). This is accomplished by the following:
 - Physical separation of networks
 - SRDs for each SIP entity (user agent)
 - IP Groups for each SIP entity (user agent)
- **Securing the Device:** This concerns two areas:
 - Management: ensuring that only authorized users can access the device's management interface
 - Defense against attacks on the device regarding SIP signaling and media (RTP)

Due to the vast number and types of potential attacks (some described in the previous section), security of your trusted VoIP network should be your paramount concern. The device provides a rich set of features to support perimeter defense for protecting your trusted network from the un-trusted ones. However, the device's security features and capabilities are only effective if implemented correctly. Improper use of the device for perimeter defense may render the overall security solution ineffective, thereby exposing your network to multiple threats.

The benefits of an IP-based telephony network are quite clear, but so are the threats and security implications that need to be addressed. The IP borders of the IP telephony network are the attack points and it is the AudioCodes security solutions that are designed to help safeguard your trusted network from such threats.

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2 Separate Network Traffic

This chapter provides recommendations for separating network traffic.

2.1 Identify Trusted and Un-trusted Networks

It is crucial that you identify the trusted network (i.e., your local LAN) and the un-trusted network (i.e., public Internet – WAN) in the environment in which the device is deployed. There may be multiple un-trusted networks in a single deployment environment. For example, far-end WAN users and a SIP trunk with an ITSP may represent two un-trusted networks.

Once identified, you need to handle the un-trusted networks with extreme caution in order to safeguard your trusted network from malicious attacks from them. One of the main precautions is to separate your trusted network from the un-trusted network, using different logical configuration entities such as SRDs etc. The precautions and security guidelines are described in detail in subsequent sections.

2.2 Employ VLANs to Separate LAN Traffic



Note: This section is applicable only to MP-11x, Mediant MSBR Series and Mediant 3000.

It is recommended to implement virtual LANs (VLAN) to separate OAMP (management), media (RTP), and SIP signaling traffic on the LAN. VLANs may increase security in your network. VLANs are configured for the device's network interfaces in the Interface table (**Configuration** tab > **VoIP** > **Network** > **IP Interfaces Table**), as shown below:

- MP-1xx; Mediant 2000; Mediant 3000:

Figure 2-1: VLANs in Interface Table

| Interface Table | | | | | | | | | |
|-----------------|------------------|----------------|------------|---------------|-----------------|---------|----------------|-------------|---------------|
| Add + | | | | | | | | | |
| Inde | Application Type | Interface Mode | IP Address | Prefix Length | Default Gateway | VLAN ID | Interface Name | Primary DNS | Secondary DNS |
| 0 | OAMP | IPv4 Manual | 10.8.6.55 | 16 | 10.8.0.1 | 100 | Mng | 10.1.1.11 | 10.1.1.10 |
| 1 | Media | IPv4 Manual | 10.8.6.56 | 16 | 10.8.0.1 | 200 | Voice-Media | 10.1.1.11 | 10.1.1.10 |
| 2 | Control | IPv4 Manual | 10.8.6.51 | 16 | 10.8.0.1 | 300 | Voice-Sig | 10.1.1.11 | 10.1.1.10 |

- Mediant MSBR Series:

Figure 2-2: Configuring VLAN IDs in Ethernet Device Table

| Ethernet Device Table | | |
|-----------------------|---------|----------|
| Index | VLAN ID | Name |
| 0 | 100 | vlan 100 |
| 1 | 200 | vlan 200 |
| 2 | 300 | vlan 300 |

Figure 2-3: Assigning VLANs (Underlying Devices) to Interfaces in Interface Table

| Interface Table | | | | | | | | | |
|-----------------|------------------|----------------|------------|---------------|-----------------|----------------|-------------|---------------|-------------------|
| Index | Application Type | Interface Mode | IP Address | Prefix Length | Default Gateway | Interface Name | Primary DNS | Secondary DNS | Underlying Device |
| 0 | OAMP | IPv4 Manual | 10.8.6.55 | 16 | 10.8.0.1 | Mng | 10.1.1.11 | 10.1.1.10 | vlan 100 |
| 1 | Media | IPv4 Manual | 10.8.6.56 | 16 | 10.8.0.1 | Voice-Media | 10.1.1.11 | 10.1.1.10 | vlan 200 |
| 2 | Control | IPv4 Manual | 10.8.6.51 | 16 | 10.8.0.1 | Voice-Sig | 10.1.1.11 | 10.1.1.10 | vlan 300 |

2.3 Implement Physical Network Separation using Ethernet Port Groups



Note: This section is applicable only to Mediant 500 Gateway & SBC, Mediant 800 Gateway & SBC, Mediant 1000B Gateway & SBC, Mediant 2600 E-SBC, Mediant 4000 SBC, Mediant 9000 SBC, and Mediant Software SBC.

For the devices mentioned in the note above, you can physically separate the network traffic by Ethernet ports, using Ethernet Groups. Each Ethernet Group can include up to two physical Ethernet ports. The Ethernet Device defines the VLAN per Ethernet Group. The Ethernet Device is then assigned to the network interface as an Underlying Device. The following procedure provides an example of assigning different ports per traffic type.

1. Assign ports to different Ethernet Groups in the Ethernet Group Settings table (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Groups Table**):

Figure 2-4: Assigning Ports to Ethernet Groups in Ethernet Group Settings Table

| Index | Group | Mode | Member 1 | Member 2 |
|-------|---------|--------|----------|----------|
| 0 | GROUP_1 | Single | GE_4_1 | None |
| 1 | GROUP_2 | Single | GE_4_2 | None |
| 2 | GROUP_3 | Single | GE_4_3 | None |

2. Configure VLAN IDs per Ethernet Group in the Ethernet Device table (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**):

Figure 2-5: Assigning VLANs to Ethernet Groups in Ethernet Device Table

| Index | VLAN ID | Underlying Interface | Name | Tagging |
|-------|---------|----------------------|--------|----------|
| 0 | 1 | GROUP_1 | VLAN 1 | Untagged |
| 1 | 2 | GROUP_2 | VLAN 2 | Untagged |
| 2 | 3 | GROUP_3 | VLAN 3 | Untagged |

- Assign the Ethernet Devices (*Underlying Device*) to the different traffic network interfaces in the Interface table (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**):

Figure 2-6: Assigning VLANs (Underlying Devices) to Interfaces in Interface Table

| Index | Application Type | Interface Mode | IP Address | Prefix Length | Default Gateway | Interface Name | Primary DNS | Secondary DNS | Underlying Device |
|-------|------------------|----------------|------------|---------------|-----------------|----------------|-------------|---------------|-------------------|
| 0 | OAMP | IPv4 Manual | 10.8.6.55 | 16 | 10.8.0.1 | Mng | 10.1.1.11 | 10.1.1.10 | vlan 1 |
| 1 | Media | IPv4 Manual | 10.8.6.56 | 16 | 10.8.0.1 | Voice-Media | 10.1.1.11 | 10.1.1.10 | vlan 2 |
| 2 | Control | IPv4 Manual | 10.8.6.51 | 16 | 10.8.0.1 | Voice-Sig | 10.1.1.11 | 10.1.1.10 | vlan 3 |

2.4 Implement Physical Network Separation



Note: This section is applicable only to Mediant 3000 with the TP-8410 blade.

It is highly recommended to separate traffic (i.e., OAMP, call control / SIP messages, and RTP media) so that each has its own dedicated physical Ethernet port. This Physical Network Separation feature eliminates the need for a VLAN-aware switch using VLAN tags to separate the different traffic of each network.

The physical interface provided for each network application type is as follows:

- **Media:** provided by the Gigabit Ethernet port on the RTM-8410
- **Control (SIP):** provide by 10/100BaseT Ethernet ports labeled **1A** (lower PEM module) and **1B** (upper PEM module) – two PEMs are used for redundancy
- **OAMP:** provided by 10/100BaseT Ethernet ports labeled **2A** (lower PEM module) and **2B** (upper PEM module) - two PEMs are used for redundancy

To enable this feature, the 'Network Physical Separation' parameter in the Interface table must be set to **Enable**.

Figure 2-7: Enabling Network Physical Separation in Interface Table

| Parameters | |
|-----------------------------|---------|
| VLAN Mode | Disable |
| Native VLAN ID | 1 |
| Network Physical Separation | Enable |
| IP Interface Status Table | |

2.5 Data-Router Specific Security Guidelines



Note: This section is applicable only to the MSBR series.

2.5.1 Use VLANs and VRFs to Separate WAN Traffic

It is recommended to implement the device's Virtual Routing and Forwarding (VRF) feature to segregate VoIP (SIP and media) / OAMP traffic from data-routing traffic into routing domains on the WAN interface. This is achieved by using the following CLI commands:

- `ip vrf, ip vrf forwarding`
- `ip route vrf`

Below shows a basic example for creating two VFRs, one for VoIP / OAMP ("VOIP") and one for data ("DATA"):

```
(config-data)# ip vrf VOIP
(conf-if-VLAN 200)# ip vrf forwarding VOIP
(config-data)# ip route vrf VOIP 10.1.1.1 255.255.0.0 10.8.0.1
vlan 200

(config-data)# ip vrf DATA
(conf-if-VLAN 300)# ip vrf forwarding DATA
(config-data)# ip route vrf VOIP 10.1.1.1 255.255.0.0 10.8.0.1
vlan 300
```

2.5.2 Use Port VLANs (PVID) to Physically Separate LAN Traffic

You can also separate the LAN traffic you configured in the Interface table between physical LAN ports, using port VLANs (PVID).

3 Implement Layer 3/4 (Network) Firewall

This section discusses Layer 3/4 (Network) firewall recommendations. By default, there are no firewall rules and this exposes the device to security risks. Therefore, configuring firewall rules is highly recommended to protect the device from external attacks.

3.1 Block Unused Network Ports

It is highly recommended that you disable network ports that are not used in your deployment. For example, if you are not using Trivial File Transfer Protocol (TFTP) in your network, then you should disable this network port application.

3.2 Define VoIP Traffic Firewall Rules

For packets whose source IP addresses are known, it is recommended to define VoIP firewall rules that allow receipt of calls or packets from this network and block all calls from elsewhere. These rules can be defined per source IP address, port, protocol, and network IP interface. If an incoming packet is received from an invalid source (as defined in the firewall), the call or packet is discarded.



Note: For devices with LAN and WAN interface ports (i.e., MSBR series), it is more important that a firewall be configured for the WAN interface, which is exposed to the public network (i.e., Internet). For more information, see Section 3.3.

Below is a list of recommended guidelines when configuring the VoIP firewall:

- **Add firewall rules per network interface:** It is recommended to define firewall rules for packets from source IP addresses received on the OAMP interface and each SIP Control (SIP) interface (defined in the Multiple Interface table). A less recommended alternative is to define a single rule that applies to all interfaces (by setting the 'Use Specific Interface' parameter to 'Disable').
- **Define bandwidth limitation per rule:** For each IP network interface, it is advised to define a rate-limiting value (byte rate, burst bytes and maximum packet size). Bandwidth limitation prevents overloading (flooding) of your network and thereby, helps in preventing attacks such as DoS on your device (on each network).
- **Define rules as specific as possible:** Define the rules as detailed as possible so that they block only the intended traffic.
- **Add an ICMP firewall rule:** ICMP is typically used for pinging. However, malicious attackers can send over-sized (floods) ICMP packets to a specific network address. Therefore, it is recommended to define a rule for limiting these packets.
- **Add a rule to block all traffic:** You must define a firewall rule that blocks **all** incoming traffic (i.e., block all protocol traffic from all source IP addresses and ports for all interfaces). This rule must be the **last** rule listed in the table, so that rules above it that allow specific traffic are valid (otherwise, all traffic is blocked).



Warning: If the 'Prefix Length' field on the Firewall Settings page is set to "0", the rule will apply to **all** IP addresses, regardless of whether an IP address is specified in the 'Source IP' field. Thus, if you need to apply a rule to a specific IP address, ensure that you also set the 'Prefix Length' field to a value other than "0".

The Layer 3-4 VoIP traffic firewall rules are configured in the Firewall Settings table (**Configuration** tab > **VoIP** menu > **Security** > **Firewall Settings**). The table below shows a configuration example of firewall rules:

Configuration Example of Firewall Rules in the Firewall Settings Table

| Parameter | Index Rule | | | | |
|-------------------------|---------------|--------------|---------|-----------|---------|
| | 1 | 2 | 3 | 4 | 5 |
| Source IP | 12.194.231.76 | 12.194.230.7 | 0.0.0.0 | 192.0.0.0 | 0.0.0.0 |
| Prefix Length | 16 | 16 | 0 | 8 | 0 |
| Start Port and End Port | 0-65535 | 0-65535 | 0-65535 | 0-65535 | 0-65535 |
| Protocol | Any | Any | icmp | Any | Any |
| Use Specific Interface | Enable | Enable | Disable | Enable | Disable |
| Interface Name | WAN | WAN | None | Voice | None |
| Byte Rate | 0 | 0 | 40000 | 40000 | 0 |
| Burst Bytes | 0 | 0 | 50000 | 50000 | 0 |
| Action Upon Match | Allow | Allow | Allow | Allow | Block |

- **Rules 1 and 2:** Typical firewall rules that allow packets ONLY from specified IP addresses (e.g., proxy servers). Note that the prefix length is configured.
- **Rule 3:** A more "advanced" firewall rule - bandwidth rule for ICMP, which allows a maximum bandwidth of 40,000 bytes/sec with an additional allowance of 50,000 bytes. If, for example, the actual traffic rate is 45,000 bytes/sec, then this allowance would be consumed within 10 seconds, after which all traffic exceeding the allocated 40,000 bytes/sec is dropped. If the actual traffic rate then slowed to 30,000 bytes/sec, the allowance would be replenished within 5 seconds.
- **Rule 4:** Allows traffic from the LAN voice interface and limits bandwidth.
- **Rule 5:** Blocks all other traffic.

3.3 Define Data-Routing Firewall Rules for WAN



Note: This section is applicable only to the MSBR series and if you are using the WAN interface.

The device provides two main network interfaces. The LAN which is the internal, private network of your enterprise, and the WAN which is the public network (Internet). Since the WAN interface is your LAN's "door" to the public network, you must define data-routing firewall rules for the WAN interface to prevent unwanted access from this unsafe, public network. The data-routing firewall rules are relevant for application protocols that are used by VoIP networks such as DHCP and TFTP. For example, firewall rules can be applied to configuration file downloads from a TFTP server that is located in the un-trusted network, by SIP user agents (e.g., IP phones) located in the trusted network.

This section discusses the guidelines for securing the device's WAN interface using the data-routing firewall capabilities.

3.3.1 Avoid Creation of DMZ Host

It is recommended not to configure a demilitarized zone (DMZ) host. The DMZ host allows a single local computer to be exposed to all services on the Internet, without any restrictions. A DMZ host is not protected by the firewall and may be vulnerable to attack. Designating a DMZ host may also put other equipment in the network at risk. When designating a DMZ host, you must consider the security implications and protect it if necessary.

3.3.2 Define Explicit Access List Rules

If your enterprise wishes to use the device's data-routing functionality, for example, a DHCP server (for distributing IP addresses to the enterprise's IP phones connected through the LAN switch), it is recommended that you define Access List rules (ACL) for blocking (*deny*) and allowing (*permit*) specified ingress (inbound) traffic from the public IP network on the WAN interface. If not defined, the device accepts all traffic.



Note: If your device is deployed solely as an SBC (VoIP functionality) without using the data-router functionality, ACL rules are unnecessary. This is due to the device's automatic NAT / port forwarding feature, which ensures that only desired traffic is allowed into the network.

You must define ACL rules as explicitly as possible:

- Include as much information as possible for each rule. For example, if you want to allow RTP traffic from only a specific IP address (e.g., 1.1.1.3), in addition to the IP address, you should also specify the allowed destination port (e.g., 6000-8000).
- Define adjacent rules (or overlapping rules) to block unwanted incoming traffic that may "slip" through an already defined allowed rule. For example, if the rule mentioned above is defined, unwanted RTP packets from IP address 1.1.1.3 destined to port 6600-6700 will also be allowed. Therefore, to block this traffic, define another rule explicitly denying traffic to port 6600-6700.

- Define rules to suite your environment so that only the required, minimum traffic is allowed (while all others are blocked).



Note: Incoming traffic that does not match any ACL rule is discarded even if a rule has not been defined to deny it (and no matching with an allowed rule exists where it can "slip" through).

Below shows an example of configured ACL rules called "WAN Firewall":

- Allow HTTPS traffic from IP address 1.1.1.1 and destined to port 443:

```
(config-data)# access-list wan-firewall permit tcp host
1.1.1.1 any eq 443
```

- Allow SSH traffic from IP address 1.1.1.1 and destined to port 22:

```
(config-data)# access-list wan-firewall permit tcp host
1.1.1.1 any eq 22
```

- Block all UDP traffic from IP address 1.1.1.3 destined to port 6600-6700. This is an "adjacent" rule for the rule below to prevent undesired traffic from "slipping" through:

```
(config-data)# access-list wan-firewall deny udp host 1.1.1.1
any range 6600 6700
```

- Allow UDP (RTP) traffic from IP address 1.1.1.3 and destined to port 6000-8000:

```
(config-data)# access-list wan-firewall permit udp host
1.1.1.1 any range 6000 8000
```

To associate the ACL with an interface (e.g., Gigabit WAN Ethernet):

```
(config-data)# interface GigabitEthernet 0/0
(conf-if-GE 0/0)# ip access-group wan-firewall in
(conf-if-GE 0/0)# exit
```

4 Secure Management Access

This section provides guidelines to secure access to the device's management interface.

4.1 Change Default Management User Login Passwords

To secure access to the device's Web management interface, please adhere to the following recommended guidelines:

- The device is shipped with a default **Security Administrator** access-level user account – username "Admin" and password "Admin". This user has full read-write access privileges to the device. It is recommended to change the default password to a hard-to-hack string. The login username and password are configured on the Web User Accounts page (**Configuration** tab > **System** > **Web User Accounts**) using the 'Current Password', 'New Password', and 'Confirm New Password' fields, as shown below:

Figure 8: Changing Password of Default Security Administrator User

| Account Data for User: Admin | |
|---|---|
| User Name | Admin <input type="button" value="Change User Name"/> |
| Access Level | Security Administrator |
| ▼ Fill in the following 3 fields to change the password | |
| Current Password | <input type="button" value="↻"/> |
| New Password | <input type="button" value="↻"/> |
| Confirm New Password | <input type="button" value="Change Password"/> |

- The device is shipped with a default **Monitor** access-level user account - username "User" and password "User". This user only has read access privileges to the device. The read access privilege is also limited to certain Web pages. However, this user can view certain SIP settings such as proxy server addresses. Therefore, in order to prevent an attacker from obtaining sensitive SIP settings that could result in possible call theft etc., change the access level of this user to **No Access** (see figure below). In addition (or alternatively), change its default login password to a hard-to-hack string.

Figure 9: Changing Access Level to No Access of Default Monitor User

| Account Data for User: User | |
|---|---|
| User Name | User <input type="button" value="Change User Name"/> |
| Access Level | <div style="border: 1px solid black; padding: 2px;"> User No Access User Monitor Administrator Security Administrator </div> <input type="button" value="Change Access Level"/> |
| ▼ Fill in the following 3 fields to change the password | |

- If you have deployed multiple devices, use a unique password for each device.
- Change the login password periodically (for example, once a month).

4.2 Implement LDAP-based User Authentication and Authorization

It is highly recommended that you implement a third-party, LDAP server in your network for authenticating and authorizing the device's management users (Web and CLI). This can be done by using an LDAP-compliant server such as Microsoft Active Directory (AD). When a user attempts to log in to one of the management platforms, the device verifies the login username and password with AD. The device can also determine the user's management access level (privileges) based on the user's profile in the AD. This is configured in the pages located under the **LDAP** menu (**Configuration** tab > **VoIP** menu > **Services** > **LDAP**).

An alternative to using an LDAP server is to use a RADIUS server, as discussed in the next section.

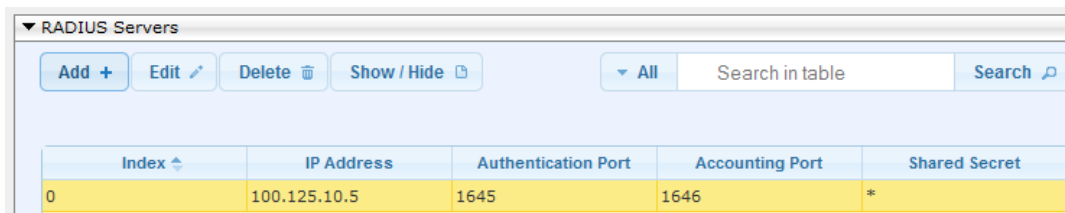
4.3 Implement RADIUS-based User Authentication

It is highly recommended that you implement a third-party, RADIUS server in your network for authenticating Web / Telnet users and thereby, preventing unauthorized access. RADIUS allows you to define different passwords for different interface users, with centralized management of the password database. When RADIUS is used, logging into the Web / Telnet interfaces is performed through the RADIUS server. The device verifies the authenticity of the user name and password with the RADIUS server.

An alternative is to use an LDAP server, as discussed in the previous section.

RADIUS authentication servers (IP address, port and 'secret' password for authenticating the device with the RADIUS server) are configured in the RADIUS Servers table (**Configuration** tab > **System** > **Management** > **RADIUS Servers**).

Figure 10: Configuring RADIUS Servers for Web User Authentication

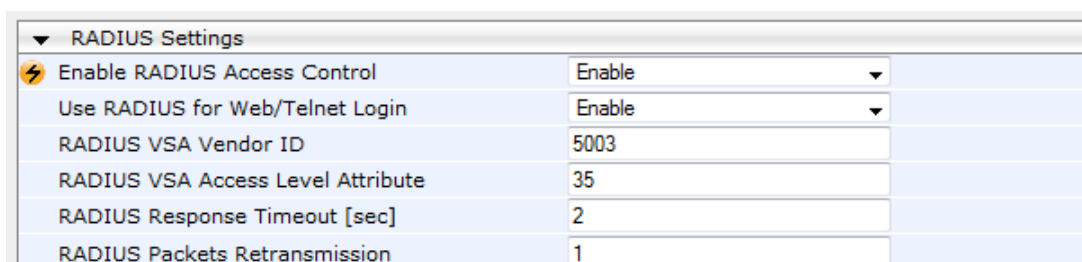


| Index | IP Address | Authentication Port | Accounting Port | Shared Secret |
|-------|--------------|---------------------|-----------------|---------------|
| 0 | 100.125.10.5 | 1645 | 1646 | * |

RADIUS functionality is enabled on the RADIUS Settings page (**Configuration** tab > **System** > **Management** > **Authentication Settings**), using the following parameters:

- 'Enable RADIUS Access Control': select **Enable**.
- 'Use RADIUS for Web/Telnet Login': select **Enable**.

Figure 11: Enabling RADIUS for Web User Authentication



| RADIUS Settings | |
|-----------------------------------|--------|
| Enable RADIUS Access Control | Enable |
| Use RADIUS for Web/Telnet Login | Enable |
| RADIUS VSA Vendor ID | 5003 |
| RADIUS VSA Access Level Attribute | 35 |
| RADIUS Response Timeout [sec] | 2 |
| RADIUS Packets Retransmission | 1 |

4.4 Implement Two-Way Authentication with X.509 Certificates

It is recommended to use two-way authentication (in addition to HTTPS) between the device's Web server and the management station (i.e., computer) accessing it. Authentication is performed and connection to the Web interface is subsequently allowed only if the following conditions are met:

- The management station possesses a client certificate from a Certification Authority (CA).
- The CA certificate is listed in the device's Trusted Root CA Store.

Otherwise, the connection is rejected. Therefore, this prevents unauthorized access to the Web management tool.



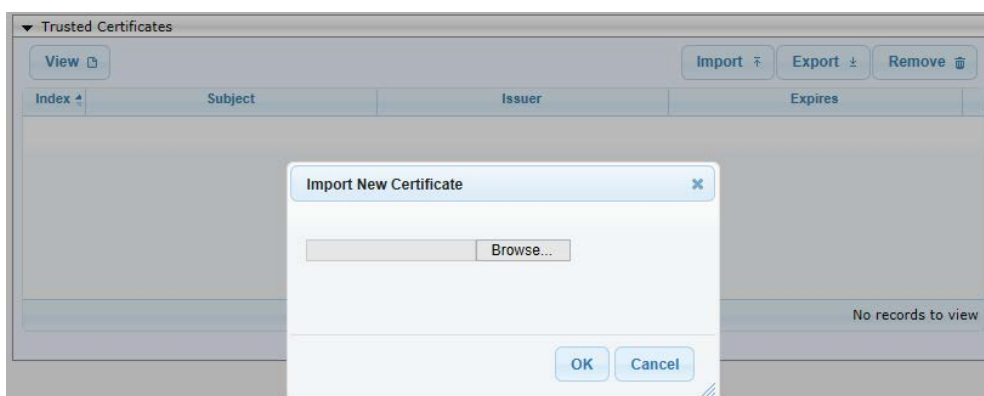
Notes:

- Implementation of two-way authentication requires a third-party security equipment vendor, CA server, and security administrator personnel. These should create certificates and deploy them to all the computers in the organization.
- The device is supplied with a working TLS configuration consisting of a unique self-signed server certificate. Replace this certificate with one provided by your security administrator. For more information, refer to the *User's Manual*.

➤ To configure client-server, two-way authentication using X.509 certificates:

1. Install a client certificate on the management station (your network administrator should provide you with a certificate).
2. Install your organization's CA certificate on the management station.
3. Open the TLS Contexts page (**Configuration** tab > **System** menu > **TLS Contexts**).
4. In the TLS Contexts table, select the required TLS Context index row, and then click the **TLS Context Trusted Root Certificates** button, located at the bottom of the TLS Contexts page; the Trusted Certificates page appears.
5. Click the **Import** button, browse to and select the Root CA certificate file (in base64-encoded PEM format), and then import the file.

Figure 12: Importing CA Certificate to CA Store



6. Since X.509 certificates have an expiration date and time, the device must be configured to use Network Time Protocol (NTP) to obtain the current date and time. Without the correct date and time, client certificates cannot operate.
7. Ensure that client certificates for HTTPS connections are required, by setting the 'Requires Client Certificates for HTTPS connection' to **Enable** on the Web Security Settings page (**Configuration** > **System** > **Management** > **Web Security Settings**).

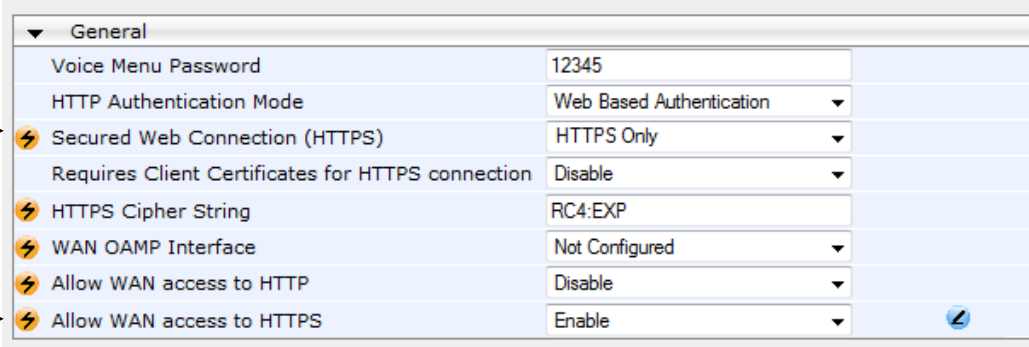
4.5 Secure HTTP Access using HTTPS

It is recommended to allow access to the Web interface using HTTPS only. In addition, it is recommended to block port 80.

This is configured on the Web Security Settings page (**Configuration** tab > **System** > **Management** > **Web Security Settings**), by setting the following parameters:

- 'Secured Web Connection (HTTPS)': **HTTPS Only**.
- **(MSBR Series)** If you are accessing the Web management through the WAN port, set the 'Allow WAN access to HTTPS' parameter to **Enable**.

Figure 13: Securing Access to Web Interface using HTTPS



| General | |
|---|--------------------------|
| Voice Menu Password | 12345 |
| HTTP Authentication Mode | Web Based Authentication |
| ⚡ Secured Web Connection (HTTPS) | HTTPS Only |
| Requires Client Certificates for HTTPS connection | Disable |
| ⚡ HTTPS Cipher String | RC4:EXP |
| ⚡ WAN OAMP Interface | Not Configured |
| ⚡ Allow WAN access to HTTP | Disable |
| ⚡ Allow WAN access to HTTPS | Enable |



Note: MSBR Series: For security reasons, it is recommended to disable access from the WAN port. However, if HTTP management access from the WAN is necessary, then it is recommended that you disable the WAN port for all other management access types (i.e., Telnet, SSH / CLI, and SNMP). In addition, if you enable WAN management access, ensure that you define Layer 5 (see Section 4.8) and Layer 3-4 (see Section 2) firewall rules to prevent access by undesirable elements from the public network.

4.6 Secure Telnet Sessions

If you require the use of Telnet and your management PC software provides a secure Telnet application, use a secured Telnet connection (i.e., Transport Layer Security / TLS). TLS protects Telnet traffic from network sniffing.

This is configured on the Telnet/SSH Settings page (**Configuration** tab > **System** > **Management** > **Telnet/SSH Settings**), by setting the following parameters:

- 'Embedded Telnet Server': **Enable Secured**.
- 'Telnet Server TCP Port': Change the default TCP port (if required). The configuration is applicable for access from the LAN.
- **(MSBR Series)** If management is through the WAN port, set the 'Allow WAN access to Telnet' parameter to **Enable**.

Figure 14: Securing Telnet with TLS

| Telnet Settings | |
|----------------------------|----------------|
| Embedded Telnet Server | Enable Secured |
| Telnet Server TCP Port | 25 |
| Telnet Server Idle Timeout | 5 |
| Allow WAN access to Telnet | Enable |



Note: MSBR Series: It is recommended to disable access from the WAN port. However, if Telnet management access from the WAN is necessary, then it is recommended that you disable the WAN port for all other management access types (i.e., HTTPS, SSH / CLI, and SNMP). In addition, if you enable WAN management access, ensure that you define Layer 5 (see Section 4.8) and Layer 3-4 (see Section 2) firewall rules to prevent access by undesirable elements from the public network.

4.7 Secure SSH Sessions

Secure SHell (SSH) is the de-facto standard for secure CLI. SSH 2.0 is a protocol built above TCP, providing methods for key exchange, authentication, encryption, and authorization. By default, SSH uses the same username and password as the Telnet and Web server. SSH supports 1024- and 2048-bit RSA public keys, providing carrier-grade security.

This is configured on the Telnet/SSH Settings page (**Configuration** tab > **System** > **Management** > **Telnet/SSH Settings**), by setting the following parameters:

- 'Enable SSH Server': select **Enable**.
- 'Server Port': If desired, you may change the default TCP port used for SSH, although this is not recommended. Note that this is applicable for access from the LAN.
- **(MSBR Series)** If management is through the WAN port, set the 'Allow WAN access to SSH' parameter to **Enable**.

Figure 15: Securing SSH (CLI) Sessions

| SSH Settings | |
|---------------------------|---------|
| Enable SSH Server | Enable |
| Server Port | 22 |
| Admin Key | |
| Require Public Key | Disable |
| Max Payload Size | 32768 |
| Max Binary Packet Size | 35000 |
| Enable Last Login Message | Enable |
| Max Login Attempts | 3 |
| Allow WAN access to SSH | Enable |
| Maximum SSH Sessions | 5 |



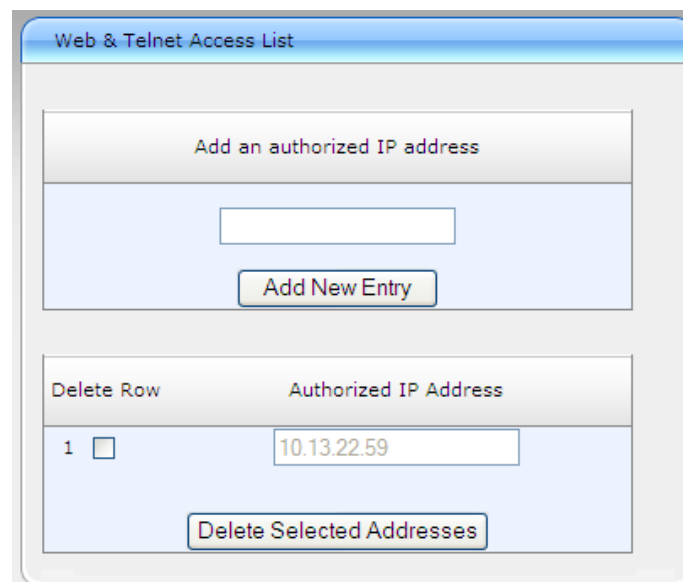
Note: MSBR Series: It is recommended to disable access through the WAN port. However, if CLI management access from the WAN is necessary, then it is recommended that you disable the WAN port for all other management access types (i.e., HTTPS, Telnet, and SNMP). In addition, if you enable WAN management access, ensure that you define Layer 5 (see Section 4.8) and Layer 3-4 (see Section 2) firewall rules to prevent access by undesirable elements from the public network.

4.8 Define Web, Telnet, and SSH Authorized Access List

Allow only user-defined LAN IP addresses to access the Web, Telnet, and SSH based management interfaces. The device denies access from undefined IP addresses.

This is configured on the Web & Telnet Access List page (**Configuration** tab > **System** > **Management** > **Web & Telnet Access List**).

Figure 16: Authorized IP Addresses for Accessing Web / Telnet / SSH Interfaces



| Delete Row | Authorized IP Address |
|----------------------------|-----------------------|
| 1 <input type="checkbox"/> | 10.13.22.59 |



Notes:

- The first authorized IP address in the list must be your PC's (terminal) IP address; otherwise, access from your PC will be denied.
- **MSBR Series:** This authorized access list concerns access only from the LAN (not WAN).
- The Web / Telnet / SSH authorized access list concerns OSI Layer 5 (Session). However, you can also add firewall rules for Layer 3 (Network) and Layer 4 (Transport) with bandwidth limitation to limit access to management interfaces (see Section 3.1).

4.9 Secure SNMP Interface Access

This section discusses recommended security guidelines relating to Simple Network Management Protocol (SNMP).

4.9.1 Prefer SNMPv3 over SNMPv2

It is highly recommended to use SNMP Version 3 (SNMPv3) over SNMPv1 and SNMPv2c, if possible. SNMPv3 provides secure access to the device using a combination of authentication (MD5 or SHA-1) and encryption (DES, 3DES, AES-128, AES-192, or AES-256) of packets over the network. It is also recommended that you periodically change the SNMPv3 authentication and privacy keys. The SNMPv3 users are configured in the SNMP v3 Users table (**Configuration** tab > **System** > **Management** > **SNMP** > **SNMP V3 Users**), as shown in the figure below:

Figure 17: SNMPv3 Users Table

| Index | User Name | Authentication Protocol | Privacy Protocol | Authentication Key | Privacy Key | Group |
|-------|-----------|-------------------------|------------------|--------------------|-------------|------------|
| 0 | JoeD | MD5 | 3DES | * | * | Read-Write |

4.9.2 Secure SMNPv2 Access

If you are using SNMPv2, change the community strings from their default values ("public") as they can easily be guessed by hackers. In addition, by default, the SNMPv2 agent accepts SNMP Get and Set requests from any IP address as long as the correct community string is used in the request. Therefore, to enhance security with SNMPv2, implement Trusted Managers. A Trusted Manager is an IP address (management station) from which the SNMP agent accepts and processes Get and Set requests. It is also recommended that you periodically change these SNMP community string values.

- SNMPv2 community strings are configured on the SNMP Community String page (**Configuration** tab > **System** > **Management** > **SNMP** > **SNMP Community Settings**):
 - 'Community String': enter a read-write community string (up to 19 characters; default string is "private").
 - **(MSBR Series)** If the SNMP management is through the WAN port, set the 'Allow WAN access to SNMP' parameter to **Enable**.

Figure 18: SNMPv2 Community Strings

| Community String | Access Level |
|------------------|--------------|
| | Read Only |
| | Read Only |
| | Read Only |
| | Read Only |
| | Read Only |
| snmpv2user_john | Read / Write |
| | Read / Write |
| | Read / Write |
| | Read / Write |
| | Read / Write |

| | |
|--|----------|
| <input type="checkbox"/> Disable SNMP | No |
| Trap Community String | trapuser |
| Trap Manager Host Name | |
| <input checked="" type="checkbox"/> Allow WAN access to SNMP | Enable |



Note: MSBR Series: It is recommended to disable access from the WAN port. However, if SNMP management access from the WAN is necessary, then it is recommended that you disable the WAN port for all other management access types (i.e., HTTPS, Telnet, and CLI / SSH). In addition, if you enable WAN management access, ensure that you define Layer 3-4 (see Section 2) firewall rules to prevent access by undesirable elements from the public network.

- SNMPv2 management stations are configured on the SNMP Trusted Managers page (Configuration tab > System > Management > SNMP > SNMP Trusted Managers):

Figure 19: SNMPv2 Trusted Managers

| Delete | Trusted Managers IP Address | |
|--------------------------|-----------------------------|-------------|
| <input type="checkbox"/> | SNMP Trusted Manager 1 | 10.13.22.25 |
| <input type="checkbox"/> | SNMP Trusted Manager 2 | 0.0.0.0 |
| <input type="checkbox"/> | SNMP Trusted Manager 3 | 0.0.0.0 |
| <input type="checkbox"/> | SNMP Trusted Manager 4 | 0.0.0.0 |
| <input type="checkbox"/> | SNMP Trusted Manager 5 | 0.0.0.0 |

4.9.3 Secure LDAP Communication

If you are using LDAP-based login management (username-password) and/or LDAP-based SIP routing in your deployment, it is recommended to employ TLS for secure device communication with the LDAP server. This ensures that the device encrypts the username and password sent to the LDAP server.

TLS for LDAP communication is configured in the LDAP Configuration table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **LDAP Configuration Table**):

- 'Use TLS': set to **Yes**
- 'TLS Context': select the TLS Context (configured in the TLS Contexts table)

This page is intentionally left blank.

5 Secure SIP using TLS (SIPS)

It is crucial that you implement the TLS-over-TCP protocol to best secure the device's SIP signaling connections. TLS provides encryption and authentication of SIP signaling for your VoIP traffic, preventing tampering of calls. Use it whenever possible for far-end users and ITSPs.

The device's TLS feature supports the following:

- **Transports:** SSL 2.0, SSL 3.0, TLS 1.0, TLS 1.1, TLS 1.2
- **Ciphers:** 3DES, RC4 compatible, Advanced Encryption Standard (AES)
- **Authentication:** X.509 certificates
- **Revocation checking:** OCSP (CRLs are currently not supported)
- Receipt of wildcards (*) in X.509 Certificates when establishing TLS connections. These wildcards can be part of the CN attribute of the Subject field or the DNSName attribute of the SubjectAltName field.

Recommended security guidelines for ensuring TLS for SIP signaling are described in the subsequent subsections.

5.1 Use Strong Authentication Passwords

Always use strong authentication passwords, which are more difficult to detect than weak ones. A strong password typically includes at least six characters with a combination of upper and lower case letters, numbers and symbols.

5.2 Use TLS Version 1.0 Only

It is highly recommended to use TLS version 1.0 only. This ensures that only TLS 1.0-based connections are accepted while offers of previous versions (i.e., SSL 2.0 and SSL 3.0) are rejected.



Note: TLS 1.0 should be implemented only if it is compatible with the rest of your network. If other network entities use SSL 2.0 / SSL 3.0 handshakes, then this recommendation should be ignored and SSL 2.0 / SSL 3.0 should be allowed.

This is configured in the TLS Contexts table (**Configuration** tab > **System** menu > **TLS Contexts**), by setting the 'TLS Version' parameter to "1", as shown below:

Figure 20: Configuring TLS Ver. 1.0 in TLS Contexts Table

| Add Record | |
|---|----------|
| Index | 1 |
| Name | |
| TLS Version | 1 |
| Cipher Server | AES:RC4 |
| Cipher Client | ALL:!ADH |
| OCSP Server | Disable |
| Primary OCSP Server | 0.0.0.0 |
| Secondary OCSP Server | 0.0.0.0 |
| OCSP Port | 2560 |
| OCSP Default Response | Reject |
| <input type="button" value="Submit"/> <input type="button" value="Cancel"/> | |

5.3 Use TLS for SIP Interfaces and Block TCP/UDP Ports

Each port can be vulnerable to attacks. Therefore, it is highly recommended that your SIP interfaces use **only** TLS. When configuring your SIP Interfaces, define the TLS port number, but set the UDP and TCP ports to zero ("0"). This configuration blocks (disables) the UDP and TCP ports. In other words, to disable UDP and TCP ports, you must define SIP Interfaces. In addition, to increase security, define only SIP Interfaces that are absolutely necessary.

SIP Interfaces are configured on the SIP Interface Table page (**Configuration** tab > **VoIP** > **VoIP Network** > **SIP Interface Table**). The figure below shows an example of a SIP Interface configured for the Voice network interface (LAN) with UDP and TCP ports set to 0:

Figure 21: SIP Interface using only TLS Port

| Add Row | |
|--|------------------|
| Index | 0 |
| SRD | DefaultSRD |
| Name | LAN |
| Network Interface | O+M+C |
| Application Type | SBC |
| UDP Port | 0 |
| TCP Port | 0 |
| TLS Port | 5061 |
| Encapsulating Protocol | No encapsulation |
| <input type="button" value="Add"/> <input type="button" value="Cancel"/> | |

5.4 Use TLS for Routing Rules

It is recommended that your routing rules use TLS only as the transport type. This is configured in the IP-to-IP Routing table (**Configuration** tab > **VoIP** > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**) by setting the 'Destination Transport Type' parameter to **TLS**:

Figure 22: IP-to-IP Routing Rule using SIP over TLS

The screenshot shows a 'Add Row' dialog box with the following configuration:

- Index: 0
- Routing Policy: Default_SBCRoutingP
- Tabbed interface:
 - Rule** (selected)
 - Action**
- Destination Type: IP Group
- Destination IP Group: IPGroup_0
- Destination SIP Interface: None
- Destination Address: (empty)
- Destination Port: 0
- Destination Transport Type: TLS (indicated by an arrow)
- Call Setup Rules: 1

Buttons: Add, Cancel

5.5 Implement X.509 Certificates for SIPS (TLS) Sessions

It is highly recommended to implement the X.509 certificate authentication mechanism for enhancing and strengthening TLS. X.509 is an ITU-T standard for Public Key Infrastructure (PKI).

The device supports the configuration of multiple TLS certificates, referred to as TLS Contexts. TLS Contexts are assigned to Proxy Sets and/or SIP Interfaces, thereby enabling specific calls to use specific TLS certificates.

The device is shipped with a working TLS configuration (TLS Context ID 0), consisting of a unique Self-Signed Server Certificate. Self-Signed Certificate is the simplest form of an X.509 Certificate that is issued by the device itself without the use of any certificate signer (CA). The Self-Signed Certificate consists of the Public Key of the device that is signed by the Private Key of the device itself. However, use of this certificate is **strongly discouraged**. The Self-Signed Certificate is typically used in testing environments or for a low-scale deployment where solution security may be sacrificed in favor of simplified configuration procedures. The Self-Signed Certificate does not utilize CA trust relationships and its authenticity cannot be reliably verified. Instead, you should establish a PKI for your organization (provided by your security administrator) and use certificates signed by genuine CAs.

In a typical PKI scheme, Certificates are issued by a CA and provide an attestation by the CA that the identity information and the public key belong together. Each party has a list of Trusted Root Certificates – certificates of the CAs (or their roots) that are well-known and trusted by the party. When the certificate from the other party is received, its signing entity (CA) is compared with the Trusted Root Certificates list and if a match is found, the certificate is accepted.

The device uses the following files to implement X.509 PKI:

- **Private Key File:** This file contains a private key that is used to perform decryption. It is the most sensitive part of security data and should never be disclosed to other entities.
- **Certificate File:** This file contains a digital signature that binds together the Public Key with identity information. The Certificate may be issued by a CA or self-signed (issued by the device itself, which is not recommended – see above).
- **Trusted Root Certificate File:** This file is the certificate of the Trusted Root CA used to authorize certificates received from remote parties, based on the identity of the CA that issued it. If the root certificate of this CA matches one of the Trusted Root Certificates, the remote party is authorized.

5.6 Use an NTP Server

It is recommended to implement a third-party, NTP server so that the device receives the correct current date and time. This is necessary for validating certificates of remote parties.

The NTP server is configured on the Time And Date page (**Configuration** tab > **System** > **Time And Date**):

Figure 5-23: NTP Server Address Configuration on Time And Date Page

| | | |
|---|-----------|------------|
| ▼ NTP Server | | |
| Primary NTP Server Address (IP or FQDN) | 0.0.0.0 | |
| Secondary NTP Server Address (IP or FQDN) | | |
| NTP Update Interval | Hours: 24 | Minutes: 0 |

It is also recommended to enable the device to authenticate and validate messages received from the NTP server. Authentication is done using an authentication key with the MD5 cryptographic hash algorithm. NTP messages that are received without authentication are ignored. This is configured on the Application Settings page (**Configuration** tab > **System** > **Application Settings**):

Figure 5-24: NTP Authentication Configuration on Application Settings Page

| | | |
|-----------------------------------|---|--|
| ▼ NTP Settings | | |
| NTP Authentication Key Identifier | 0 | |
| NTP Authentication Secret Key | | |

6 Implement LDAP-based Conditional Call Routing

It is recommended that you implement a third-party, LDAP server in your network for determining whether a call from a specific source is permitted to be routed to its destination. This setup uses Call Setup rules, configured in the Call Setup Rules table, to define a condition-based script that queries an LDAP server for the caller's number (for example) in a specific LDAP attribute. If the number exists, the device routes the call to the destination; otherwise, the call is dropped. The device executes a Call Setup rule upon the receipt of an incoming call (dialog) at call setup, if a matching routing rule exists in the IP-to-IP Routing table, before the <device> routes the call to its destination.

For configuring the LDAP server for LDAP-based routing, use the configuration pages located under the **LDAP** menu (**Configuration** tab > **VoIP** menu > **Services** > **LDAP**). For configuring Call Setup rules, use the Call Setup Rules table (**Configuration** tab > **VoIP** menu > **Services** > **LDAP** > **Call Setup Rules**).

The below Call Setup rule example routes the incoming call according to whether or not the source number of the incoming call exists in the AD server. The device queries the AD server for the attribute record, "telephoneNumber" whose value is the same as the received source number (e.g., "telephoneNumber=4064"). If such an attribute is found, the device routes the call to the destination as specified in the IP-to-IP Routing table; if the query fails (i.e., the source number does not exist in the AD server), the device rejects the call.

Figure 6-1: Call Setup Rule for Conditional LDAP-based Routing

| Index | Rules Set ID | Attributes To Query | Attributes To Get | Row Role | Condition | Action Subject | Action Type | Action Value |
|-------|--------------|---------------------|-------------------|---------------|----------------|----------------|-------------|--------------|
| 0 | 0 | 'telephoneNum | telephoneNum | Use Current C | ldap.found !ex | | Exit | false |

Page 1 of 1 10 View 1 - 1 of 1

Selected Row #0

| | | | |
|----------------------|--|-----------------|--------------------|
| Index: | 0 | Condition: | ldap.found !exists |
| Rules Set ID: | 0 | Action Subject: | |
| Attributes To Query: | 'telephoneNumber=' + param.call.src.user | Action Type: | Exit |
| Attributes To Get: | telephoneNumber | Action Value: | false |
| Row Role: | Use Current Condition | | |

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7 Define Maximum Call Duration

It is recommended to define maximum call duration (in minutes) to prevent calls from utilizing valuable device resources that could otherwise be used for additional new calls. If a call exceeds this duration, the device terminates the call.

This is configured on the Advanced Parameters page (**Configuration** tab > **VoIP** > **SIP Definitions** > **Advanced Parameters**) using the 'Max Call Duration' field:

Figure 2: Configured Maximum Call Duration

| | | |
|-------------------------|----|---|
| Max Call Duration [min] | 45 |  |
|-------------------------|----|---|

This page is intentionally left blank.

8 Define SIP Message Blacklist/Whitelist

It is recommended to configure SIP message policy rules for blocking (blacklist) unwanted incoming SIP messages or allowing (whitelist) receipt of desired messages. This allows you to define legal and illegal characteristics of a SIP message.

SIP message policy is helpful against VoIP fuzzing (also known as robustness testing), which sends different types of packets to its "victims" for finding bugs and vulnerabilities. For example, the attacker might try sending a SIP message containing an oversized parameter or too many occurrences of a parameter.

Each SIP message policy rule can be configured with, for example, maximum message length, header length, body length, number of headers, and number of bodies. Each rule is then set as a blacklist or whitelist.

The SIP message policy rules are configured in the Message Policy table (**Configuration** tab > **VoIP** > **SIP Definitions** > **Message Policy Table**). Below shows a configured policy that defines maximum SIP messages to 32,768 characters, headers to 256 characters, bodies to 512 characters, number of headers to 16, and permits up to 8 bodies. Invalid requests are rejected. Only INVITE and BYE requests are permitted.

Figure 8-1: Configured Message Policy Rule

The screenshot displays the 'Message Policy Table' configuration window. At the top, there are buttons for 'Add +', 'Edit', 'Delete', and 'Show/Hide'. A search bar is also present. The table below has the following data:

| Index | Name | Max Message Length | Max Header Length | Max Body Length | Send Rejection |
|-------|---------|--------------------|-------------------|-----------------|----------------|
| 0 | Example | 32768 | 256 | 512 | Policy Reject |

Below the table, the 'Selected Row #0' details are shown:

| | | | |
|---------------------|---------------|-------------------|------------------|
| Index: | 0 | Max Num Headers: | 16 |
| Name: | Example | Max Num Bodies: | 8 |
| Max Message Length: | 32768 | Method List: | invite/bye |
| Max Header Length: | 256 | Method List Type: | Policy Whitelist |
| Max Body Length: | 512 | Body List: | |
| Send Rejection: | Policy Reject | Body List Type: | Policy WhiteList |

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9 Monitor and Log Events

It is highly recommended that you log and monitor device events (including device operations and calls). The importance of monitoring device events is that you can quickly detect unauthorized access and subsequently take counter measures to effectively terminate the attacker before any potential damage is done to your network.

9.1 Implement Dynamic Blacklisting of Malicious Activity (IDS)

It is important to configure the Intrusion Detection System feature (IDS) to enable the device to detect malicious attacks targeted on the device (e.g., DoS, SPAM, and Theft of Service). It is crucial to be aware of any attacks to ensure the legitimate call service is maintained at all times. If any user-defined attacks are identified, the device can do the following:

- Block (blacklist) remote hosts (IP addresses / ports) considered as malicious. The device automatically blacklists the malicious source for a user-defined period after which it is removed from the blacklist.
- Send SNMP traps to notify of the malicious activity and/or whether an attacker has been added to or removed from the blacklist.

The IDS configuration is based on IDS Policies, where each policy can be configured with a set of IDS rules. Each rule defines a type of malicious attack to detect and the number of attacks (alarm threshold) during an interval (threshold window) before an SNMP trap is sent. Each policy is then applied to a target under attack (SIP Interface) and/or source of attack (Proxy Set and/or subnet address).

For configuring IDS, use the tables under the **Intrusion Detection and Prevention** menu (**Configuration** tab > **VoIP** menu > **Security** > **Intrusion Detection and Prevention**):

- **Global Parameters** – enables IDS
- **Policy Table** – defines IDS Policies and rules
- **Match Table** – assigns the IDS Policies to targets under attack (SIP Interface) and/or source of attacks (Proxy Set and/or subnet address)

Below is an example of an IDS rule for identifying DoS attacks from ITSP:

1. IDS Policy "ITSP DOS" configured (highlighted):

Table 9-1: Configured IDS Policy Name in IDS Policy Table

| Index ↕ | Name | Description |
|---------|----------------|---------------------------------|
| 0 | DEFAULT_FEU | Default policy for FEU |
| 1 | DEFAULT_PROXY | Default policy for proxies |
| 2 | DEFAULT_GLOBAL | Default policy for global scope |
| 3 | ITSP DoS | Denial of Service |

2. IDS Rules configured for "ITSP DOS" IDS policy:

Table 9-2: Configured Rules in IDS Rule Table

| Index ↕ | Reason | Threshold Scope | Threshold Window | Minor-Alarm Threshold | Major-Alarm Threshold | Critical-Alarm Threshold | Deny Threshold | Deny Period |
|---------|------------------------|-----------------|------------------|-----------------------|-----------------------|--------------------------|----------------|-------------|
| 0 | Malformed message | Global | 30 | 10 | 15 | 30 | -1 | -1 |
| 1 | Connection abuse | Global | 20 | -1 | 70 | -1 | -1 | -1 |
| 2 | Authentication failure | Global | 1 | -1 | 5 | -1 | -1 | -1 |

3. IDS Policy assigned to a specific SIP interface and subnet:

Table 9-3: Applying IDS Policy to Elements in IDS Match Table

| Index ↕ | SIP Interface ID | Proxy Set ID | Subnet | Policy |
|---------|------------------|--------------|--------------|----------|
| 0 | 1 | | | ITSP DoS |
| 1 | | | 10.33.0.0/16 | ITSP DoS |

9.2 Enable Syslog

The device supports the generation and reporting of Syslog messages and SNMP traps to external logging servers. It is crucial that you enable one or both of these features (preferably Syslog) so that you can monitor events on your device. In addition, as the device does not retain logged reports (SNMP is limited), it is recommended that you ensure that your Syslog server saves all logged events for future analysis and reference.

This is configured on the Syslog Settings page (**Configuration** tab > **System** > **Syslog Settings**), as shown below:

Figure 2: Enabling Syslog Server

| ▼ Syslog Settings | |
|--------------------------|------------|
| Enable Syslog | Enable ▼ |
| Syslog Server IP Address | 10.15.8.8 |
| Syslog Server Port | 514 |
| Syslog CPU Protection | Enabled ▼ |
| Syslog Optimization | Disabled ▼ |
| Debug Level | Detailed ▼ |

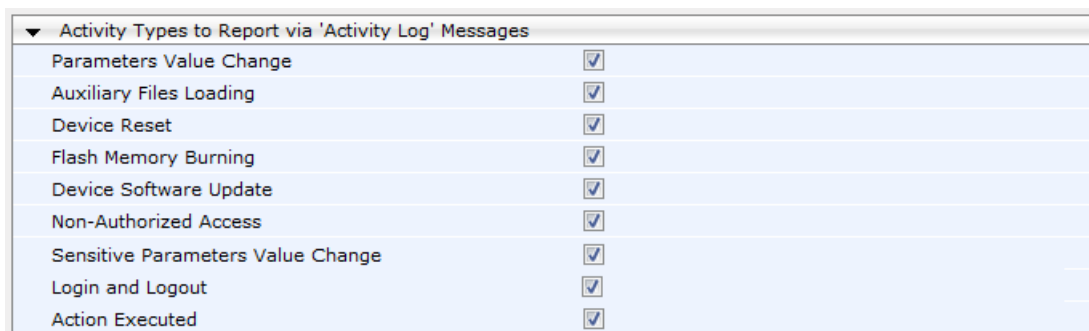
9.3 Enable Logging of Management-Related Events

Through Syslog, you can log and monitor management-related events to help you detect and identify unauthorized management-related activities such as:

- Unauthorized Web login attempts (attempts to access the Web interface with a false or empty user name or password)
- Access to restricted Web pages such as the page on which firewall rules are defined
- Modifications to parameter values (for example, deletion of firewall rules, allowing future unauthorized access)
- Modifications to "sensitive" parameters - changes made to important parameters such as IP addresses
- Unauthorized SIP messages (logged SIP messages)

This is configured in the Activity Types to Report section on the Syslog Settings page (**Configuration** tab > **System** > **Syslog Settings**), as shown below:

Figure 3: Enabling Logging of Management Events to a Syslog Server



| ▼ Activity Types to Report via 'Activity Log' Messages | |
|--|-------------------------------------|
| Parameters Value Change | <input checked="" type="checkbox"/> |
| Auxiliary Files Loading | <input checked="" type="checkbox"/> |
| Device Reset | <input checked="" type="checkbox"/> |
| Flash Memory Burning | <input checked="" type="checkbox"/> |
| Device Software Update | <input checked="" type="checkbox"/> |
| Non-Authorized Access | <input checked="" type="checkbox"/> |
| Sensitive Parameters Value Change | <input checked="" type="checkbox"/> |
| Login and Logout | <input checked="" type="checkbox"/> |
| Action Executed | <input checked="" type="checkbox"/> |

9.4 Enable Call Detail Records

Call Detail Records (CDR) provide vital information on SIP calls made through the device. This information includes numerous attributes related to the SIP call such as port number, physical channel number, source IP address, call duration, and termination reason (provided by the CDR field *TrmReason*). The device can be configured to generate and report CDRs for various stages of the call (beginning, initial connection, and end of the call). Once generated, the CDR logs are sent to a user-defined logging server.

This is configured on the Advanced Parameters page (**Configuration** tab > **System** > **Call Detail Record** > **Call Detail Record Settings**), as shown below:

Figure 4: Enabling CDR Generation

| ▼ CDR and Debug | |
|----------------------------|--------------------|
| CDR Server IP Address | 10.15.8.1 |
| CDR Report Level | Start & End Call ▼ |
| Media CDR Report Level | End Media ▼ |
| CDR Syslog Sequence Number | Enable ▼ |



Note: Syslog must be enabled for the CDR feature.

10 SBC-Specific Security Guidelines

This section provides basic SBC security guidelines that should be implemented in your network deployment.



Note: This section is applicable only to AudioCodes Session Border Controllers (SBC).

10.1 General Guidelines

It is crucial that you separate trusted from un-trusted networks:

- Separate un-trusted networks from trusted networks, by using different SRDs, IP Groups, SIP Interfaces, and SIP Media Realms (with limited port range).
- Similarly, separate un-trusted networks from one another. In particular, far-end users must be separated from the ITSP SIP trunk, using a different SRD, IP Group, SIP interface, and SIP Media Realms. This separation helps in preventing attacks targeted on far-end user ports from affecting other users.
- For un-trusted networks, use strict classification rules over vague rules. For example, if the ITSP's proxy IP address, port and host name are known, then use them in the classification rules. This ensures that all other potentially malicious SIP traffic is rejected.
- Unclassified packets must be discarded (rejected).

10.2 Secure Media (RTP) Traffic using SRTP

It is recommended to use Secured RTP (SRTP) for encrypting the media (RTP and RTCP) path and thereby, protecting the VoIP traffic. The device supports SRTP according to RFC 3711. SRTP performs a Key Exchange mechanism (according to RFC 4568). This is done by adding a 'crypto' attribute to the SDP. This attribute is used (by both sides) to declare the supported cipher suites and to attach the encryption key. If negotiation of the encryption data is successful, the call is established. The device's SRTP feature supports various suites such as AES_CM_128_HMAC_SHA1_32.

- Globally: Media Security page (**Configuration** tab > **VoIP** > **Media** > **Media Security**) by setting the 'Media Security' parameter to **Enable**.

Figure 5: Enabling SRTP Globally

| General Media Security Settings | |
|---|------------|
| Media Security | Enable |
| Media Security Behavior | Preferable |
| Authentication On Transmitted RTP Packets | Active |
| Encryption On Transmitted RTP Packets | Active |
| Encryption On Transmitted RTCP Packets | Active |
| SRTP Tunneling Authentication for RTP | Disable |
| SRTP Tunneling Authentication for RTCP | Disable |

- SRTP is enforced on the SBC legs, using IP Profiles (**Configuration** tab > **VoIP** > **Coder and Profiles** > **IP Profiles**). For each IP Profile associated with a leg, set the 'SBC Media Security Behavior' parameter to **SRTP**. This enforces the SBC legs to negotiate only SRTP media lines; RTP media lines are removed from the incoming SDP offer \ answer.

Figure 6: Enforcing SRTP per SBC Leg

Add Row ✕

Index:

Common | GW | SBC Signaling | **SBC Media**

| | |
|----------------------------|------------------|
| Transcoding Mode | Only If Required |
| Extension Coders | None |
| Allowed Audio Coders | None |
| Allowed Coders Mode | Restriction |
| Allowed Video Coders | None |
| Allowed Media Types | |
| SBC Media Security Mode | SRTP |
| Media Security Method | SDES |
| Enforce MKI Size | Don't enforce |
| SDP Remove Crypto Lifetime | No |
| RFC 2833 Mode | As Is |
| Alternative DTMF Method | As Is |
| RFC 2833 DTMF Payload Type | 0 |

Add Cancel

10.3 Implement SIP Authentication and Encryption

It is paramount that your network implements authentication and encryption to secure the network and ensure integrity and confidentiality of sensitive communications over untrusted networks. Some of the main authentication and encryption guidelines are discussed in the subsequent sections.

10.3.1 Authenticating Users as an Authentication Server

Instead of relying on external, third-party authentication servers, the device can be configured to act as an Authentication server, performing authentication and validation challenges with SIP user agents. The SIP method (INVITE or REGISTER) on which it challenges can be defined. If the message is received without an Authorization header, the device challenges the client by sending a 401 or 407 SIP response. The client then resends the request with an Authorization header containing its username and password. The device validates the SIP message and if it fails, the message is rejected and the device sends a 403 "Forbidden" response. If the SIP message is validated, the device verifies identification of the UA by checking whether the username and password received from the user is correct. The usernames and passwords are obtained from the User Information table. If after three attempts the UA is not successfully authenticated, the device sends a 403 "Forbidden" response. The device can also perform authentication on behalf of its UAs with an external third-party server.

To setup the SBC as an Authentication server, you need to do the following:

1. Set the following parameters in the User-type IP Group of the UAs (**Configuration** tab > **VoIP** > **VoIP Network** > **IP Group Table**):
 - 'Authentication Mode': select **SBC as Server**.
 - 'Authentication Method List': enter "INVITE\REGISTER".

Figure 7: Setting SBC as Authentication Server for User IP Group

The screenshot shows the 'Add Row' configuration window for a User-type IP Group. The window has three tabs: 'Common', 'GW', and 'SBC'. The 'SBC' tab is selected. The configuration parameters are as follows:

| Parameter | Value |
|--|-------------------------|
| Index | 1 |
| SRD | DefaultSRD |
| SBC Operation Mode | Not Configured |
| Classify By Proxy Set | Enable |
| SBC Client Forking Mode | Sequential |
| Inbound Message Manipulation Set | -1 |
| Outbound Message Manipulation Set | -1 |
| Message Manipulation User-Defined String 1 | |
| Message Manipulation User-Defined String 2 | |
| Registration Mode | User Initiates Registra |
| Max. Number of Registered Users | -1 |
| Authentication Mode | SBC as Server |
| Authentication Method List | INVITE\REGISTER |

Two arrows on the left side of the window point to the 'Authentication Mode' and 'Authentication Method List' fields, indicating the settings to be modified.

2. Enable the use of the SBC User Info table, by setting the 'Enable User-Information Usage' parameter to **Enable** on the Advanced Parameters page (**Configuration > VoIP > SIP Definitions > Advanced Parameters**).

Figure 8: Enabling User Info File Usage


3. Create a User Info table: Add users with authentication usernames and passwords in the SBC User Info table (**Configuration tab > VoIP menu > SIP Definitions > User Information > SBC User Info Table**)

Figure 9: Configured User in SBC User Info Table

| Index ↕ | Local User | Username | Password | IP Group | Status |
|---------|------------|----------|----------|--------------------|----------------|
| 0 | John Dee | johnd | * | Application-Server | Not Registered |

10.3.2 Authenticating Users by RADIUS Server

Instead of authorizing calls locally, digest authentication of SIP users can be done by a RADIUS server (according to RFC 5090). In this way, the device offloads the MD5 calculation (validation) to the RADIUS server, where the device is classed as a RADIUS client.

To implement this, the following configuration is required:

1. Configure the RADIUS sever (IP address, port and shared secret password) in the RADIUS Servers table (**Configuration tab > System menu > Management > RADIUS Servers**).

Figure 10: Configured RADIUS Server for User Authentication in RADIUS Servers Table

| Index ↕ | IP Address | Authentication Port | Accounting Port | Shared Secret |
|---------|-------------|---------------------|-----------------|---------------|
| 0 | 202.100.0.2 | 1645 | 0 | * |

2. Set the SBCServerAuthMode parameter to 1 to enable authentication by a RFC 5090 compliant RADIUS server.

10.3.3 Authenticating SIP Servers as an Authentication Server

It is recommended to enable the device to authenticate remote SIP servers (for example, Proxy servers). This provides protection from rogue SIP servers, preventing unauthorized usage of the device's resources and functionality. The device authenticates remote servers by challenging them with a user-defined username and password that is shared with the remote server. From such a challenge, the device can confirm the server's identity as being genuine. The type of SIP message (e.g., INVITE) to authenticate must also be defined.

SIP server authentication is configured per IP Group in the IP Group table, enabling unique authentication handling per specific server:

Figure 11: Configured SIP Server Authentication by SBC in IP Group Table

The screenshot shows the 'Add Row' configuration window for SIP Server Authentication by SBC in IP Group Table. The window has a title bar with 'Add Row' and a close button. Below the title bar, there is a dropdown menu for 'SRD' set to 'DefaultSRD'. There are three tabs: 'Common', 'GW', and 'SBC', with 'SBC' being the active tab. The configuration options are as follows:

| Field | Value |
|--|-------------------------|
| SBC Operation Mode | Not Configured |
| Classify By Proxy Set | Enable |
| SBC Client Forking Mode | Sequential |
| Inbound Message Manipulation Set | -1 |
| Outbound Message Manipulation Set | -1 |
| Message Manipulation User-Defined String 1 | |
| Message Manipulation User-Defined String 2 | |
| Registration Mode | User Initiates Registra |
| Max. Number of Registered Users | -1 |
| Authentication Mode | SBC as Server |
| Authentication Method List | INVITE |
| Username | ipbx_foo |
| Password | |

At the bottom of the window, there are 'Add' and 'Cancel' buttons. Four arrows on the left side of the window point to the 'Authentication Mode', 'Authentication Method List', 'Username', and 'Password' fields.

10.3.4 Enforce SIP Client Authentication by SIP Proxy

When the device is located between a SIP client and a third-party SIP proxy server and SIP Digest Authentication is used, the device relays authentication messages between these entities. Although the device gathers and maintains some information in its registration database (Address of Record / AOR) it does not actively participate in the authentication process. Instead, it is the SIP proxy that handles and enforces SIP client authentication. Therefore, it is imperative that your SIP proxy server be configured to enforce SIP client authentication.

10.3.5 Enforce SIP Digest Authentication by IP PBX

If TLS cannot be configured (for whatever reason) and if you are using an on-premise IP PBX, it is crucial that your IP PBX implements SIP Digest Authentication for remote users. In addition, authentication should be applied to as many SIP methods as possible (i.e., not only on REGISTER messages, but also INVITEs, re-INVITEs, etc.).

10.4 Secure Routing Rules

This section provides recommended security guidelines regarding routing rules.

10.4.1 Classify by Classification Rules versus Proxy Set

An important security functionality of the SBC is to make sure that it does not mistakenly identify incoming SIP dialog-initiating requests (e.g., INVITE messages) from malicious attackers as belonging to a configured server-type IP Group entity. The SBC provides two optional mechanisms that can be employed to identify incoming dialogs as coming from a specific server-type IP Group:

- **Classification rules in the Classification table:** Identifies incoming dialogs based on the characteristics of the SIP message such as host name in the INVITE message (Layer 4-7), and/or based on the source IP address (Layer-3).
- **Proxy Set:** Identifies incoming dialogs based on source IP address (Layer-3) only. The Proxy Set defines the address of the IP Group.

Regarding which classification method to employ, please adhere to the following guidelines:

- If the IP address of the IP Group entity is known, it is recommended to employ SIP dialog classification based on a Classification rule, where the rule is configured with not only the IP address, but also with SIP message characteristics to increase the strictness of the classification process. For more information on configuring Classification rules, see Section 49.
- If the IP address is unknown, in other words, the Proxy Set associated with the IP Group is configured with an FQDN, it is recommended to employ SIP dialog classification based on Proxy Set. This allows the SBC to classify the incoming dialog based on the DNS-resolved IP address. The reason for classifying by Proxy Set is that IP address forgery (commonly known as IP spoofing) is more difficult than malicious SIP message tampering and therefore, using a Classification rule without an IP address offers a weaker form of security. When classification is based on Proxy Set, the Classification table is ignored.

Classification by Proxy Set is enabled in the IP Group table, using the 'Classify By Proxy Set' parameter:

Figure 12: Enabling Classification based on Proxy Set in the IP Group Table

The screenshot shows a configuration window titled "Add Row" with a close button (X) in the top right corner. It contains the following elements:

- Index:** A text input field containing the number "1".
- SRD:** A dropdown menu currently showing "DefaultSRD".
- Tabs:** Four tabs are visible: "Common", "GW", "SBC" (which is selected and highlighted in orange), and "GW Group Status".
- SBC Operation Mode:** A dropdown menu currently showing "Not Configured".
- Classify By Proxy Set:** A dropdown menu currently showing "Enable". An arrow points to this dropdown from the left.
- Buttons:** "Add" and "Cancel" buttons are located at the bottom right of the window.

10.4.2 Define Strict Classification Rules

Classification rules are used to identify incoming SIP dialog-initiating requests (e.g., INVITE messages) and bond them to IP Groups. In other words, these rules identify the source of the call. Once the source IP Group is identified, the traffic can then be routed to its destination according to IP-to-IP routing rules.

When defining Classification rules, adhere to the following recommendations:

- For Server-type IP Groups, use Classification rules **only** if the IP address of the IP Group is known. If known, include the IP address in the Classification rule ('Source IP Address' parameter). In addition, to increase classification strictness, configure SIP message characteristics in the rule as well.



Note: If the IP address is unknown (i.e., the Proxy Set associated with the IP Group is configured with an FQDN), it is recommended to employ SIP dialog classification based on Proxy Set (see Section 10.4.1). In such a scenario, the Classification table is ignored and must not be configured for the specific IP Group.

- For Server-type IP Groups whose IP addresses are known, it is recommended to also configure VoIP firewall rules (see Section 3.1).
- Use strict Classification rules over vague ones so that all other potentially malicious SIP traffic is rejected. In other words, configure the rule with as much information as possible that accurately characterizes the incoming SIP dialog (e.g. source and destination host name).
- Define a range for the source and destination prefix numbers.
- Define a combination of Classification rules to guarantee correct and accurate identity of sender of call.
- Use Message Condition rules to increase the strictness of the Classification process. Message Condition rules enhance the process of classifying incoming SIP dialogs to an IP Group. When a Classification rule is associated with a Message Condition rule, the Classification rule is used only if its' associated Message Condition rule are matched. Message Condition rules are SIP message conditions based on the same syntax used in the Message Manipulations table. You can define complex rules using the "AND" or "OR" Boolean operands. You can also use regular expressions (regex) as Message Condition rules, for example:
 - "body.sdp regex pcmu" can be used to enable routing based on the offered codec (G.711 Mu) in the incoming SDP message
 - "body.sdp regex (AVP[0-9]|\s)*\s8[\s|\n])" can be used to enable routing based on payload type 8 in the incoming SDP message

Message Condition rules are configured in the Message Condition table (**Configuration** tab > **VoIP** > **SBC** > **Routing SBC** > **Message Condition Table**) and then later associated with Classification rules in the Classification table. The figure below shows a Message Condition rule for P-Asserted-Identity headers that contain "abc":

Figure 13: Configured Message Condition Rule in Message Condition Table

| Index ↕ | Name | Condition |
|---------|-------------------------------|--|
| 0 | P-Asserted-Identity has "abc" | header.p-asserted-identity.url.user contains " |

- The last Classification rule in the Classification table should be one that denies all calls. This is done by using the asterisk (*) symbol and the **Any** option for the matching characteristics (under the **Rule** tab), and then setting the 'Action Type' parameter to **Deny** (under the **Action** tab).

Figure 14: Last Classification Rule in Classification Table to Deny All Other Calls

Classification rules are configured in the Classification table (**Configuration** tab > **VoIP** > **SBC** > **Routing SBC** > **Classification Table**). The figure below shows an example of two Classification rules:

- **Index 0 "ITSP"**: Classifies received calls to Server-type IP Group "ITSP" if they have the following incoming matching characteristics:
 - 'Source SIP Interface': 10.15.7.96
 - 'Source Username Prefix': 2 through 4
 - 'Source Host': domain.com
 - 'Destination Username Prefix': 1 through 7
 - 'Message Condition': SIP message with P-Asserted-Identity header containing "abc" (Message Condition rule described previously in this section)
- **Index 2 "Deny"**: Denies calls that cannot be classified (unknown calls).

Figure 15: Configured Classification Rules in Classification Table

| Index | Name | SRD | Source SIP Interface | Source Username Prefix | Source Host | Destination Username Prefix | Destination Host | Action Type | Source IP Group |
|-------|------|-----------------|----------------------|------------------------|-------------|-----------------------------|------------------|-------------|-----------------|
| 0 | ITSP | DefaultSRD (#0) | Any | [2-4] | domain.com | [1-7] | * | Allow | ITSP |
| 1 | Deny | DefaultSRD (#0) | Any | * | * | * | * | Deny | None |

Selected Row #0

| Rule | | Action | |
|------------------------------|-------------------------------|-----------------------------|-------|
| Name: | ITSP | Action Type: | Allow |
| Source SIP Interface: | Any | Source IP Group: | ITSP |
| Source Username Prefix: | [2-4] | Destination Routing Policy: | None |
| Source Host: | domain.com | IP Profile: | None |
| Destination Username Prefix: | [1-7] | | |
| Destination Host: | * | | |
| Message Condition: | P-Asserted-Identity has "abc" | | |
| Source IP Address: | 10.15.7.96 | | |
| Source Port: | 0 | | |

10.4.3 Allow Calls Only with Specific SIP User-Agent Header Value

The SIP User-Agent header contains information about the User Agent Client (UAC) initiating the SIP dialog request. This information is unique to the Enterprise and therefore, it is recommended to configure your device so that it only accepts calls that have a specified User-Agent header value. This is configured by adding a Message Condition rule (in the Message Condition table) for this SIP header type and then assigning it to a Classification rule (in the Classification table).

The figure below shows a Message Condition rule in the Message Condition table (**Configuration** tab > **VoIP** > **SBC** > **Routing SBC** > **Message Condition Table**) whose condition is for the SIP User-Agent header to have the value "abc.com":

Figure 16: Message Condition Rule for SIP User-Agent Header in Message Condition Table

| Index ↕ | Name | Condition |
|---------|---------------------------|-----------------------------|
| 0 | Accept only abc.com calls | header.user-agent='abc.com' |

To apply the Message Condition rule, assign it to the Classification rule in the Classification table (**Configuration** tab > **VoIP** > **SBC** > **Routing SBC** > **Classification Table**):

Figure 17: Assigning Message Condition Rule to Classification Rule in Classification Table

The screenshot shows the 'Add Row' dialog box for configuring a classification rule. The 'Index' field is set to 2 and the 'SRD' dropdown is set to 'DefaultSRD'. Under the 'Rule' tab, the 'Message Condition' field is set to 'Accept only abc.com' and is highlighted with an arrow. Other fields include Name, Source SIP Interface (Any), Source IP Address, Source Transport Type (Any), Source Port (0), Source Username Prefix (*), Source Host (*), Destination Username Prefix (*), and Destination Host (*). At the bottom, there are 'Add' and 'Cancel' buttons and a 'Classic View' link.

10.4.4 Block Unclassified Calls

It is recommended that you block incoming calls that cannot be classified to an IP Group, based on the rules in the Classification table (discussed in the previous section). If unclassified calls are not blocked, they are sent to the default SRD / IP Group and therefore, illegitimate calls can be established.

This is configured on the General Settings page (**Configuration** tab > **VoIP** > **SBC** > **SBC General Settings**), by setting the Unclassified Calls' parameter to **Reject**:

Figure 18: Blocking Unclassified Incoming Calls

| | |
|---|----------------|
| SBC Survivability Registration Time [sec] | 0 |
| Forking Handling Mode | Latch On First |
| Unclassified Calls | Reject |
| Session-Expires [sec] | 180 |
| Direct Media | Disable |

10.4.5 Define Strict Routing Rules

It is crucial that you adhere to the following guidelines when configuring IP-to-IP Routing rules:

- Ensure that your routing rules are accurate and correctly defined. Inaccurate or weak routing rules can easily result in Service Theft.
- Ensure that your routing rules from **source IP Group** to **destination IP Group** are accurately defined for the desired call routing outcome.
- If possible, avoid using the asterisk (*) symbol to indicate "any" for a specific parameter in your routing rule. This constitutes weak routing rules that can be vulnerable to attackers. For strong routing rules, enter specific alphanumeric values instead of the asterisk.

10.5 Define Call Admission Control Rules

It is recommended to define call admission control (CAC) rules for regulating VoIP traffic volume. CAC rules can assist in limiting the rate of call requests, preventing excessive signaling requests originating from malicious and legitimate sources from overwhelming your network resources.

CAC rules can limit the number of concurrent calls (SIP dialogs) per IP Group, SIP Interface or SRD. The call limitation can be defined per SIP-dialog initiating request type (e.g., INVITE or REGISTER messages), request direction (inbound, outbound, or both), and user. Requests that exceed the user-defined limits are rejected (with SIP 480 "Temporarily Unavailable" responses). You can also limit the incoming packet rate based on the "token bucket" mechanism.

Adhere to the following CAC recommendations:

- It is crucial that your CAC rules include call limitations per user. This ensures that a user does not make unlimited, simultaneous calls.
- Define rules as specific as possible. For example, instead of defining one rule for all SIP request types, create rules for each SIP request type.

Note that if the call routing to a specific IP Group is blocked due to a CAC rule, the device searches for an alternative route (if defined) in the SBC IP-to-IP Routing table. If this alternative route does not exceed the CAC rule limitation, the device uses it to route the call.

The CAC rules are configured in the Admission Control table (**Configuration** tab > **VoIP** > **SBC** > **Admission Control**). The figure below displays a CAC rule that defines a maximum of 100 concurrent SIP dialog-initiating requests for IP Group "ITSP". SIP requests received above this threshold are rejected:

Figure 19: Configured CAC Rule in Admission Control Table

| Index ↕ | Name | Limit Type | SRD | IP Group | SIP Interface | Request Type | Request Direction | Limit | Limit per User |
|---------|----------------|------------|------|----------|---------------|--------------|-------------------|-------|----------------|
| 0 | Max Calls ITSP | IP Group | None | ITSP | None | All | Both | 100 | -1 |

10.6 Secure SIP User Agent Registration

Service theft can result from a lack of security in the SIP user registration process. This section provides recommended guidelines regarding user registration.

10.6.1 Configure Identical Registration Intervals

In scenarios where the device does not forward user registrations to a server (e.g., a PBX) and the device receives a new REGISTER request from the same number (i.e., same AOR) but without an Authentication header, the device still sends a SIP 200 OK response to the user. This is because the AOR already exists in the device's registration database. Therefore, if an illegitimate user attempts to connect with a legitimate IP address and phone number (without authentication), the malicious user is able to connect and steal calls.

To overcome this issue and prevent stealing of calls, ensure that you set the user and proxy registration times with **identical** values. On the SBC General Settings page (**Configuration** tab > **VoIP** > **SBC** > **SBC General Settings**), do the following:

- Define the duration of the periodic registrations between the user and the device in the 'SBC User Registration Time' field.
- Define the time interval (in seconds) that the device must register to the server (e.g., PBX) in the 'SBC Proxy Registration Time' field.

Figure 20: Configured User Registration Time on SBC General Settings Page

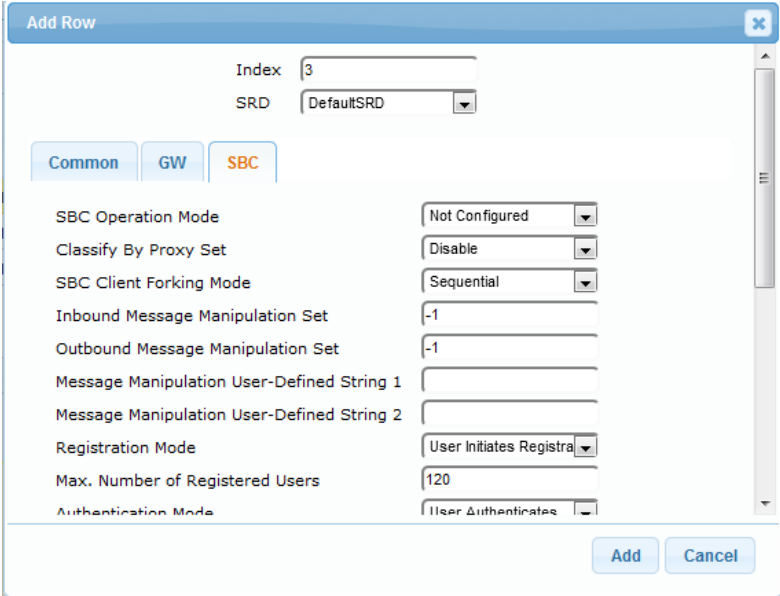
| | | |
|---|-----------------------------------|-----|
| → | SBC User Registration Time [sec] | 100 |
| → | SBC Proxy Registration Time [sec] | 100 |

10.6.2 Limit SBC Registered Users per IP Group, SIP Interface or SRD

It is recommended that you define a maximum number of allowed registered users per IP Group (User-type IP Group), SIP Interface or SRD. This ensures that illegitimate users are blocked from registering with the IP Group. This can be configured in the IP Group table, SIP Interface table or SRD table, by using the 'Max. Number of Registered Users' parameter.

- Per IP Group: IP Group table (**Configuration** tab > **VoIP** > **VoIP Network** > **IP Group Table**):

Figure 21: Configured Maximum Registered Users per User-Type IP Group in IP Group Table



The screenshot shows the 'Add Row' configuration window for the IP Group Table. The window has a title bar with 'Add Row' and a close button. Below the title bar, there are fields for 'Index' (set to 3) and 'SRD' (set to DefaultSRD). There are three tabs: 'Common', 'GW', and 'SBC'. The 'SBC' tab is selected. The 'SBC' tab contains several configuration options:

| Parameter | Value |
|--|-------------------------|
| SBC Operation Mode | Not Configured |
| Classify By Proxy Set | Disable |
| SBC Client Forking Mode | Sequential |
| Inbound Message Manipulation Set | -1 |
| Outbound Message Manipulation Set | -1 |
| Message Manipulation User-Defined String 1 | |
| Message Manipulation User-Defined String 2 | |
| Registration Mode | User Initiates Registra |
| Max. Number of Registered Users | 120 |
| Authentication Mode | User Authenticates |

An arrow points to the 'Max. Number of Registered Users' field, which is set to 120. At the bottom right of the window, there are 'Add' and 'Cancel' buttons.

- Per SIP Interface: SIP Interface table (**Configuration** tab > **VoIP** > **VoIP Network** > **SIP Interface Table**):

Figure 22: Configured Maximum Registered Users per SIP Interface in SIP Interface Table

The screenshot shows a configuration window titled "Add Row" with a close button in the top right corner. The window contains a list of configuration parameters for a SIP interface, each with a corresponding input field or dropdown menu. An arrow points to the "Max. Number of Registered Users" field, which is set to "120".

| | |
|---------------------------------|------------------|
| Index | 3 |
| SRD | DefaultSRD |
| Name | Nomadic Users |
| Network Interface | O+M+C |
| Application Type | SBC |
| UDP Port | 5060 |
| TCP Port | 5060 |
| TLS Port | 5061 |
| Encapsulating Protocol | No encapsulation |
| Media Realm | None |
| SBC Direct Media | Disable |
| TLS Context Name | default |
| TLS Mutual Authentication | |
| Block Unregistered Users | Yes |
| Max. Number of Registered Users | 120 |

At the bottom right of the dialog, there are two buttons: "Add" and "Cancel".

10.6.3 Block Calls from Unregistered Users

Ensure that calls from unregistered users are blocked (rejected) and that calls from only registered users are allowed. This can be configured per SRD or SIP Interface, by setting the 'Block Unregistered Users' parameter to **Yes**:

- Per SRD: SRD table (**Configuration** tab > **VoIP** > **VoIP Network** > **SRD Table**):

Figure 23: Blocking Unregistered Users

| Edit Row | |
|---------------------------------------|---------------------|
| Index | 0 |
| Name | DefaultSRD |
| Sharing Policy | Shared |
| SBC Operation Mode | B2BUA |
| SBC Routing Policy | Default_SBCRoutingP |
| Max. Number of Registered Users | 120 |
| Block Unregistered Users | Yes |
| Enable Un-Authenticated Registrations | Enable |

- Per SIP Interface: SIP Interface table (**Configuration** tab > **VoIP** > **VoIP Network** > **SIP Interface Table**):

Figure 24: Block Registered Users per SIP Interface in SIP Interface Table

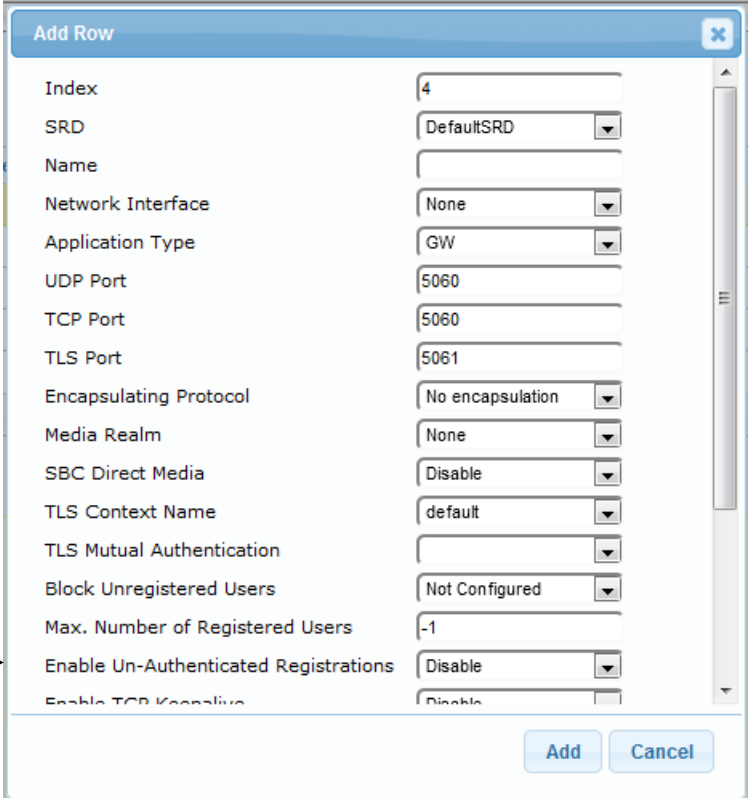
| Add Row | |
|---------------------------|------------------|
| Index | 3 |
| SRD | DefaultSRD |
| Name | Nomadic Users |
| Network Interface | O+M+C |
| Application Type | SBC |
| UDP Port | 5060 |
| TCP Port | 5060 |
| TLS Port | 5061 |
| Encapsulating Protocol | No encapsulation |
| Media Realm | None |
| SBC Direct Media | Disable |
| TLS Context Name | default |
| TLS Mutual Authentication | |
| Block Unregistered Users | Yes |

10.6.4 Block Registration from Un-Authenticated New Users

In normal operation scenarios in which a SIP proxy (registrar) server is available, the SBC forwards REGISTER requests from new users to the proxy, and if authenticated by the proxy (i.e., SBC receives a success response) the SBC adds the user to its registration database. However, if the proxy becomes unavailable at any time (e.g., due to network connectivity loss), the REGISTER requests cannot therefore be authenticated. In such scenarios, make sure that the SBC is configured to reject such unauthenticated REGISTER messages from new users. Note that the SBC does accept registration refreshes from users already in its database.

Blocking registration of un-authenticated users can be configured per SRD or SIP Interface, by setting the 'Enable Un-Authenticated Registrations' parameter to **Disable**. The following figure shows the SIP Interface configuration (**Configuration** tab > **VoIP** > **VoIP Network** > **SIP Interface Table**).

Figure 25: Blocking Local Registration of Un-Authenticated Users in SIP Interface Table



| Add Row | |
|---------------------------------------|------------------|
| Index | 4 |
| SRD | DefaultSRD |
| Name | |
| Network Interface | None |
| Application Type | GW |
| UDP Port | 5060 |
| TCP Port | 5060 |
| TLS Port | 5061 |
| Encapsulating Protocol | No encapsulation |
| Media Realm | None |
| SBC Direct Media | Disable |
| TLS Context Name | default |
| TLS Mutual Authentication | |
| Block Unregistered Users | Not Configured |
| Max. Number of Registered Users | -1 |
| Enable Un-Authenticated Registrations | Disable |
| Enable TCP Keepalive | Disable |

Add Cancel

10.7 Authenticate SIP BYE Messages

It is recommended to enable the device to authenticate all incoming SIP BYE requests before it releases the call. This prevents, for example, a scenario in which the SBC SIP client receives a BYE request from a third-party imposer assuming the identity of a participant in the call and as a consequence, the call between the legitimate first and second parties is inappropriately disconnected.

When the SBC is configured to authenticate BYE messages, it sends a SIP authentication response to the sender of the BYE request and waits for the sender (user) to authenticate it. The call is disconnected only if the authenticating server responds with a 200 OK.

This is configured on the SBC General Settings page (**Configuration** tab > **VoIP** > **SBC** > **SBC General Settings**) by setting the 'BYE Authentication' to **Enable**.

Figure 10-26: Enabling SIP BYE Authentication



10.8 Use SIP Message Manipulation for Topology Hiding

The device intrinsically employs topology hiding, limiting the amount of topology information displayed to external parties (i.e., un-trusted networks). This anonymous information minimizes the chances of directed attacks on your network.

The device employs topology hiding by implementing back-to-back user agent (B2BUA) leg routing:

- Strips all incoming SIP Via header fields and creates a new Via value for the outgoing message
- Each leg has its own Route/Record Route set
- Generates a new SIP Call-ID header value (different between legs)
- Changes the SIP Contact header to the device's address
- Performs Layer-3 topology hiding by modifying the source IP address in the SIP IP header (for example, IP addresses of ITSPs equipment such as proxies, gateways, and application servers can be hidden from outside parties)

In addition, to enhance topology hiding, you can modify the SIP To header, From header, and/or Request-URI host name. This can be done using the Message Manipulation table or the IP Group (for SIP URI host part manipulations). The Message Manipulation table also supports Regular Expressions (Regex).

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11 Gateway-Specific Security Guidelines

This section describes recommended security guidelines for the device's supporting the Gateway application. These guidelines are important for preventing malicious attacks such as DoS.

11.1 Block Calls from Unknown IP Addresses

Ensure that the device accepts incoming calls only from source IP addresses that are defined in the Proxy Set table or Outbound IP Routing table. In addition, if an FQDN is defined in these tables, the call is accepted only if the resolved DNS IP address of the call is defined in any one of these tables. All other calls whose source IP address is not defined in these tables are rejected. This is useful in preventing unwanted SIP calls, SIP messages, and VoIP spam.

This is configured on the Advanced Parameters page (**Configuration** tab > **VoIP** > **SIP Definitions** > **Advanced Parameters**), by setting the 'IP Security' parameter to **Secure All calls**:

Figure 27: Allowing Calls only from Defined IP Addresses



The screenshot shows a configuration window with a 'General' tab selected. Underneath, the 'IP Security' parameter is set to 'Secure Incoming calls' via a dropdown menu.

11.2 Enable Secure SIP (SIPS)

Ensure that you enable Secure SIP (SIPS) so that the device initiates TLS all the way to the destination, i.e., over multiple hops. SIPS runs SIP over TLS on a hop-by-hop basis. This is important as using TLS as a transport by itself guarantees only encryption over a single hop. Since it is very common for a SIP call to traverse multiple proxy servers from one end to the other, there is a need to guarantee end-to-end security for SIP traffic. A call to a SIPS URI is guaranteed to be encrypted from end to end. All SIP traffic within this call is secured using TLS from the sender to the domain of the final recipient.

This is configured , by setting the following parameters:

1. Enable secure SIP (SIPS): On the SIP General Parameters page (**Configuration** tab > **VoIP** > **SIP Definitions** > **General Parameters**), set the 'Enable SIPS' to **Enable**.

Figure 28: Defining SIP over TLS (For Gateway Application)

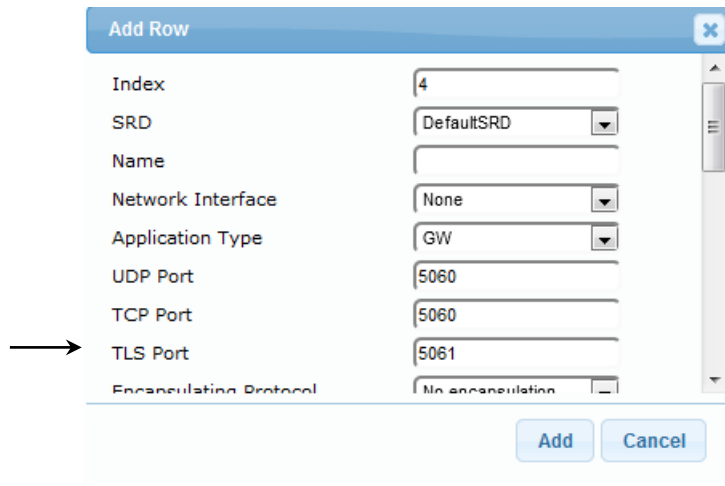


The screenshot shows a configuration window with the 'Enable SIPS' parameter set to 'Enable' via a dropdown menu.



Note: It is highly recommended to use the 'Enable SIPS' parameter and not the 'SIP Transport Type' parameter to define TLS. The 'SIP Transport Type' parameter provides only a TLS connection to the next network hop, whereas the 'Enable SIPS' parameter provides TLS to the final destination (over multiple hops).

2. Define the local SIP TLS port for the SIP Interface (**Configuration** tab > **VoIP** > **VoIP Network** > **SIP Interface Table**), using the 'TLS Port':



| Add Row | |
|------------------------|------------------|
| Index | 4 |
| SRD | DefaultSRD |
| Name | |
| Network Interface | None |
| Application Type | GW |
| UDP Port | 5060 |
| TCP Port | 5060 |
| TLS Port | 5061 |
| Encapsulating Protocol | No encapsulation |

11.3 Define Strict Routing Rules

When defining IP-to-Tel (IP-to-Trunk Group Routing table) and Tel-to-IP (Tel-to-IP Routing table) routing rules, it is crucial that you adhere to the following security guidelines:

- Ensure that your routing rules are accurate and correctly defined for the desired routing outcome. Inaccurate or "loose" routing rules can easily result in service theft.
- Avoid, if possible, using the asterisk "*" symbol and **Any** option to indicate any for a specific parameter in your routing rules. This constitutes weak routing rules that can be vulnerable to attackers. For strong routing rules, enter specific alphanumeric values instead of the asterisk.

11.4 Define Call Admission Control

Ensure that you set the maximum, allowed concurrent calls per routing rule or IP Group. This is done by defining a call limit for an IP Profile and then assigning the IP Profile to IP-to-Tel and/or Tel-to-IP routing rules, or IP Groups. Note that the maximum number of calls takes into account incoming and outgoing calls (i.e., summation of all calls to which the IP Profile is assigned).

This is configured in the 'Number of Calls Limit' field on the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**):

Figure 29: Configured Maximum Concurrent Calls for IP Profile in IP Profile Table (GW Tab)

The screenshot shows a configuration window titled "Add Row" with a close button (X) in the top right corner. The window contains a list of settings, each with a label and a corresponding input field (either a dropdown menu or a text box). An arrow points to the "Number of Calls Limit" field, which is set to "120".

| Setting | Value |
|--|------------------|
| CNG Detector Mode | Disable |
| Vxx Modem Transport Type | Enable Bypass |
| NSE Mode | Disable |
| Play RB Tone to IP | Disable |
| Early Media | Disable |
| Progress Indicator to IP | |
| Early 183 | Disable |
| Early Answer Timeout [sec] | 0 |
| Hold | Enable |
| Add IE In Setup | |
| QSIG Tunneling | Disable |
| Remote RTP Base UDP Port | 0 |
| Copy Destination Number to Redirect Number | Disable |
| Number of Calls Limit | 120 |
| AMD Mode | Don't Disconnect |

At the bottom right of the window, there is a link labeled "Classic View" and two buttons: "Add" and "Cancel".

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12 Network Port Assignment

The table below lists the device's network port assignments. This table also shows whether these ports are enabled or disabled by default and how to configure them. For ports that you do not need in your deployment but that are enabled by default, it is highly recommended that you disable them for security reasons.

Table 12-1: Network Port Assignments

| Port | Application | Default | Port Configuration |
|------|-------------------|---|---|
| 22 | SSH | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ LAN: <i>Enable SSH Server</i> (SSHServerEnable) ✓ WAN: <i>Allow WAN access to SSH</i> (Applicable only to MSBR) ▪ Port Definition: <ul style="list-style-type: none"> ✓ LAN: <i>Server Port</i> (SSHServerPort) ▪ Access Control: <i>Layer 3/4 Firewall and Web & Telnet Access List table</i> (WebAccessList_x) |
| 23 | Telnet | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ LAN: <i>Embedded Telnet Server</i> (TelnetServerEnable) ✓ WAN: <i>Allow WAN access to Telnet</i> (AllowWanTelnet) (Applicable only to MSBR) ▪ Port Definition: <ul style="list-style-type: none"> ✓ LAN: <i>Telnet Server TCP Port</i> (TelnetServerPort) ▪ Access Control: <i>Layer 3/4 Firewall and Web & Telnet Access List table</i> (WebAccessList_x) |
| 68 | DHCP | <ul style="list-style-type: none"> ▪ LAN: Disabled ▪ WAN: Enabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ LAN: <i>Enable DHCP</i> (DHCPEnable) ✓ WAN: <i>Web (VoIP and data-router) or CLI</i> ▪ Port Definition: Fixed ▪ Access Control: Firewall |
| 80 | Web server (HTTP) | <ul style="list-style-type: none"> ▪ LAN: Enabled ▪ WAN: Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ LAN: <i>Secured Web Connection</i> (HTTPSOOnly) – when set to HTTPS Only ✓ WAN: <i>Allow WAN access to HTTP</i> (AllowWanHTTP) (Applicable only to MSBR) ▪ Port Definition: <ul style="list-style-type: none"> ✓ LAN: <i>HTTPPort</i> ▪ Access Control: <i>Web & Telnet Access List table</i> (WebAccessList_x) |
| 161 | SNMP Traps | Enabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <i>Disable SNMP</i> (DisableSNMP) ▪ Port Definition: <i>SNMP Trap Destinations table</i> (SNMPManagerIsUsed_x) ▪ WAN: <i>Allow WAN access to SNMP</i> (AllowWanSNMP) |

| Port | Application | Default | Port Configuration |
|-----------------|--|---|---|
| | | | (Applicable only to MSBR) <ul style="list-style-type: none"> ▪ Access Control: N/A |
| 161 | SNMP GET / SET | <ul style="list-style-type: none"> ▪ LAN: Enabled ▪ WAN: Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <i>Disable SNMP</i> (DisableSNMP) ▪ Port Definition: <ul style="list-style-type: none"> ✓ LAN: <i>SNMPPort</i> ✓ WAN: <i>Allow WAN access to SNMP</i> (AllowWanSNMP) (Applicable only to MSBR) ▪ Access Control: <i>SNMP Trusted Managers table</i> (SNMPTrustedMgr_x) |
| 443 | Web server (HTTPS) | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ WAN: <i>Allow WAN access to HTTPS</i> (AllowWanHTTPS) (Applicable only to MSBR) ▪ Port Definition: <ul style="list-style-type: none"> ✓ LAN: <i>HTTPSPort</i> ▪ Access Control: <i>Web & Telnet Access List table</i> (WebAccessList_x) |
| 500 | IPSec IKE | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ LAN: Web (IPSec configuration) or IP Security Associations table ✓ WAN: Web data-router configuration or CLI (<i>crypto</i>) ▪ Port Definition: Fixed ▪ Access Control: Firewall (see Enable / Disable above) |
| 926 | Debugging Note: Applicable only to non-MSBR devices. | Enabled | Not configurable. |
| 4000, 4010, ... | RTP traffic | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: Enabled by SIP during RTP session establishment ▪ Port Definition: <ul style="list-style-type: none"> ✓ LAN: <i>RTP Base UDP Port</i> (BaseUDPport or IpProfile_RemoteBaseUDPPort); <i>Media Realm table</i> (CpMediaRealm) ✓ WAN: Media Realms defined on WAN interface (Applicable only to MSBR) ▪ Access Control: Layer 3/4 firewall |
| 4001, 4011, ... | RTCP traffic | Disabled | Always adjacent to the RTP port number |
| 4002, 4012, ... | T.38 traffic | Disabled | Always adjacent to the RTCP port number |
| 5060 | SIP (UDP / TCP) | Enabled Note: For hybrid devices (Gateway and SBC applications), port 5060 is enabled for the Gateway application only (SIP Interface Index 0), by default. | <ul style="list-style-type: none"> ▪ Enable / Disable: <i>SIP Interface table – UDP Port / TCP Port</i> (SIPInterface) ▪ Port Definition: SIP Interface table – UDP Port / TCP Port (SIPInterface) ▪ Access Control: N/A |

| Port | Application | Default | Port Configuration |
|--|---|---|--|
| 5061 | SIP over TLS (SIPS) | Enabled Note: For hybrid devices (Gateway and SBC applications), port 5061 is enabled for the Gateway application only (SIP Interface Index 0), by default. | <ul style="list-style-type: none"> ▪ Enable / Disable: SIP Interface table – TLS Port (SIPInterface) ▪ Port Definition: SIP Interface table – TLS Port (SIPInterface) ▪ Access Control: N/A |
| Arbitrary port for TCP client (not configurable) | LDAP Note: Applicable only to LAN. | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: LDAP Service (LDAPServiceEnable) ▪ Port Definition: LDAP Configuration table – LDAP Server Port (LdapConfiguration_LdapConfServerPort) |
| Random | RADIUS client | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: Enable RADIUS Access Control (EnableRADIUS) ▪ Port Definition: Random ▪ Access Control: Firewall and RADIUS Servers table - IP Address (RadiusServers_IPAddress) |
| Random | NTP client | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: NTP Server Address (NTPServerIP) ▪ Port Definition: Random ▪ Access Control: Firewall & NTP Server Address (NTPServerIP) |
| Random | OCSP client | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: Primary Server IP (OCSPServerIP); Secondary Server IP (OCSPSecondaryServerIP) ▪ Port Definition: Random ▪ Access Control: Firewall; Primary Server IP (OCSPServerIP); Secondary Server IP (OCSPSecondaryServerIP) |
| Random | TFTP Note: Applicable only to LAN and Mediant 3000. | Enabled | <ul style="list-style-type: none"> ▪ Enable / Disable: Determined by BootP response. Can be disabled using <i>EnableSecureStartup</i> ▪ Port Definition: Random ▪ Access Control: See Enable / Disable above |
| Random | Syslog | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: Enable Syslog (EnableSyslog) ▪ Port Definition: Syslog Server Port (SyslogServerPort) ▪ Access Control: N/A |
| Random | DNS client | Disabled | <ul style="list-style-type: none"> ▪ Enable / Disable: <ul style="list-style-type: none"> ✓ LAN: Interface table - Primary DNS (InterfaceTable_PrimaryDNSServerIPAddress) ✓ WAN: Web data-routing for defining WAN (Applicable only to MSBR) ▪ Port Definition: N/A ▪ Access Control: N/A |

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