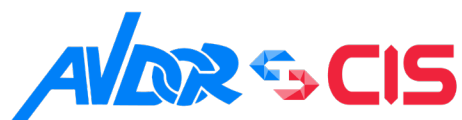


## **Avdor CIS Crystal Quality® SIPREC with AudioCodes Mediant™ SBC**

Version 7.2





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## Abbreviations and Terminology

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

## Document Revision Record

LTRT	Description
39328	Initial document release for Version 7.2.
39329	Renaming company name and changing logo.

## Documentation Feedback

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# 1 Introduction

This Configuration Note describes how to set up the AudioCodes Enterprise Session Border Controller (hereafter, referred to as *SBC*) for interworking with Avdor CIS Crystal Quality SIPREC recording solutions.

## 1.1 Intended Audience

This document is intended for engineers, or AudioCodes and Avdor CIS Crystal Quality partners who are responsible for installing and configuring AudioCodes SBC for enabling SIPREC streaming to the Avdor CIS Crystal Quality recording system.

## 1.2 About AudioCodes SBC Product Series

AudioCodes' family of SBC devices enables reliable connectivity and security between the Enterprise's and the service provider's VoIP networks.

The SBC provides perimeter defense as a way of protecting Enterprises from malicious VoIP attacks; mediation for allowing the connection of any PBX and/or IP-PBX to any service provider; and Service Assurance for service quality and manageability.

Designed as a cost-effective appliance, the SBC is based on field-proven VoIP and network services with a native host processor, allowing the creation of purpose-built multiservice appliances, providing smooth connectivity to cloud services, with integrated quality of service, SLA monitoring, security and manageability. The native implementation of SBC provides a host of additional capabilities that are not possible with standalone SBC appliances such as VoIP mediation, PSTN access survivability, and third-party value-added services applications. This enables Enterprises to utilize the advantages of converged networks and eliminate the need for standalone appliances.

AudioCodes SBC is available as an integrated solution running on top of its field-proven Mediant Media Gateway and Multi-Service Business Router platforms, or as a software-only solution for deployment with third-party hardware. The SBC can be offered as a Virtualized SBC, supporting the following platforms: Hyper-V, AWS, AZURE, AWP, KVM and VMWare.

## 1.3 About Avdor CIS Crystal Quality Recording

Crystal Quality is a unified, web-based multimedia recording solution for both TDM, analog and VoIP environments.

It supports a wide range of telephony interfaces including analog trunk/extensions, digital trunk (ISDN BRI/PRI, E1/T1) and digital extensions, VoIP trunk/ extensions, as well as discrete radios and trunked radios.

(Avdor CIS) Crystal Quality, provides management of all types of recordings (call centers, contact centers, telephony, mobile phones, screen captures, SMS, Skype for Business and more) using a simple, easy to operate and modern-looking user interface. Avdor CIS offers several convenient pricing plans, enabling you to choose the most suitable for your business requirement needs, providing the most benefits and at the most cost-effective price.

Crystal Quality call recording and monitoring features offer a joint solution so that in one place, you can easily find a unique solution for all your business needs. It includes 100% smooth and perfect integration with the leading communication companies, such as Cisco, Avaya, Genesys, Speakerbus, Huawei and many more.

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## 2 Component Information

### 2.1 AudioCodes SBC Version

**Table 2-1: AudioCodes SBC Version**

<b>SBC Vendor</b>	AudioCodes
<b>Models</b>	<ul style="list-style-type: none"> <li>▪ Mediant 500 Gateway &amp; E-SBC</li> <li>▪ Mediant 500L Gateway &amp; E-SBC</li> <li>▪ Mediant 800B Gateway &amp; E-SBC</li> <li>▪ Mediant 800C Gateway &amp; E-SBC</li> <li>▪ Mediant 1000B Gateway &amp; E-SBC</li> <li>▪ Mediant 2600 E-SBC</li> <li>▪ Mediant 4000 SBC</li> <li>▪ Mediant 4000B SBC</li> <li>▪ Mediant 9000 SBC</li> <li>▪ Mediant 9030 SBC</li> <li>▪ Mediant 9080 SBC</li> <li>▪ Mediant Software SBC (VE/SE/CE)</li> </ul>
<b>Software Version</b>	7.20A.260.012 or later
<b>Protocol</b>	SIP/UDP or SIP/TCP (to the Avdor CIS Crystal Quality SIPREC)
<b>Additional Notes</b>	None

### 2.2 Avdor CIS Crystal Quality Version

**Table 2-2: Avdor CIS Crystal Quality Version**

<b>Vendor/Service Provider</b>	Avdor CIS
<b>SSW Model/Service</b>	Crystal Quality
<b>Software Version</b>	6.03.4652.0 or later
<b>Protocol</b>	SIP/UDP or SIP/TCP (SIPREC)
<b>Additional Notes</b>	None

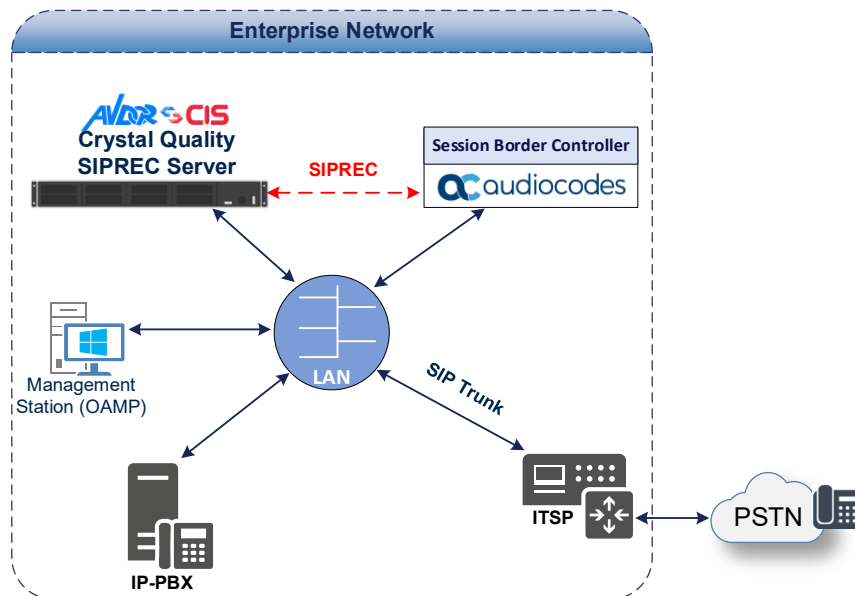
## 2.3 Interoperability Test Topology

The interoperability testing between the AudioCodes SBC and the Avdor CIS Crystal Quality recording system, was done using the following topology setup:

- Enterprise deployed with IP-PBX, the administrator's management station and Avdor CIS Crystal Quality recording system, located on the LAN.
- Enterprise offers its employees enterprise-voice capabilities and Crystal Quality call recording and monitoring connections to the PSTN network using AudioCodes SBC. On the other hand, the enterprise wants to be able to manage call recordings for different business needs.
- AudioCodes SBC is implemented to interconnect between the IP-PBX, SIP Trunk and the Avdor CIS Crystal Quality recording system. Connection to the SIP Trunk can be located on the LAN or WAN. In the current interoperability topology, the SIP Trunk is located on the LAN.

The figure below illustrates this interoperability test topology:

**Figure 2-1: Layout of an Interoperability Test Environment**



### 2.3.1 Environment Setup

The interoperability test topology includes the following environment setup:

**Table 2-3: Environment Setup**

Area	Setup
<b>Network</b>	<ul style="list-style-type: none"><li>All components are located on the Enterprise's LAN</li></ul>
<b>Signaling</b>	<ul style="list-style-type: none"><li>All components are operating with SIP-over-UDP or SIP-over-TCP transport types</li></ul>
<b>Codecs</b>	<ul style="list-style-type: none"><li>IP-PBX and SIP Trunk supports G.711A-law, G.711U-law, and G.729 coders</li><li>Avdor CIS Crystal Quality SIPREC system supports G.711A-law, G.711U-law, G.722 and G.729 coders</li></ul>
<b>Media Transcoding</b>	<ul style="list-style-type: none"><li>All components are operating with RTP media type</li></ul>

### 2.3.2 Known Limitations

There were no limitations observed in the interoperability tests done for the AudioCodes SBC and the Avdor CIS Crystal Quality recording system.

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## 3 Working with Avdor CIS Crystal Quality SIPREC Recording System

This section describes an example of the Avdor CIS Crystal Quality SIPREC Recording System settings and monitoring tools.

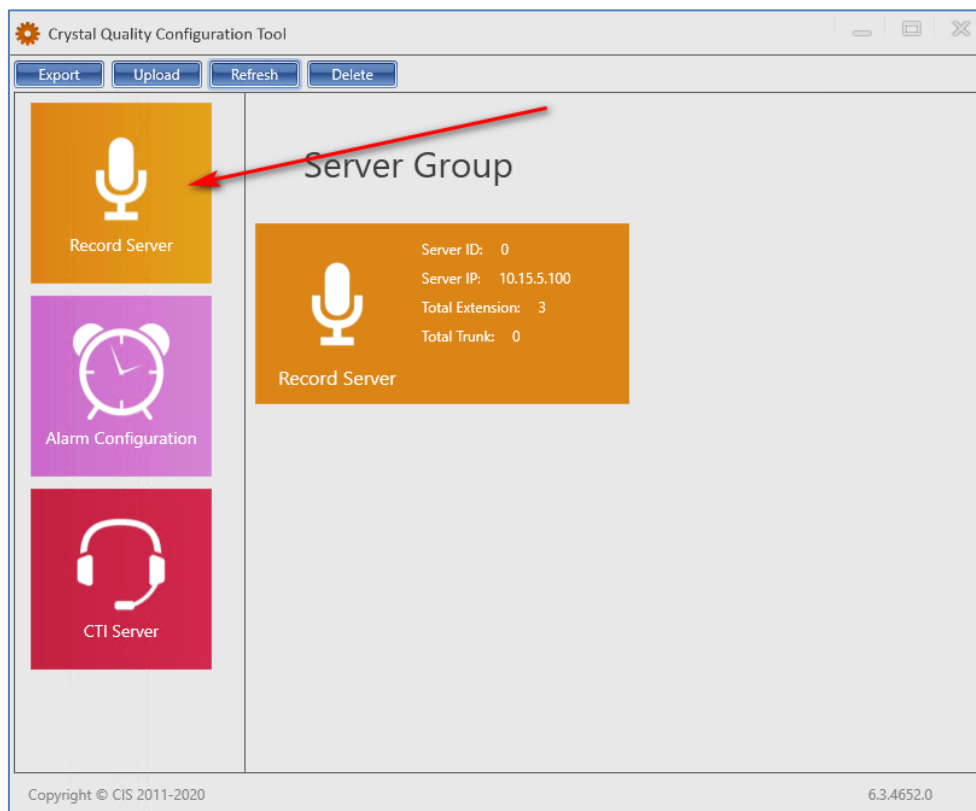
### 3.1 Configuring the Record Server

This section describes the basic parameters needed to configure the Avdor CIS Crystal Quality Record Server.

➤ **To configure the Record Server:**

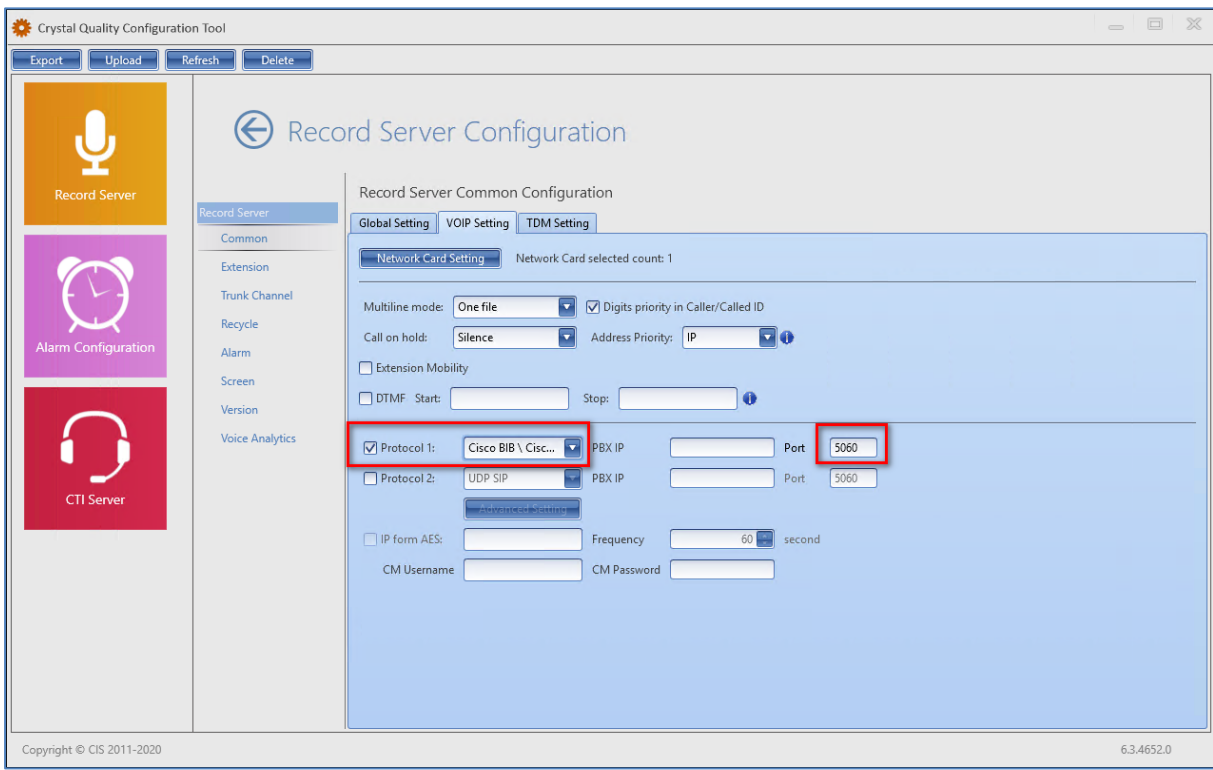
1. After installing all the required components, open **Crystal Quality**. Select **Configuration Tool**. (After a standard installation, there is a shortcut on the desktop). Log in with the Admin user.
2. Click **Record Server**:

**Figure 3-1: Avdor CIS Crystal Quality Record Server**



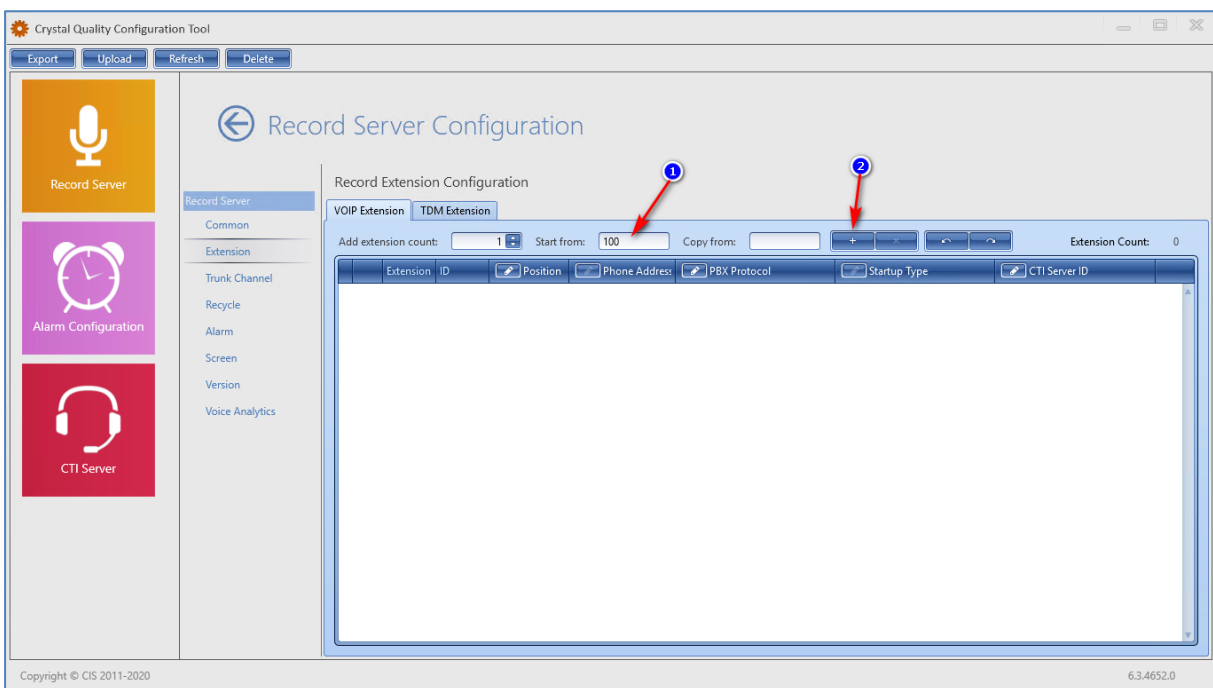
3. In the **VOIP Settings** section go to the Common Configuration page.
4. From the 'Protocol1' drop-down list, select **Cisco BIB \ Cisco SIP Fork \ SIP Rec** and the required port (usually **5060**).

**Figure 3-2: Record Server Common Configuration**



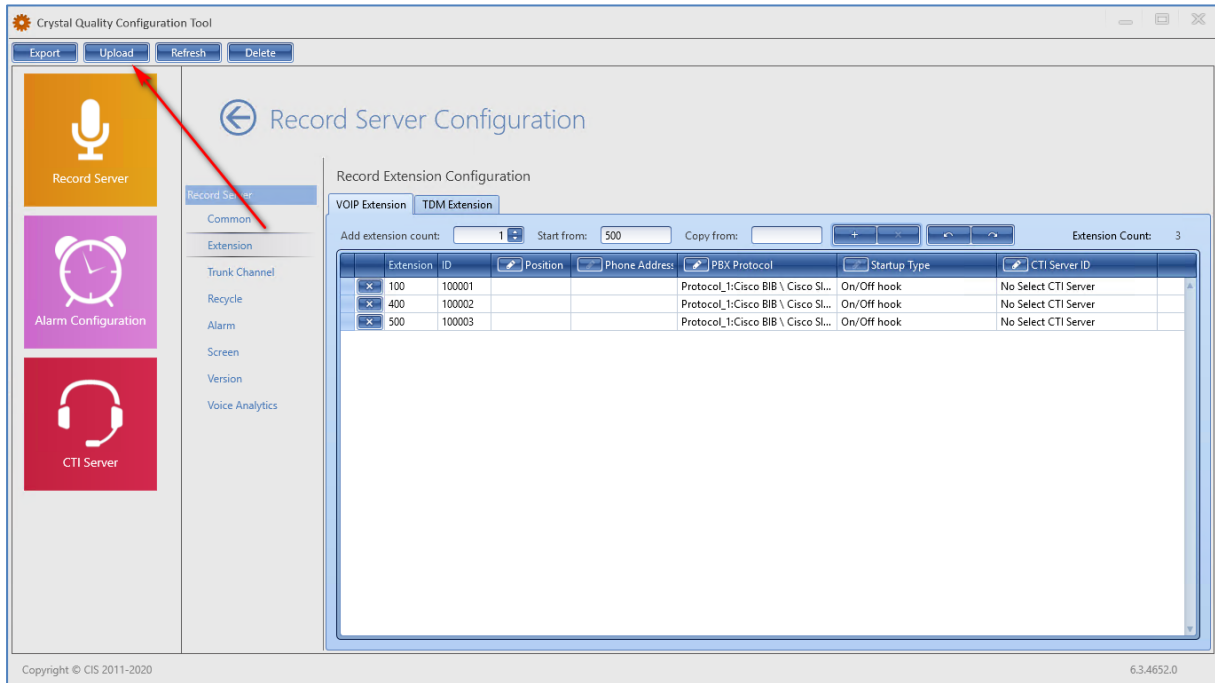
5. Go to the Extension Configuration page and added extensions for monitoring:

**Figure 3-3: Record Extension Configuration**



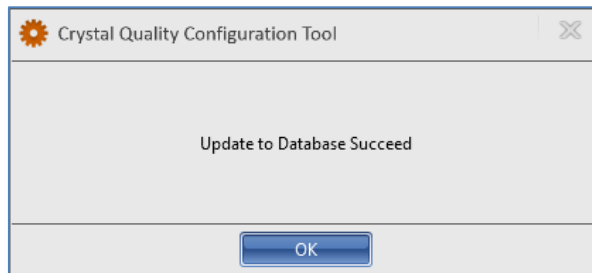
- After adding all required extensions, upload the configuration to the server.

**Figure 3-4: Record Extension Configuration**



- Confirm that the database was successfully updated.

**Figure 3-5: Record Extension Configuration**



**Note:** For better performance, it is recommended to restart *CrystalLive.RecordService* through Windows Services, after performing configuration changes.

## 3.2 Basic Monitoring

This section describes the basic monitoring options, provided by Avdor CIS Crystal Quality Record Server.

1. To enter to the web monitor, open the web browser and enter the localhost (if you working locally on the server) or the IP address of the server in the address bar. (After performing a standard installation, a shortcut appears on the desktop).
2. Log in as the Admin user.

**Figure 3-6: Crystal Quality Web Login**

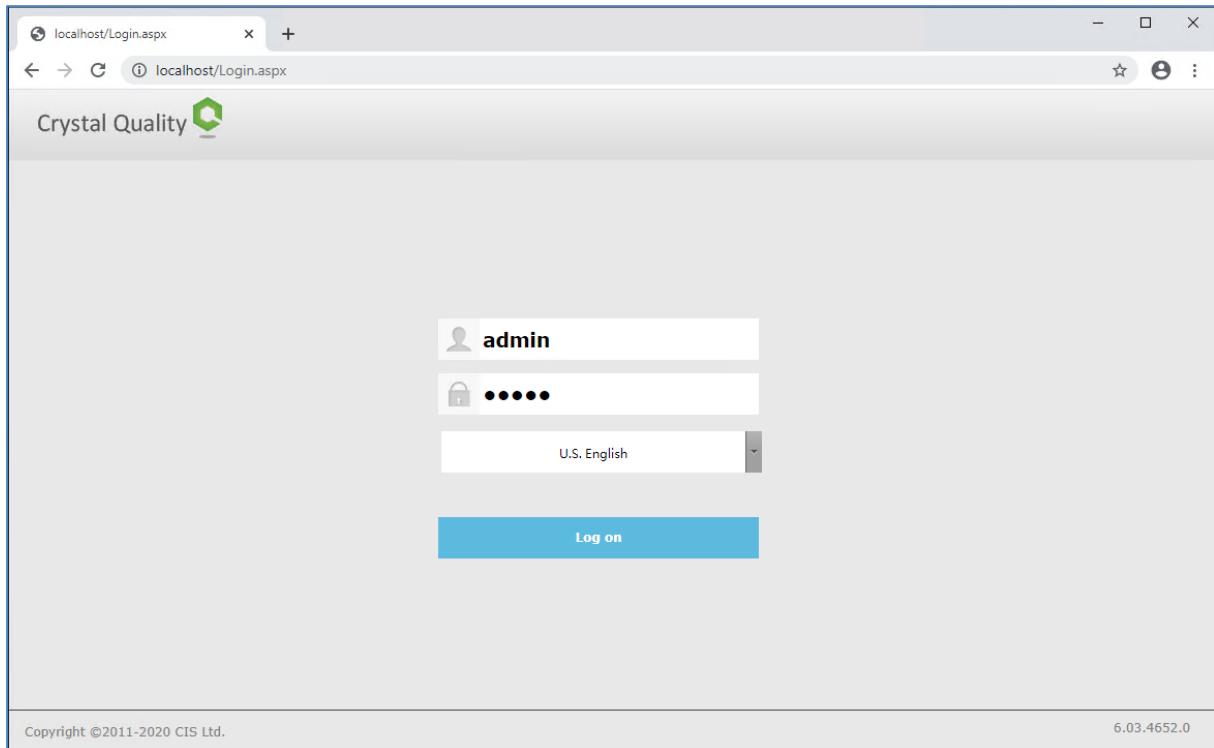
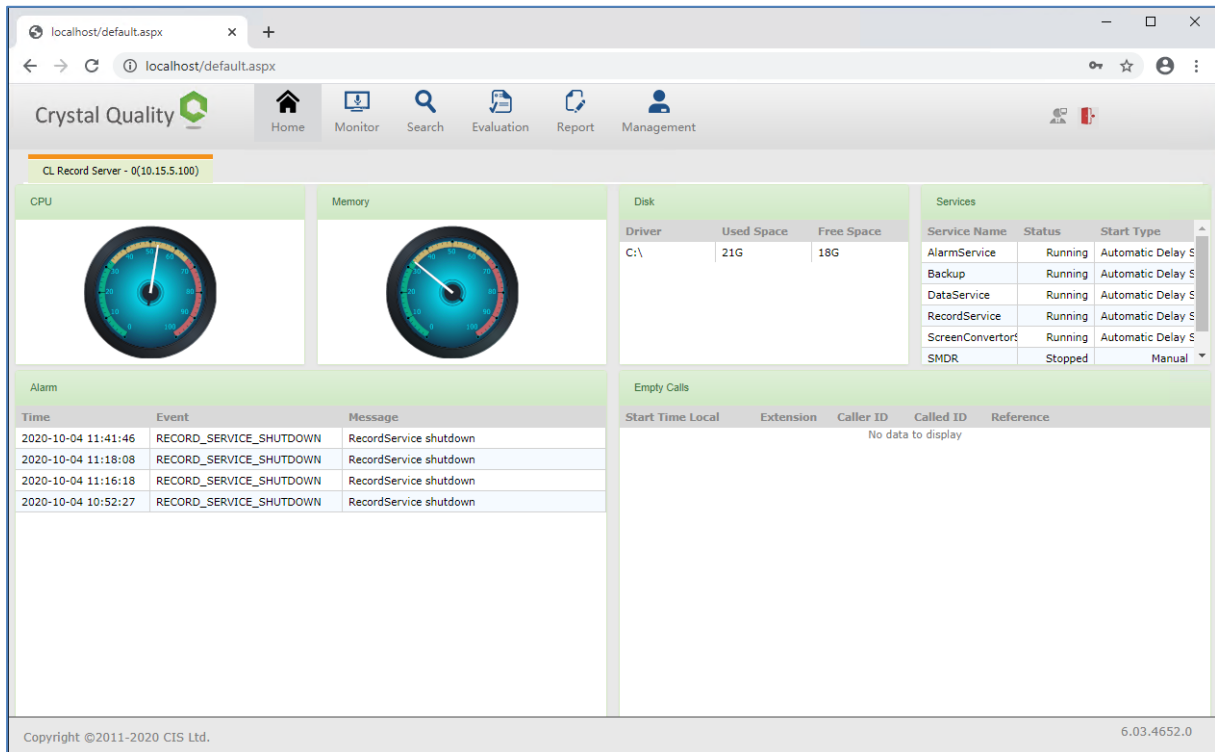


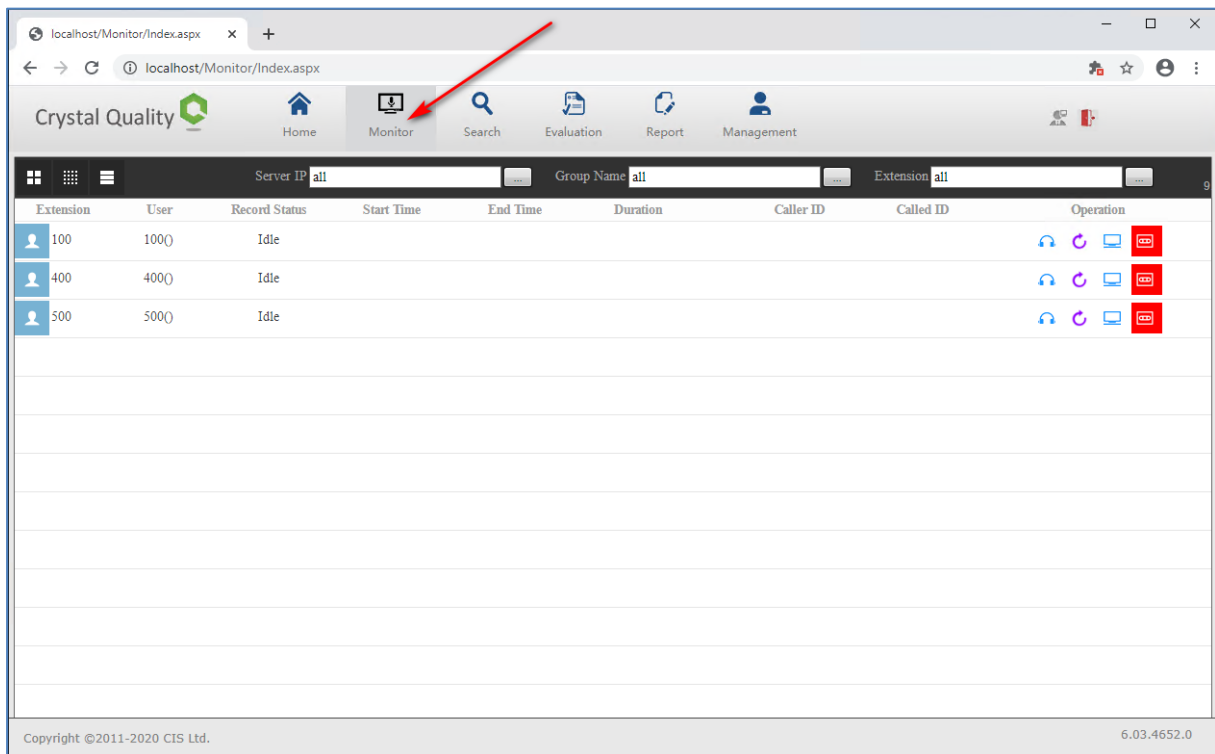


Figure 3-7: Crystal Quality Web Dashboard



3. For real-time monitoring, go to the **Monitor** page:

Figure 3-8: Crystal Quality Real-Time Monitor



- For historical data use the **Search** page.

**Figure 3-9: Crystal Quality Web Dashboard**

#	SDK	Dir.	Duration	Local Start Time	Caller ID	Called ID	Extension	Score	Reference
1			00:00:18	2020-10-04 15:12:05	100	500	500		001000030
2			00:00:44	2020-10-04 15:11:40	400	100	100		001000010
3			00:00:25	2020-10-04 15:11:40	400	400	400		001000020
4			00:00:10	2020-10-04 15:09:38	100	500	500		001000030
5			00:00:19	2020-10-04 15:09:26	400	100	100		001000010
6			00:00:08	2020-10-04 15:09:26	400	400	400		001000020
7			00:00:46	2020-10-04 12:22:52	500	100	100		001000010
8			00:00:30	2020-10-04 12:19:51	400	100	100		001000010
9			00:00:17	2020-10-04 12:17:01	500	100	100		001000010
10	✓		00:00:14	2020-10-04 12:12:27	400	100	100		001000010

- Use different filters to find the required records.
- Records can be listened to, saved, copied or removed.

## 4 Configuring AudioCodes SBC

This section provides step-by-step procedures examples on how to configure AudioCodes SBC for enabling SIPREC streaming to the Avdor CIS Crystal Quality recording system. These configuration procedures are based on the interoperability test topology described in Section 2.3 on page 10, and includes the following main areas:

- **SBC LAN Interface:** Defines the Management Station, IP-PBX SIP Trunk and Avdor CIS Crystal Quality SIPREC recording system environment.

This configuration is done using the SBC's embedded web server (hereafter, referred to as the *Web interface*).

### Notes:

- For implementing the SIPREC streaming to the Avdor CIS Crystal Quality recording system, based on the configuration described in this section, AudioCodes SBC must be installed with a License Key that includes the following software features:
    - **SIPREC Sessions** [Based on requirements]
    - **Number of SBC sessions** [Based on requirements]
    - **DSP Channels** [If media transcoding is needed]
    - **Transcoding sessions** [If media transcoding is needed]
- For more information about the License Key, contact your AudioCodes sales representative.
- The scope of this document does **not** cover all security aspects for configuring this topology. Comprehensive security measures should be implemented per your organization's security policies. For security recommendations on AudioCodes' products, refer to the *Recommended Security Guidelines* document, which can be found in the AudioCodes web site.

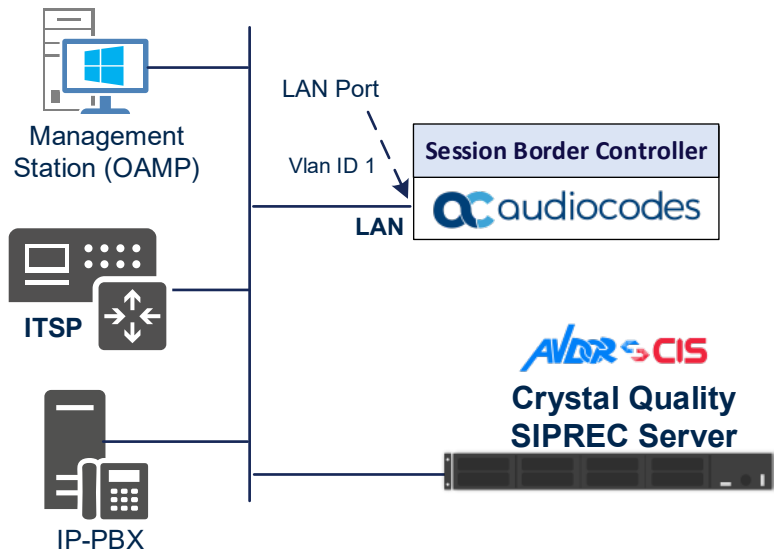


## 4.1 IP Network Interfaces Configuration

This section describes how to configure the SBC's IP network interfaces. There are several ways to deploy the SBC; however, this interoperability test topology employs the following deployment method:

- SBC interfaces with the following IP entities:
  - Management Servers, IP-PBX, SIP Trunk and Avdor CIS Crystal Quality recording system, located on the LAN
- Physical connection: The type of physical connection depends on the method used to connect to the Enterprise's network. In the interoperability test topology, SBC connects to the LAN using dedicated Ethernet port.

**Figure 4-1: Network Interfaces in Interoperability Test Topology**



### 4.1.1 Configure VLANs

This section describes how to configure VLANs for each of the following interfaces:

- LAN (assigned the name "LAN\_IF")
- WAN (assigned the name "WAN\_IF")

➤ **To configure the VLANs:**

1. Open the Ethernet Device table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **Ethernet Devices**).
2. There will be one existing row for VLAN ID 1 and underlying interface GROUP\_1.
3. Add another VLAN ID 2 for the WAN side.

**Figure 4-2: Configured VLAN IDs in Ethernet Device**

INDEX	VLAN ID	UNDERLYING INTERFACE	NAME	TAGGING
0	1	GROUP_1	vlan 1	Untagged
1	2	GROUP_2	vlan 2	Untagged

### 4.1.2 Configure Network Interfaces

This section describes how to configure the IP network interfaces for each of the following interfaces:

- LAN Interface (assigned the name "LAN\_IF")
- WAN Interface (assigned the name "WAN\_IF")

➤ **To configure the IP network interfaces:**

1. Open the IP Interfaces table (**Setup** menu > **IP Network** tab > **Core Entities** folder > **IP Interfaces**).
2. Configure the IP interfaces as follows (your network parameters might be different):

**Table 4-1: Configuration Example of the Network Interface Table**

Index	Application Types	Interface Mode	IP Address	Prefix Length	Gateway	DNS	I/F Name	Ethernet Device
0	OAMP+ Media + Control	IPv4 Manual	10.15.77.55	16	10.15.0.1	10.1.1.6	LAN_IF	vlan 1
1	Media + Control	IPv4 Manual	195.189.192.153 (DMZ IP address of SBC)	25	195.189.192.129 (router's IP address)	8.8.8.8	WAN_IF	vlan 2

The configured IP network interfaces are shown below:

**Figure 4-3: Configured Network Interfaces in IP Interfaces Table**

INDEX	NAME	APPLICATION TYPE	INTERFACE MODE	IP ADDRESS	PREFIX LENGTH	DEFAULT GATEWAY	PRIMARY DNS	SECONDARY DNS	ETHERNET DEVICE
0	LAN_IF	OAMP + Media +	IPv4 Manual	10.15.77.55	16	10.15.0.1	10.1.1.6	10.1.1.10	vlan 1
1	WAN_IF	Media + Control	IPv4 Manual	195.189.192.153	25	195.189.192.129	8.8.8.8	8.8.8.8	vlan 2

## 4.2 Configure Media Realms

This section describes how to configure Media Realms. For specific interworking tests, the default Media Realm configuration was used. No changes were done for the Media Realm configuration.



**Note:** This document describes the configuration which was used for interoperability tests. Your implementation can be different. So, for a detailed configuration of the Media Realms, refer to the device's User's Manual.

➤ **To configure Media Realms:**

1. Open the Media Realms table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Media Realms**).
2. Configure Media Realms according to your deployment.

**Figure 4-4: Default Media Realm in Media Realm Table**

INDEX	NAME	IPV4 INTERFACE NAME	UDP PORT RANGE START	NUMBER OF MEDIA SESSION LEGS	UDP PORT RANGE END	DEFAULT MEDIA REALM
0	DefaultRealm	LAN_IF	6000	5953	65529	Yes

### 4.3 Configure SIP Signaling Interfaces

This section describes how to configure SIP Interfaces. For specific interworking tests, the default SIP Interface configuration was used. No changes were made for the SIP Interface configuration.



**Note:** This document describes the configuration, which was used for interoperability tests. Your implementation can be different. So, for a detailed configuration of the SIP Interface, refer to the device's User's Manual.

➤ **To configure SIP Interface:**

1. Open the SIP Interfaces table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **SIP Interfaces**).
2. Configure a SIP Interface according to your deployment.

**Figure 4-5: Default SIP Interface in SIP Interface Table**

INDEX	NAME	SRD	NETWORK INTERFACE	APPLICATION TYPE	UDP PORT	TCP PORT	TLS PORT	ENCAPSULATION PROTOCOL	MEDIA REALM
0	SIPInterface_0	DefaultSRD	LAN_IF	SBC	5060	5060	5061	No encapsulat	--

## 4.4 Configure Proxy Sets and Proxy Address

This section describes how to configure Proxy Sets. The Proxy Set defines the destination address (IP address or FQDN) of the IP entity server. Proxy Sets can also be used to configure load balancing between multiple servers.

For the interoperability test topology, two Proxy Sets need to be configured for the following IP entities:

- IP-PBX
- SIP Trunk
- Avdor CIS Crystal Quality recording system

The Proxy Sets will later be applied to the VoIP network by assigning them to IP Groups.

➤ **To configure Proxy Sets:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**).
2. Configure Proxy Sets as shown in the table below:

**Table 4-2: Configuration Example Proxy Sets in Proxy Sets Table**

Index	Name	SBC IPv4 SIP Interface	Proxy Keep-Alive
1	<b>IP-PBX</b> (arbitrary name)	SIPInterface_0	Using Options
2	<b>SIPTrunk</b> (arbitrary name)	SIPInterface_0	Using Options
3	<b>CQ-SIPREC</b> (arbitrary name)	SIPInterface_0	Using Options

The configured Proxy Sets are shown in the figure below:

**Figure 4-6: Configured Proxy Sets in Proxy Sets Table**

INDEX	NAME	SRD	GATEWAY IPv4 SIP INTERFACE	SBC IPv4 SIP INTERFACE	PROXY KEEP-ALIVE TIME [SEC]	REDUNDANCY MODE	PROXY HOT SWAP
0	ProxySet_0	DefaultSRD (#0)	--	SIPInterface_0	60		Disable
1	IP-PBX	DefaultSRD (#0)	--	SIPInterface_0	60		Disable
2	SIP Trunk	DefaultSRD (#0)	--	SIPInterface_0	60		Disable
3	CQ-SIPREC	DefaultSRD (#0)	--	SIPInterface_0	60		Disable



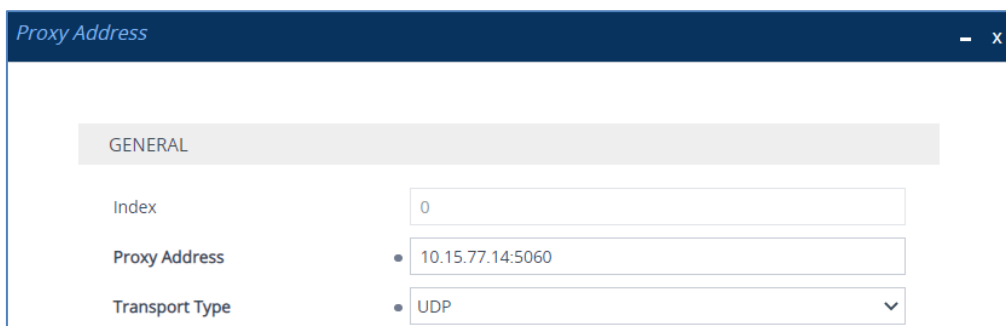
### 4.4.1 Configure a Proxy Address

This section shows how to configure a Proxy Address.

➤ **To configure a Proxy Address for IP-PBX:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) and then click the Proxy Set **IP-PBX**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

**Figure 4-7: Configuring Proxy Address for IP-PBX**



3. Configure the address of the Proxy Set according to the parameters described in the table below:

**Table 4-3: Configuration Proxy Address for IP-PBX**

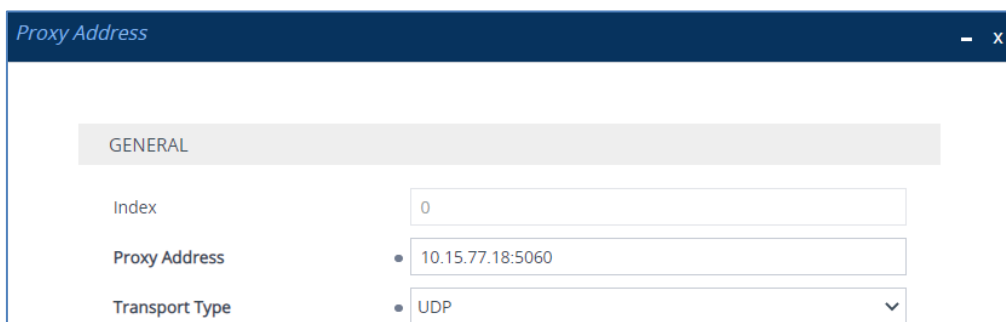
Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	<b>10.15.77.14:5060</b> (IP-PBX IP and port)	UDP	0	0

4. Click **Apply**.

➤ **To configure a Proxy Address for SIP Trunk:**

1. Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) and then click the Proxy Set **SIP Trunk**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
2. Click **+New**; the following dialog box appears:

**Figure 4-8: Configuring Proxy Address for SIP Trunk**



- Configure the address of the Proxy Set according to the parameters described in the table below:

**Table 4-4: Configuration Proxy Address for SIP Trunk**

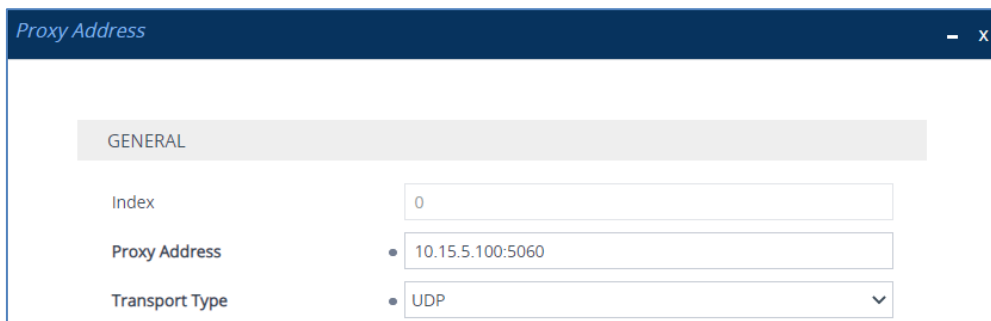
Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	<b>10.15.77.18:5060</b> (SIP Trunk IP and port)	UDP	0	0

- Click **Apply**.

➤ **To configure a Proxy Address for the AWS Chime Voice Connector:**

- Open the Proxy Sets table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **Proxy Sets**) and then click the Proxy Set **CQ-SIPREC**, and then click the **Proxy Address** link located below the table; the Proxy Address table opens.
- Click **+New**; the following dialog box appears:

**Figure 4-9: Configuring Proxy Address for Avdor CIS Crystal Quality Recording System**



- Configure the address of the Proxy Set according to the parameters described in the table below:

**Table 4-5: Configuration Proxy Address for Avdor CIS Crystal Quality Recording System**

Index	Proxy Address	Transport Type	Proxy Priority	Proxy Random Weight
0	<b>10.15.5.100:5060</b> (CQ-SIPREC IP and port)	UDP	0	0

- Click **Apply**.

## 4.5 Configure IP Profiles

This section describes how to configure IP Profiles. The IP Profile defines a set of call capabilities relating to signaling (e.g., SIP message terminations such as REFER) and media (e.g., coder and transcoding method).

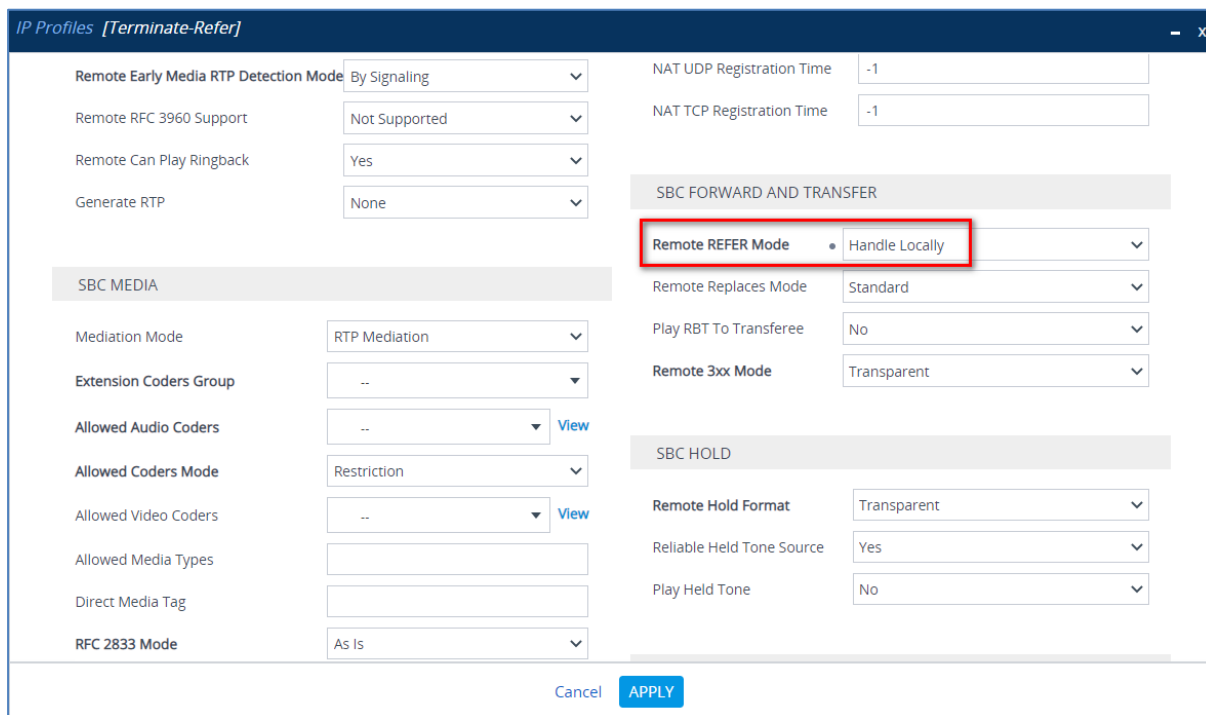
In this interoperability test topology, only one IP Profile need to be configured for terminating REFER messages on the SBC.

➤ **To configure IP Profile for the IP-PBX:**

1. Open the IP Profiles table (**Setup** menu > **Signaling & Media** tab > **Coders & Profiles** folder > **IP Profiles**).
2. Click **New**, and then configure the parameters as follows:

Parameter	Value
<b>General</b>	
Index	1
Name	Terminate-Refer (arbitrary descriptive name)
<b>SBC Forward and Transfer</b>	
Remote REFER Mode	Handle Locally

**Figure 4-10: Configuring IP Profile for Terminate Refer**



3. Click **Apply**.

## 4.6 Configure IP Groups

This section describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the SBC communicates. This can be a server (e.g., IP-PBX or ITSP) or it can be a group of users (e.g., LAN IP phones). For servers, the IP Group is typically used to define the server's IP address by associating it with a Proxy Set. Once IP Groups are configured, they are used to configure IP-to-IP routing rules for denoting source and destination of the call.

In this interoperability test topology, IP Groups must be configured for the following IP entities:

- IP-PBX
- SIP Trunk
- Avdor CIS Crystal Quality recording system

➤ **To configure IP Groups:**

1. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
2. Configure IP Groups as follows:

**Table 4-6: Configuration Example of the IP Groups Table**

Index	Name	Type	Proxy Set	IP Profile	Media Realm
1	IP-PBX	Server	IP-PBX	Terminate-Refer	DefaultRealm
2	SIP Trunk	Server	SIP Trunk	Terminate-Refer	DefaultRealm
3	CQ-SIPREC	Server	CQ-SIPREC	-	DefaultRealm

The configured IP Groups are shown in the figure below:

**Figure 4-11: Configured IP Groups in IP Group Table**

INDEX	NAME	SRD	TYPE	SBC OPERATION MODE	PROXY SET	IP PROFILE	MEDIA REALM	SIP GROUP NAME	CLASSIFY BY PROXY SET	INBOUND MESSAGE MANIPULAT SET	OUTBOUND MESSAGE MANIPULATI SET
0	Default_IPG	DefaultSF	Server	Not Configu	ProxySet_0	--	DefaultRealr		Disable	-1	-1
1	IP-PBX	DefaultSF	Server	Not Configu	IP-PBX	Terminate-R	DefaultRealr		Enable	-1	-1
2	SIP Trunk	DefaultSF	Server	Not Configu	SIP Trunk	Terminate-R	DefaultRealr		Enable	-1	-1
3	CQ-SIPREC	DefaultSF	Server	Not Configu	CQ-SIPREC	--	DefaultRealr		Enable	-1	1

## 4.7 Configure IP-to-IP Call Routing Rules

This section describes how to configure IP-to-IP call routing rules. These rules define the routes for forwarding SIP messages (e.g., INVITE) received from one IP entity to another. The SBC selects the rule whose configured input characteristics (e.g., IP Group) match those of the incoming SIP message. If the input characteristics do not match the first rule in the table, they are compared to the second rule, and so on, until a matching rule is located. If no rule is matched, the message is rejected.

For the interoperability test topology, the following IP-to-IP routing rules need to be configured:

- Terminate SIP OPTIONS messages on the SBC that are received from any entity
- Calls from IP-PBX to SIP Trunk
- Calls from SIP Trunk to IP-PBX

➤ **To configure IP-to-IP routing rules:**

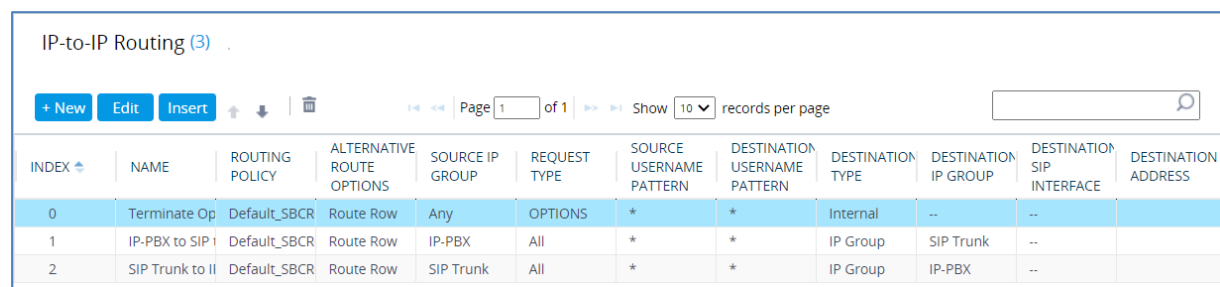
1. Open the IP-to-IP Routing table (**Setup** menu > **Signaling & Media** tab > **SBC** folder > **Routing** > **IP-to-IP Routing**).
2. Configure routing rules as shown in the table below:

**Table 4-7: Configuration IP-to-IP Routing Rules**

Index	Name	Source IP Group	Request Type	Dest Type	Dest IP Group	Internal Action
0	Terminate OPTIONS	Any	OPTIONS	Internal		Reply(Response='200')
1	IP-PBX to SIP Trunk (arbitrary name)	IP-PBX		IP Group	SIP Trunk	
2	SIP Trunk to IP-PBX (arbitrary name)	SIP Trunk		IP Group	IP-PBX	

The configured routing rules are shown in the figure below:

**Figure 4-12: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table**



INDEX	NAME	ROUTING POLICY	ALTERNATIVE ROUTE OPTIONS	SOURCE IP GROUP	REQUEST TYPE	SOURCE USERNAME PATTERN	DESTINATION USERNAME PATTERN	DESTINATION TYPE	DESTINATION IP GROUP	DESTINATION SIP INTERFACE	DESTINATION ADDRESS
0	Terminate Op	Default_SBCR	Route Row	Any	OPTIONS	*	*	Internal	--	--	
1	IP-PBX to SIP	Default_SBCR	Route Row	IP-PBX	All	*	*	IP Group	SIP Trunk	--	
2	SIP Trunk to IP	Default_SBCR	Route Row	SIP Trunk	All	*	*	IP Group	IP-PBX	--	



**Note:** The routing configuration may change according to your specific deployment topology.

## 4.8 Configure SIP Recording

This section describes SBC's SIP Recording configuration.

### 4.8.1 Configure SIP Recording Rules

This section describes how to configure SIP Recording rules through the Web interface. A SIP Recording Rule defines call routes that you want to record.

➤ **To configure a SIP Recording Routing rule:**

1. Open the SIP Recording Rules table (**Setup** menu > **Signaling & Media** tab > **SIP Recording** folder > **SIP Recording Rules**).
2. Click **New** and configure a SIP recording rule according to the table below:

**Table 4-8: SIP Recording Rule**

Index	Recorded IP Group	Peer IP Group	Caller	Recording Server (SRS) IP Group
0	IP-PBX	Any	Both	CQ-SIPREC
1	SIP Trunk	Any	Both	CQ-SIPREC

The configured SIP recording rules are shown in the figure below:

**Figure 4-13: Configured SIP Recording Rules**

INDEX	RECORDED IP GROUP	PEER IP GROUP	PEER TRUNK GROUP ID	CALLER	RECORDING SERVER (SRS) IP GROUP
0	IP-PBX	Any	-1	Both	CQ-SIPREC
1	SIP Trunk	Any	-1	Both	CQ-SIPREC

## 4.9 Configure Number Manipulation Rules (Optional)

IP-to-IP manipulation rules manipulate the SIP Request-URI user part (source or destination number). The manipulation rules use the configured IP Groups (as configured in Section 4.5 on page 26) to denote the source and destination of the call.



**Note:** Configure Number Manipulation Rules only if this is required by the SIP Trunk. For a detailed description, refer to the Configuration Notes document for the specific SIP Trunk.

## 4.10 Configure Message Manipulation Rules

This section describes how to configure SIP message manipulation rules. SIP message manipulation rules can include insertion, removal, and/or modification of SIP headers. Manipulation rules are grouped into Manipulation Sets, enabling you to apply multiple rules to the same SIP message (IP entity).

Once you have configured the SIP message manipulation rules, you need to assign them to the relevant IP Group (in the IP Group table) and determine whether they must be applied to inbound or outbound messages.

➤ **To configure SIP message manipulation rule:**

1. Open the Message Manipulations page (**Setup** menu > **Signaling & Media** tab > **Message Manipulation** folder > **Message Manipulations**).
2. Configure a manipulation rule (Manipulation Set 1) for the Avdor CIS Crystal Quality recording system IP Group. This rule applies to messages sent to the Avdor CIS Crystal Quality recording system. It modifies metadata according to Avdor CIS Crystal Quality requirements – all AOR records should be started with 'sip:' string.

Parameter	Value
Index	0
Name	SIPREC - Fix Metadata
Manipulation Set ID	1
Message Type	Invite.Request
Condition	Body.application/rs-metadata regex (.*)(<nameID aor=")(.*)(@)(.*)(<nameID aor=")(.*)(@)(.*)
Action Subject	Body.application/rs-metadata
Action Type	Modify
Action Value	\$1+\$2+'sip:' +\$3+\$4+\$5+\$6+'sip:'+\$7+\$8+\$9

Figure 4-14: Configuring SIP Message Manipulation Rule 0 (for CQ-SIPREC)

The screenshot shows the configuration interface for a SIP message manipulation rule. It is titled "Message Manipulations [SIPREC - Fix Metadata]".

- GENERAL**
  - Index: 0
  - Name: SIPREC - Fix Metadata
  - Manipulation Set ID: 1
  - Row Role: Use Current Condition
- ACTION**
  - Action Subject: Body.application/rs-metadata
  - Action Type: Modify
  - Action Value: \$1+\$2+'sip:' +\$3+\$4+\$5+\$6+'sip:'+\$7+\$8+\$9
- MATCH**
  - Message Type: Invite.Request
  - Condition: Body.application/rs-metadata regex (.\*)

Buttons for "Cancel" and "APPLY" are visible at the bottom.

3. Assign Manipulation Set ID 1 to the CQ-SIPREC IP Group:
  - a. Open the IP Groups table (**Setup** menu > **Signaling & Media** tab > **Core Entities** folder > **IP Groups**).
  - b. Select the row of the CQ-SIPREC IP Group, and then click **Edit**.
  - c. Set the 'Outbound Message Manipulation Set' field to 1.

**Figure 4-15: Assigning Manipulation Set 1 to the CQ-SIPREC IP Group**

- d. Click **Apply**, and then save your settings to flash memory.



**Note:** Configure additional Message Manipulation Rules only if this is required by the SIP Trunk. For a detailed description, refer to the Configuration Notes document for the specific SIP Trunk.



## A AudioCodes INI File

The *ini* configuration file of the SBC, corresponding to the Web-based configuration as described in Section 4 on page 19, is shown below:



**Note:** To load or save an *ini* file, use the Configuration File page (**Setup** menu > **Administration** tab > **Maintenance** folder > **Configuration File**).

```
;*****
;** Ini File **
;*****

[SYSTEM Params]

SyslogServerIP = 10.15.77.100
EnableSyslog = 1
NTPServerUTCOffset = 7200
ENABLEPARAMETERSMONITORING = 1
ActivityListToLog = 'pvc', 'afl', 'dr', 'fb', 'swu', 'naa', 'spc', 'll',
'cli', 'ae'
NTPServerIP = '0.0.0.0'
SBCWizardFilename = 'templates4.zip'

[BSP Params]

PCMLawSelect = 3
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

[ControlProtocols Params]

AdminStateLockControl = 0

[Voice Engine Params]

PLThresholdLevelsPerMille_0 = 5
PLThresholdLevelsPerMille_1 = 10
PLThresholdLevelsPerMille_2 = 20
PLThresholdLevelsPerMille_3 = 50
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

Languages = 'en-US', '', '', '', '', '', '', '', ''

[SIP Params]

GWDEBUGLEVEL = 5
```

```

MSLDAPPRIMARYKEY = 'telephoneNumber'
ANSWERDETECTORCMD = 10486144

[ DeviceTable ]

FORMAT Index = VlanID, UnderlyingInterface, DeviceName, Tagging, MTU;
DeviceTable 0 = 1, "GROUP_1", "vlan 1", 0, 1500;

[ \DeviceTable ]

[ InterfaceTable ]

FORMAT Index = ApplicationTypes, InterfaceMode, IPAddress, PrefixLength,
Gateway, InterfaceName, PrimaryDNSServerIPAddress,
SecondaryDNSServerIPAddress, UnderlyingDevice;
InterfaceTable 0 = 6, 10, 10.15.77.55, 16, 10.15.0.1, "LAN_IF", 10.1.1.6,
10.1.1.10, "vlan 1";

[ \InterfaceTable ]

[ TLSContexts ]

FORMAT Index = Name, TLSVersion, DTLSVersion, ServerCipherString,
ClientCipherString, ServerCipherTLS13String, ClientCipherTLS13String,
KeyExchangeGroups, RequireStrictCert, TlsRenegotiation,
MiddleboxCompatMode, OcsEnable, OcsServerPrimary, OcsServerSecondary,
OcsServerPort, OcsDefaultResponse, DHKeySize;
TLSContexts 0 = "default", 4, 0, "DEFAULT", "DEFAULT",
"TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1305_SHA256:TLS_AES_128_GCM_SHA2
56",
"TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1305_SHA256:TLS_AES_128_GCM_SHA2
56", "X25519:P-256:P-384:X448", 0, 1, 0, 0, 0.0.0.0, 0.0.0.0, 2560, 0,
2048;

[ \TLSContexts ]

[ AudioCodersGroups ]

FORMAT Index = Name;
AudioCodersGroups 0 = "AudioCodersGroups_0";

[ \AudioCodersGroups ]

[ IpProfile ]

FORMAT Index = ProfileName, IpPreference, CodersGroupName, IsFaxUsed,
JitterBufMinDelay, JitterBufOptFactor, IPDiffServ, SigIPDiffServ,
RTPRedundancyDepth, CNGmode, VxxTransportType, NSEMode, IsDTMFUsed,
PlayRBTone2IP, EnableEarlyMedia, ProgressIndicator2IP,
EnableEchoCanceller, CopyDest2RedirectNumber, MediaSecurityBehaviour,
CallLimit, DisconnectOnBrokenConnection, FirstTxDTMFOption,
SecondTxDTMFOption, RxDTMFOption, EnableHold, InputGain, VoiceVolume,
AddIEInSetup, SBCExtensionCodersGroupName, MediaIPVersionPreference,
TranscodingMode, SBCAllowedMediaTypes, SBCAllowedAudioCodersGroupName,
SBCAllowedVideoCodersGroupName, SBCAllowedCodersMode,
SBCMediaSecurityBehaviour, SBCRFC2833Behavior, SBCAlternativeDTMFMethod,
    
```

```

SBCSendMultipleDTMFMethods, SBCAssertIdentity,
AMDSensitivityParameterSuit, AMDSensitivityLevel, AMDMaxGreetingTime,
AMDMaxPostSilenceGreetingTime, SBCDiversioMode, SBCHistoryInfoMode,
EnableQSIGTunneling, SBCFaxCodersGroupName, SBCFaxBehavior,
SBCFaxOfferMode, SBCFaxAnswerMode, SbcPrackMode, SBCSessionExpiresMode,
SBCRemoteUpdateSupport, SBCRemoteReinviteSupport,
SBCRemoteDelayedOfferSupport, SBCRemoteReferBehavior,
SBCRemote3xxBehavior, SBCRemoteMultiple18xSupport,
SBCRemoteEarlyMediaResponseType, SBCRemoteEarlyMediaSupport,
EnableSymmetricMKI, MKISize, SBCEnforceMKISize, SBCRemoteEarlyMediaRTP,
SBCRemoteSupportsRFC3960, SBCRemoteCanPlayRingback, EnableEarly183,
EarlyAnswerTimeout, SBC2833DTMFPayloadType, SBCUserRegistrationTime,
ResetSRTPStateUponRekey, AmdMode, SBCReliableHeldToneSource,
GenerateSRTPKeys, SBCPlayHeldTone, SBCRemoteHoldFormat,
SBCRemoteReplacesBehavior, SBCSDPptimeAnswer, SBCPreferredPTime,
SBCUseSilenceSupp, SBCRTPRedundancyBehavior, SBCPlayRBTToTransferee,
SBCRTPMode, SBCJitterCompensation, SBCRemoteRenegotiateOnFaxDetection,
JitterBufMaxDelay, SBCUserBehindUdpNATRegistrationTime,
SBCUserBehindTcpNATRegistrationTime, SBCSDPHandlerTCPAttribute,
SBCRemoveCryptoLifetimeInSDP, SBCIceMode, SBCRTCPMux,
SBCMediaSecurityMethod, SBCHandleXDetect, SBCRTCPFeedback,
SBCRemoteRepresentationMode, SBCKeepVIAHeaders, SBCKeepRoutingHeaders,
SBCKeepUserAgentHeader, SBCRemoteMultipleEarlyDialogs,
SBCRemoteMultipleAnswersMode, SBCDirectMediaTag,
SBCAdaptRFC2833BWToVoiceCoderBW, CreatedByRoutingServer,
SBCFaxReroutingMode, SBCMaxCallDuration, SBCGenerateRTP,
SBCISUPBodyHandling, SBCISUPVariant, SBCVoiceQualityEnhancement,
SBCMaxOpusBW, SBCEnhancedPlc, LocalRingbackTone, LocalHeldTone,
SBCGenerateNoOp, SBCRemoveUnknownCrypto, SBCMultipleCoders, DataDiffServ,
SBCMSRPreinviteUpdateSupport, SBCMSRPOfferSetupRole, SBCMSRPEmpMsg;
IpProfile 1 = "Terminate-Refer", 1, "AudioCodersGroups_0", 0, 10, 10, 46,
24, 0, 0, 2, 0, 0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", "",
0, 0, "", "", "", 0, 0, 0, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, "", 0, 0, 1,
3, 0, 2, 2, 1, 3, 0, 1, 0, 1, 0, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 1, 0,
0, 0, 0, 0, 0, 0, 0, 0, 0, 0, 300, -1, -1, 0, 0, 0, 0, 0, 0, -1, -
1, -1, -1, -1, 0, "", 0, 0, 0, 0, 0, 0, 0, 0, 0, 0, -1, -1, 0, 0, 0, 0,
1, 2, 0;

[ \IpProfile ]

[ CpMediaRealm ]

FORMAT Index = MediaRealmName, IPv4IF, IPv6IF, RemoteIPv4IF,
RemoteIPv6IF, PortRangeStart, MediaSessionLeg, PortRangeEnd,
TCPPortRangeStart, TCPPortRangeEnd, IsDefault, QoeProfile, BWProfile,
TopologyLocation;
CpMediaRealm 0 = "DefaultRealm", "LAN_IF", "", "", "", 6000, 5953, 65529,
0, 0, 1, "", "", 0;

[ \CpMediaRealm ]

[ SBCRoutingPolicy ]

FORMAT Index = Name, LCREnable, LCRAverageCallLength, LCRDefaultCost,
LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";

[ \SBCRoutingPolicy ]

[ SRD ]

```

```

FORMAT Index = Name, BlockUnRegUsers, MaxNumOfRegUsers,
EnableUnAuthenticatedRegistrations, SharingPolicy, UsedByRoutingServer,
SBCOperationMode, SBCRoutingPolicyName, SBCDialPlanName,
AdmissionProfile;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, "Default_SBCRoutingPolicy", "",
"";

[ \SRD ]

[ MessagePolicy ]

FORMAT Index = Name, MaxMessageLength, MaxHeaderLength, MaxBodyLength,
MaxNumHeaders, MaxNumBodies, SendRejection, MethodList, MethodListType,
BodyList, BodyListType, UseMaliciousSignatureDB;
MessagePolicy 0 = "Malicious Signature DB Protection", -1, -1, -1, -1, -
1, 1, "", 0, "", 0, 1;

[ \MessagePolicy ]

[ SIPInterface ]

FORMAT Index = InterfaceName, NetworkInterface,
SCTPSecondaryNetworkInterface, ApplicationType, UDPPort, TCPPort,
TLSPort, SCTPPort, AdditionalUDPPorts, AdditionalUDPPortsMode, SRDName,
MessagePolicyName, TLSContext, TLSMutualAuthentication,
TCPKeepaliveEnable, ClassificationFailureResponseType,
PreClassificationManSet, EncapsulatingProtocol, MediaRealm,
SBCDirectMedia, BlockUnRegUsers, MaxNumOfRegUsers,
EnableUnAuthenticatedRegistrations, UsedByRoutingServer,
TopologyLocation, PreParsingManSetName, AdmissionProfile,
CallSetupRulesSetId;
SIPInterface 0 = "SIPInterface_0", "LAN_IF", "", 2, 5060, 5060, 5061, 0,
"", 0, "DefaultSRD", "", "default", -1, 0, 500, -1, 0, "", 0, -1, -1, -1,
0, 0, "", "", -1;

[ \SIPInterface ]

[ ProxySet ]

FORMAT Index = ProxyName, EnableProxyKeepAlive, ProxyKeepAliveTime,
ProxyLoadBalancingMethod, IsProxyHotSwap, SRDName, ClassificationInput,
TLSContextName, ProxyRedundancyMode, DNSResolveMethod,
KeepAliveFailureResp, GWIPv4SIPInterfaceName, SBCIPv4SIPInterfaceName,
GWIPv6SIPInterfaceName, SBCIPv6SIPInterfaceName, MinActiveServersLB,
SuccessDetectionRetries, SuccessDetectionInterval,
FailureDetectionRetransmissions;
ProxySet 0 = "ProxySet_0", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"", "SIPInterface_0", "", "", 1, 1, 10, -1;
ProxySet 1 = "IP-PBX", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "", "",
"SIPInterface_0", "", "", 1, 1, 10, -1;
ProxySet 2 = "SIP Trunk", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"", "SIPInterface_0", "", "", 1, 1, 10, -1;
ProxySet 3 = "CQ-SIPREC", 1, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
"", "SIPInterface_0", "", "", 1, 1, 10, -1;

[ \ProxySet ]
    
```

```
[ IPGroup ]

FORMAT Index = Type, Name, ProxySetName, VoiceAIC Connector, SIPGroupName,
ContactUser, SipReRoutingMode, AlwaysUseRouteTable, SRDName, MediaRealm,
InternalMediaRealm, ClassifyByProxySet, ProfileName, MaxNumOfRegUsers,
InboundManSet, OutboundManSet, RegistrationMode, AuthenticationMode,
MethodList, SBCTServerAuthType, OAuthHTTPService, EnableSBCClientForking,
SourceUriInput, DestUriInput, ContactName, Username, Password, UIFormat,
QOEProfile, BWProfile, AlwaysUseSourceAddr, MsgManUserDefl,
MsgManUserDef2, SIPConnect, SBCPSAPMode, DTLSTContext,
CreatedByRoutingServer, UsedByRoutingServer, SBCOperationMode,
SBCRouteUsingRequestURIPort, SBCKeepOriginalCallID, TopologyLocation,
SBCDialPlanName, CallSetupRulesSetId, Tags, SBCUserStickiness,
UserUDPPortAssignment, AdmissionProfile, ProxyKeepAliveUsingIPG,
SBCAltRouteReasonsSetName, TeamsLocalMediaOptimization,
TeamsLocalMOInitialBehavior, SIPSourceHostName;
IPGroup 0 = 0, "Default_IPG", "ProxySet_0", "", "", "", -1, 0,
"DefaultSRD", "DefaultRealm", "", 0, "", -1, -1, -1, 0, 0, "", -1, "", 0,
-1, -1, "", "", "$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -
1, 0, 0, 0, "", -1, "", 0, 0, "", 0, "", 0, 0, "";
IPGroup 1 = 0, "IP-PBX", "IP-PBX", "", "", "", -1, 0, "DefaultSRD",
"DefaultRealm", "", 1, "Terminate-Refer", -1, -1, -1, 0, 0, "", -1, "",
0, -1, -1, "", "", "$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0,
0, -1, 0, 0, 0, "", -1, "", 0, 0, "", 0, "", 0, 0, "";
IPGroup 2 = 0, "SIP Trunk", "SIP Trunk", "", "", "", -1, 0, "DefaultSRD",
"DefaultRealm", "", 1, "Terminate-Refer", -1, -1, -1, 0, 0, "", -1, "",
0, -1, -1, "", "", "$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0,
0, -1, 0, 0, 1, "", -1, "", 0, 0, "", 0, "", 0, 0, "";
IPGroup 3 = 0, "CQ-SIPREC", "CQ-SIPREC", "", "", "", -1, 0, "DefaultSRD",
"DefaultRealm", "", 1, "", -1, -1, 1, 0, 0, "", -1, "", 0, -1, -1, "",
"", "$1$gQ==", 0, "", "", 0, "", "", 0, 0, "default", 0, 0, -1, 0, 0, 0,
"", -1, "", 0, 0, "", 0, "", 0, 0, "";

[ \IPGroup ]

[ ProxyIp ]

FORMAT Index = ProxySetId, ProxyIpIndex, IpAddress, TransportType,
Priority, Weight;
ProxyIp 0 = "1", 0, "10.15.77.14:5060", 0, 0, 0;
ProxyIp 1 = "2", 0, "10.15.77.18:5060", 0, 0, 0;
ProxyIp 2 = "3", 0, "10.15.5.100:5060", 0, 0, 0;

[ \ProxyIp ]

[ IP2IPRouting ]

FORMAT Index = RouteName, RoutingPolicyName, SrcIPGroupName,
SrcUsernamePrefix, SrcHost, DestUsernamePrefix, DestHost, RequestType,
MessageConditionName, ReRouteIPGroupName, Trigger, CallSetupRulesSetId,
DestType, DestIPGroupName, DestSIPInterfaceName, DestAddress, DestPort,
DestTransportType, AltRouteOptions, GroupPolicy, CostGroup, DestTags,
ModifiedDestUserName, SrcTags, IPGroupSetName, RoutingTagName,
InternalAction;
IP2IPRouting 0 = "Terminate Options", "Default_SBCRoutingPolicy", "Any",
"*, "*", "*", "*", 6, "", "Any", 0, -1, 13, "", "", "", 0, -1, 0, 0, "",
"", "", "", "", "default", "Reply (Response = '200')";
```

```

IP2IPRouting 1 = "IP-PBX to SIP trunk", "Default_SBCRoutingPolicy", "IP-
PBX", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "SIP Trunk", "", "", 0,
-1, 0, 0, "", "", "", "", "", "default", "";
IP2IPRouting 2 = "SIP Trunk to IP-PBX", "Default_SBCRoutingPolicy", "SIP
Trunk", "*", "*", "*", "*", 0, "", "Any", 0, -1, 0, "IP-PBX", "", "", 0,
-1, 0, 0, "", "", "", "", "", "default", "";

[ \IP2IPRouting ]

[ MessageManipulations ]

FORMAT Index = ManipulationName, ManSetID, MessageType, Condition,
ActionSubject, ActionType, ActionValue, RowRole;
MessageManipulations 0 = "SIPREC - Fix Metadata", 1, "Invite.Request",
'Body.application/rs-metadata regex (.*)(<nameID
aor=*)(.*)(@)(.*)(<nameID aor=*)(.*)(@)(.*)', "Body.application/rs-
metadata", 2, "$1+$2+'sip:' +$3+$4+$5+$6+'sip:'+$7+$8+$9", 0;

[ \MessageManipulations ]

[ GwRoutingPolicy ]

FORMAT Index = Name, LCREnable, LCRAverageCallLength, LCRDefaultCost,
LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 0, "";

[ \GwRoutingPolicy ]

[ ResourcePriorityNetworkDomains ]

FORMAT Index = Name, Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 1;
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 1;

[ \ResourcePriorityNetworkDomains ]

[ SIPRecRouting ]

FORMAT Index = RecordedIPGroupName, RecordedSourcePrefix,
RecordedDestinationPrefix, ConditionName, PeerIPGroupName,
PeerTrunkGroupID, Caller, SRSIPGroupName, SRSRedundantIPGroupName;
SIPRecRouting 0 = "IP-PBX", "*", "*", "", "Any", -1, 0, "CQ-SIPREC", "";
SIPRecRouting 1 = "SIP Trunk", "*", "*", "", "Any", -1, 0, "CQ-SIPREC",
"";

[ \SIPRecRouting ]

[ MaliciousSignatureDB ]

FORMAT Index = Name, Pattern;
    
```

```
MaliciousSignatureDB 0 = "SIPVicious", "Header.User-Agent.content prefix
'friendly-scanner'";
MaliciousSignatureDB 1 = "SIPScan", "Header.User-Agent.content prefix
'sip-scan'";
MaliciousSignatureDB 2 = "Smap", "Header.User-Agent.content prefix
'smap'";
MaliciousSignatureDB 3 = "Sipsak", "Header.User-Agent.content prefix
'sipsak'";
MaliciousSignatureDB 4 = "Sipcli", "Header.User-Agent.content prefix
'sipcli'";
MaliciousSignatureDB 5 = "Sivus", "Header.User-Agent.content prefix
'SIVuS'";
MaliciousSignatureDB 6 = "Gulp", "Header.User-Agent.content prefix
'Gulp'";
MaliciousSignatureDB 7 = "Sipv", "Header.User-Agent.content prefix
'sipv'";
MaliciousSignatureDB 8 = "Sundayddr Worm", "Header.User-Agent.content
prefix 'sundayddr'";
MaliciousSignatureDB 9 = "VaxIPUserAgent", "Header.User-Agent.content
prefix 'VaxIPUserAgent'";
MaliciousSignatureDB 10 = "VaxSIPUserAgent", "Header.User-Agent.content
prefix 'VaxSIPUserAgent'";
MaliciousSignatureDB 11 = "SipArmyKnife", "Header.User-Agent.content
prefix 'siparmyknife'";

[ \MaliciousSignatureDB ]

[ AudioCoders ]

FORMAT Index = AudioCodersGroupId, AudioCodersIndex, Name, pTime, rate,
PayloadType, Sce, CoderSpecific;
AudioCoders 0 = "AudioCodersGroups_0", 0, 1, 2, 90, -1, 0, "";
AudioCoders 1 = "AudioCodersGroups_0", 1, 2, 2, 90, -1, 0, "";

[ \AudioCoders ]
```

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