

## **Interactive Intelligence Customer Interaction Center and British Telecommunications SIP Trunk using AudioCodes Mediant™ E-SBC**

Version 7.0



**INTERACTIVE  
INTELLIGENCE®**





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## Notice

This document describes how to set up AudioCodes Enterprise Session Border Controller for interworking between British Telecommunications' (BT) SIP Trunk and Interactive Intelligence Customer Interaction Center environment.

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## Document Revision Record

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# 1 Introduction

This Configuration Note describes how to set up AudioCodes Enterprise Session Border Controller (hereafter, referred to as *E-SBC*) for interworking between British Telecommunications' (BT) SIP Trunk and Interactive Intelligence Customer Interaction Center environment.

## 1.1 Intended Audience

The document is intended for engineers, or AudioCodes and Interactive Intelligence or BT Partners who are responsible for installing and configuring BT's SIP Trunk and Interactive Intelligence Customer Interaction Center for enabling VoIP calls using AudioCodes E-SBC.

## 1.2 About AudioCodes E-SBC Product Series

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

The E-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 E-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 E-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.

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## 2 Component Information

### 2.1 AudioCodes E-SBC Version

Table 2-1: AudioCodes E-SBC Version

<b>SBC Vendor</b>	AudioCodes
<b>Models</b>	<ul style="list-style-type: none"> <li>▪ Mediant 500 E-SBC</li> <li>▪ Mediant 800 Gateway &amp; E-SBC</li> <li>▪ Mediant 1000B Gateway &amp; E-SBC</li> <li>▪ Mediant 3000 Gateway &amp; E-SBC</li> <li>▪ Mediant 2600 E-SBC</li> <li>▪ Mediant 4000 E-SBC</li> </ul>
<b>Software Version</b>	SIP_7.00A.013.006
<b>Protocol</b>	<ul style="list-style-type: none"> <li>▪ SIP/UDP (to the BT SIP Trunk)</li> <li>▪ SIP/TCP (to Interactive Intelligence)</li> </ul>
<b>Additional Notes</b>	None

### 2.2 BT SIP Trunking Version

Table 2-2: BT Version

<b>Vendor/Service Provider</b>	BT
<b>SSW Model/Service</b>	
<b>Software Version</b>	
<b>Protocol</b>	SIP
<b>Additional Notes</b>	None

### 2.3 Interactive Intelligence Customer Interaction Center Version

Table 2-3: 2.3 Interactive Intelligence Customer Interaction Center Version

<b>Vendor</b>	Interactive Intelligence
<b>Model</b>	Customer Interaction Center
<b>Software Version</b>	
<b>Protocol</b>	SIP
<b>Additional Notes</b>	None

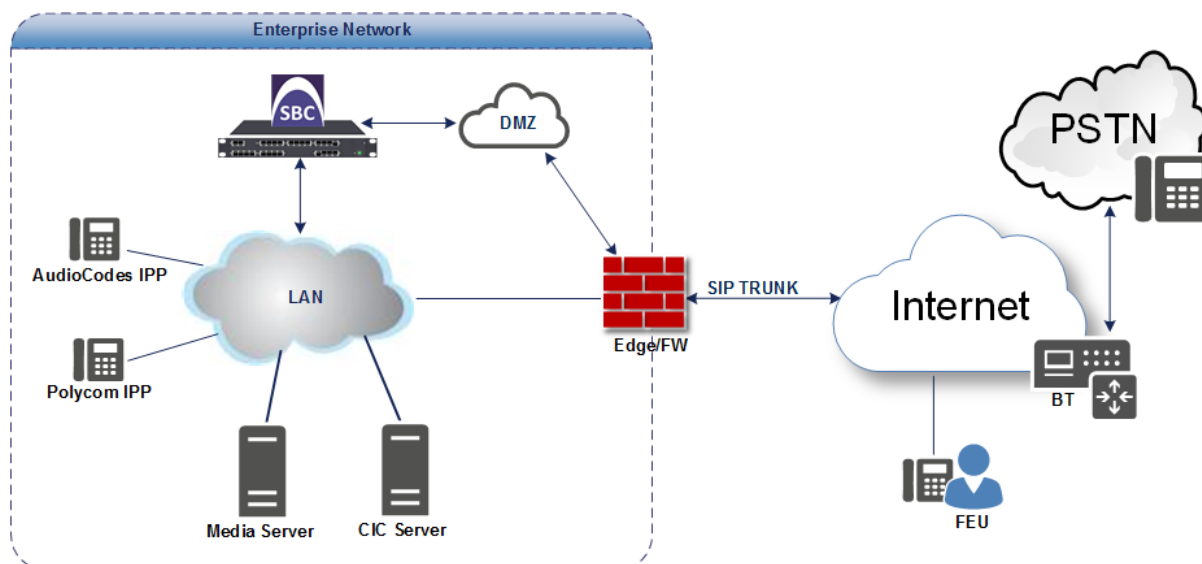
## 2.4 Interoperability Test Topology

The interoperability testing between AudioCodes E-SBC and BT SIP Trunk with Interactive Intelligence Customer Interaction Center was done using the following topology setup:

- Enterprise deployed with Interactive Intelligence Customer Interaction Center in its private network for enhanced communication within the Enterprise.
- Enterprise wishes to offer its employees enterprise-voice capabilities and to connect the Enterprise to the PSTN network using BT's SIP Trunking service using public external network.
- AudioCodes E-SBC is implemented to interconnect between the Enterprise LAN and the SIP Trunk.
  - **Session:** Real-time voice session using the IP-based Session Initiation Protocol (SIP).
  - **Border:** IP-to-IP network border between Interactive Intelligence Customer Interaction Center network in the Enterprise LAN and BT's SIP Trunk located in the public network.

The figure below illustrates this interoperability test topology:

**Figure 2-1: Interoperability Test Topology between E-SBC and Interactive Intelligence Customer Interaction Center with BT SIP Trunk**



## 2.4.1 Environment Setup

The interoperability test topology includes the following environment setup:

**Table 2-4: Environment Setup**

Area	Setup
<b>Network</b>	<ul style="list-style-type: none"> <li>▪ Interactive Intelligence Customer Interaction Center environment is located on the Enterprise's LAN</li> <li>▪ BT SIP Trunk is located on the WAN</li> </ul>
<b>Signaling Transcoding</b>	<ul style="list-style-type: none"> <li>▪ Interactive Intelligence Customer Interaction Center operates with SIP-over-TCP transport type</li> <li>▪ BT SIP Trunk operates with SIP-over-UDP transport type</li> </ul>
<b>Codecs Transcoding</b>	<ul style="list-style-type: none"> <li>▪ Interactive Intelligence Customer Interaction Center supports G.711A-law, G.711U-law, and G.729 coder</li> <li>▪ BT SIP Trunk supports G.711A-law, G.711U-law, and G.729 coder</li> </ul>
<b>Media Transcoding</b>	<ul style="list-style-type: none"> <li>▪ Interactive Intelligence Customer Interaction Center operates with RTP media type</li> <li>▪ BT SIP Trunk operates with RTP media type</li> </ul>

## 2.4.2 Known Limitations

There were no limitations observed in the interoperability tests done for the AudioCodes E-SBC interworking between Interactive Intelligence Customer Interaction Center and BT's SIP Trunk.

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## 3 Configuring AudioCodes E-SBC

This chapter provides step-by-step procedures on how to configure AudioCodes E-SBC for interworking between Interactive Intelligence Customer Interaction Center and the BT SIP Trunk. These configuration procedures are based on the interoperability test topology described in Section 2.4 on page 10, and includes the following main areas:

- E-SBC WAN interface - BT SIP Trunking environment
- E-SBC LAN interface - Interactive Intelligence Customer Interaction Center environment

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

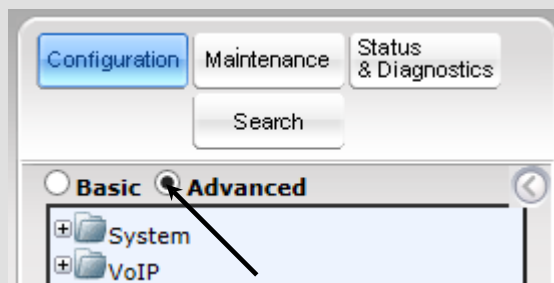
### Notes:

- For implementing Interactive Intelligence Customer Interaction Center and BT SIP Trunk based on the configuration described in this section, AudioCodes E-SBC must be installed with a Software License Key that includes the following software features:

- ✓ **SBC**
- ✓ **Security**
- ✓ **DSP**
- ✓ **RTP**
- ✓ **SIP**

For more information about the Software License Key, contact your AudioCodes sales representative.

- The scope of this interoperability test and document does **not** cover all security aspects for connecting the SIP Trunk to the Interactive Intelligence Customer Interaction Center environment. 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.
- Before you begin configuring the E-SBC, ensure that the E-SBC's Web interface Navigation tree is in Advanced-menu display mode. To do this, select the **Advanced** option, as shown below:



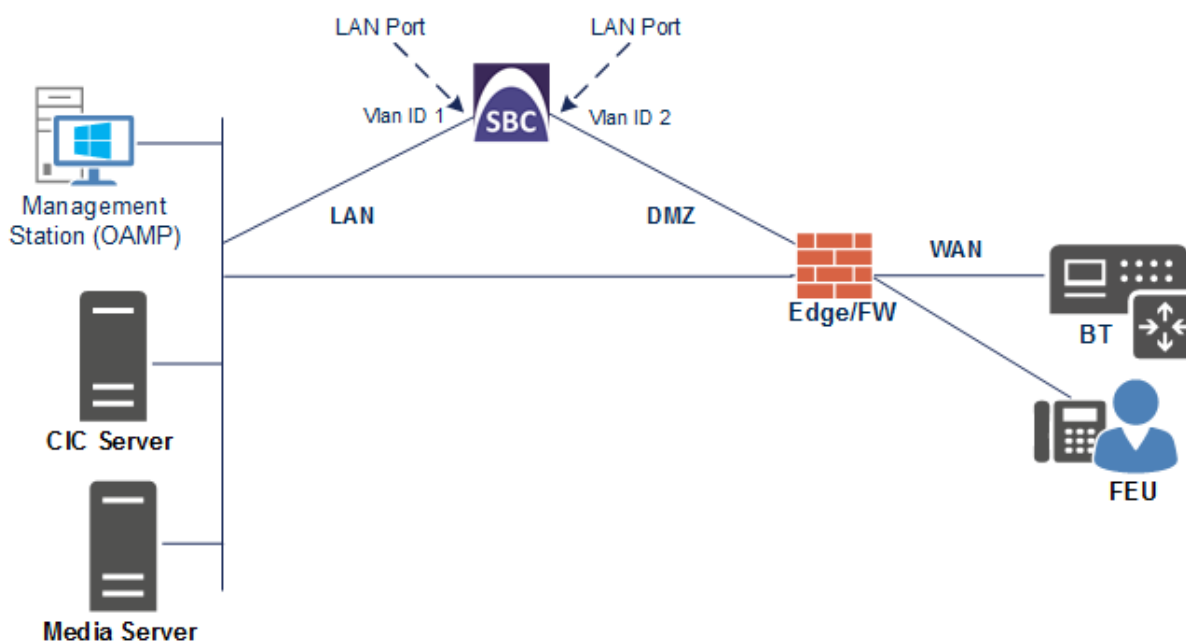
- When the E-SBC is reset, the Navigation tree reverts to Basic-menu display.

### 3.1 Step 1: IP Network Interfaces Configuration

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

- E-SBC interfaces with the following IP entities:
  - Interactive Intelligence Customer Interaction Center servers, located on the LAN
  - BT SIP Trunk, located on the WAN
- E-SBC connects to the WAN through a DMZ network
- Physical connection: The type of physical connection to the LAN depends on the method used to connect to the Enterprise's network. In the interoperability test topology, E-SBC connects to the LAN and WAN using dedicated LAN ports (i.e., two ports and two network cables are used).
- E-SBC also uses two logical network interfaces:
  - LAN (VLAN ID 1)
  - WAN (VLAN ID 2)

**Figure 3-1: Network Interfaces in Interoperability Test Topology**



### 3.1.1 Step 1a: Configure VLANs

This step describes how to define VLANs for each of the following interfaces:

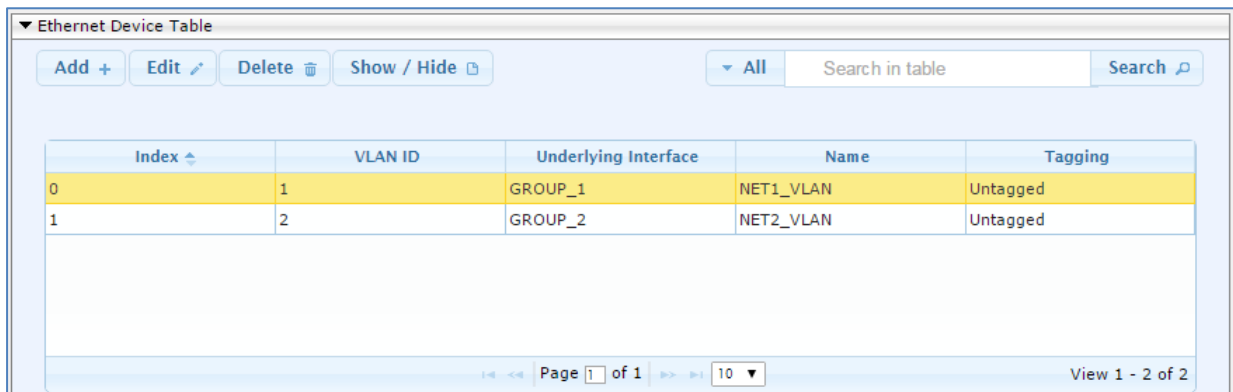
- LAN VoIP (assigned the name " NET1")
- WAN VoIP (assigned the name " NET2")

➤ **To configure the VLANs:**

1. Open the Ethernet Device Table page (**Configuration** tab > **VoIP** menu > **Network** > **Ethernet Device Table**).
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 as follows:

Parameter	Value
Index	<b>1</b>
VLAN ID	<b>2</b>
Underlying Interface	<b>GROUP_2</b> (Ethernet port group)
Name	<b>NET2_VLAN</b>
Tagging	<b>Untagged</b>

**Figure 3-2: Configured VLAN IDs in Ethernet Device Table**



### 3.1.2 Step 1b: Configure Network Interfaces

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

- LAN VoIP (assigned the name "NET1")
- WAN VoIP (assigned the name "NET2")

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

1. Open the IP Interfaces Table page (**Configuration** tab > **VoIP** menu > **Network** > **IP Interfaces Table**).
2. Modify the existing LAN network interface:
  - a. Select the 'Index' radio button of the **OAMP + Media + Control** table row, and then click **Edit**.
  - b. Configure the interface as follows:

Parameter	Value
IP Address	<b>192.168.1.212</b> (LAN IP address of E-SBC)
Prefix Length	<b>16</b> (subnet mask in bits for 255.255.0.0)
Default Gateway	<b>192.168.1.210</b>
Interface Name	<b>NET1</b> (arbitrary descriptive name)
Primary DNS Server IP Address	<b>192.168.1.201</b>
Underlying Device	<b>NET1_VLAN</b>

3. Add a network interface for the WAN side:
  - a. Enter **1**, and then click **Add Index**.
  - b. Configure the interface as follows:

Parameter	Value
Application Type	<b>Media + Control</b>
IP Address	<b>217.33.37.220</b> (WAN IP address of E-SBC)
Prefix Length	<b>25</b> (for 255.255.255.128)
Default Gateway	<b>217.33.37.193</b> (router's IP address)
Interface Name	<b>NET2</b> (arbitrary descriptive name)
Primary DNS Server IP Address	<b>8.8.8.8</b>
Underlying Device	<b>NET2_VLAN</b>

4. Click **Apply**.
5. Click **Done**.



The configured IP network interfaces are shown below:

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

Index	Interface Name	Application Type	Interface Mode	IP Address	Prefix Length	Default Gateway	Primary DNS	Secondary DNS	Underlying Device
0	NET1	OAMP + Media + IPv4 Manual	IPv4 Manual	192.168.1.212	16	192.168.1.210	192.168.1.201	0.0.0.0	NET1_VLAN
1	NET2	Media + Control	IPv4 Manual	217.33.37.220	25	217.33.37.193	8.8.8.8	0.0.0.0	NET2_VLAN

## 3.2 Step 2: Enable the SBC Application

This step describes how to enable the SBC application.

➤ **To enable the SBC application:**

1. Open the Applications Enabling page (**Configuration** tab > **VoIP** menu > **Applications Enabling** > **Applications Enabling**).

**Figure 3-4: Enabling SBC Application**



2. From the 'SBC Application' drop-down list, select **Enable**.
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for this setting to take effect (see Section 3.13 on page 50).

### 3.3 Step 3: Configure Media Realms

This step describes how to configure Media Realms. The simplest configuration is to create three Media Realms - one for internal (LAN) traffic, one for external (WAN) traffic towards SIP Trunk and another for external (WAN) traffic towards Far End Users.

➤ **To configure Media Realms:**

1. Open the Media Realm Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Media Realm Table**).
2. Add a Media Realm for the LAN interface. You can use the default Media Realm (Index 0), but modify it as shown below:

Parameter	Value
Index	<b>0</b>
Name	<b>realm0</b> (descriptive name)
IPv4 Interface Name	<b>NET1</b>
Port Range Start	<b>6000</b> (represents lowest UDP port number used for media on LAN)
Number of Media Session Legs	<b>100</b> (media sessions assigned with port range)

**Figure 3-5: Configuring Media Realm for LAN**

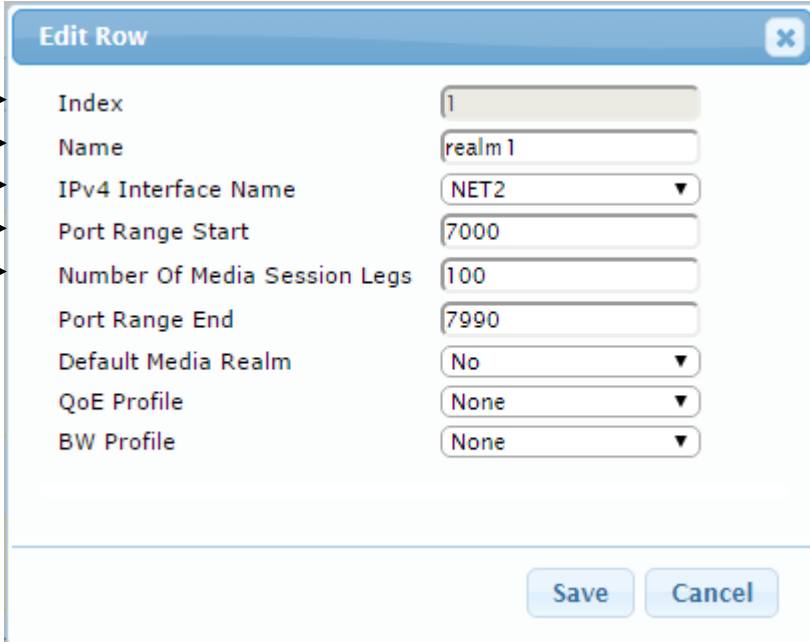
Edit Row
✕

Index	<input type="text" value="0"/>
Name	<input type="text" value="realm0"/>
IPv4 Interface Name	<input type="text" value="NET1"/>
Port Range Start	<input type="text" value="6000"/>
Number Of Media Session Legs	<input type="text" value="100"/>
Port Range End	<input type="text" value="6990"/>
Default Media Realm	<input type="text" value="No"/>
QoE Profile	<input type="text" value="None"/>
BW Profile	<input type="text" value="None"/>

3. Configure a Media Realm for WAN traffic towards SIP Trunk:

Parameter	Value
Index	<b>1</b>
Name	<b>realm1</b> (arbitrary name)
IPv4 Interface Name	<b>NET2</b>
Port Range Start	<b>7000</b> (represents lowest UDP port number used for media on WAN)
Number of Media Session Legs	<b>100</b> (media sessions assigned with port range)

**Figure 3-6: Configuring Media Realm for WAN towards SIP Trunk**



The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. The window contains the following fields and values:

- Index: 1
- Name: realm1
- IPv4 Interface Name: NET2 (dropdown menu)
- Port Range Start: 7000
- Number Of Media Session Legs: 100
- Port Range End: 7990
- Default Media Realm: No (dropdown menu)
- QoE Profile: None (dropdown menu)
- BW Profile: None (dropdown menu)

At the bottom of the window are "Save" and "Cancel" buttons. On the left side of the window, five horizontal arrows point to the Index, Name, IPv4 Interface Name, Port Range Start, and Number Of Media Session Legs fields.

4. Configure a Media Realm for WAN traffic towards Far End Users:

Parameter	Value
Index	2
Name	realmFEU (arbitrary name)
IPv4 Interface Name	NET2
Port Range Start	9000 (represents lowest UDP port number used for media on WAN towards FEU)
Number of Media Session Legs	100 (media sessions assigned with port range)

Figure 3-7: Configuring Media Realm for WAN Towards FEU

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

- Index: 2
- Name: realmFEU
- IPv4 Interface Name: NET2
- Port Range Start: 9000
- Number Of Media Session Legs: 100
- Port Range End: 9990
- Default Media Realm: No
- QoS Profile: None
- BW Profile: None

The configured Media Realms are shown in the figure below:

Figure 3-8: Configured Media Realms in Media Realm Table

Index	Name	IPv4 Interface Name	Port Range Start	Number Of Media Session Legs	Port Range End	Default Media Realm
0	realm0	NET1	6000	100	6990	Yes
1	realm1	NET2	7000	100	7990	No
2	realmFEU	NET2	9000	100	9990	No

## 3.4 Step 4: Configure SIP Signaling Interfaces

This step describes how to configure SIP Interfaces. For the interoperability test topology, one internal and two external SIP Interfaces must be configured for the E-SBC.

➤ **To configure SIP Interfaces:**

1. Open the SIP Interface Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **SIP Interface Table**).
2. Add a SIP Interface for the LAN interface:

Parameter	Value
Index	1
Interface Name	SIPInterface_1
Network Interface	NET1
Application Type	SBC
TCP Port	5060
TCP and UDP	0
Media Realm	realm0

3. Configure a SIP Interface for the WAN for SIP Trunk:

Parameter	Value
Index	2
Interface Name	SIPInterface_2
Network Interface	NET2
Application Type	SBC
UDP Port	5060
TCP and TLS	0
Media Realm	realm1

4. Configure a SIP Interface for the WAN for Far End Users:

Parameter	Value
Index	4
Interface Name	SIPInterface_4
Network Interface	NET2
Application Type	SBC
UDP Port	5070
TCP and TLS	0
Media Realm	realmFEU

The configured SIP Interfaces are shown in the figure below:

**Figure 3-9: Configured SIP Interfaces in SIP Interface Table**

The screenshot shows a web-based configuration interface for SIP interfaces. At the top, there are buttons for 'Add +', 'Edit', 'Delete', and 'Show / Hide'. To the right, there is a search bar with a dropdown menu set to 'All' and a 'Search' button. Below these elements is a table with the following data:

Index	Name	SRD	Network Interface	Application Type	UDP Port	TCP Port	TLS Port	Encapsulatio Protocol	Media Realm
1	SIPInterface_1	DefaultSRC	NET1	SBC	0	5060	0	No encapsula	realm0
2	SIPInterface_2	DefaultSRC	NET2	SBC	5060	0	0	No encapsula	realm1
4	SIPInterface_4	DefaultSRC	NET2	SBC	5070	0	0	No encapsula	realmFEU

At the bottom of the table, there is a pagination control showing 'Page 1 of 1' and a dropdown menu set to '10'. On the far right, it says 'View 1 - 3 of 3'.

### 3.5 Step 5: Configure Proxy Sets

This step 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, Proxy Sets need to be configured for the following IP entities:

- Interactive Intelligence Customer Interaction Center
- BT SIP Trunk

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

➤ **To configure Proxy Sets:**

1. Open the Proxy Sets Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table**).
2. Add a Proxy Set for the Interactive Intelligence Customer Interaction Center:

Parameter	Value
Index	1
Name	ProxySet_1
SBC IPv4 SIP Interface	SIPInterface_1
Redundancy Mode	Homing
Proxy Hot Swap	Enable

Figure 3-10: Configuring Proxy Set for Interactive Intelligence Customer Interaction Center

The screenshot shows the 'Edit Row' configuration window for a Proxy Set. The parameters and their values are as follows:

- Index: 1
- SRD: DefaultSRD
- Name: ProxySet\_1
- Gateway IPv4 SIP Interface: None
- SBC IPv4 SIP Interface: SIPInterface\_1
- Proxy Keep-Alive: Disable
- Proxy Keep-Alive Time [sec]: 60
- Redundancy Mode: Homing
- Proxy Load Balancing Method: Disable
- DNS Resolve Method: (empty)
- Proxy Hot Swap: Enable
- Keep-Alive Failure Responses: (empty)
- Classification Input: IP Address only
- TLS Context Name: None

Arrows on the left side of the window point to the Index, Name, SBC IPv4 SIP Interface, Redundancy Mode, and Proxy Hot Swap fields.



3. Configure a Proxy Address Table for Proxy Set for Interactive Intelligence Customer Interaction Center:
  - a. Navigate to **Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**.

Parameter	Value
Index	<b>0</b>
Proxy Address	<b>192.168.1.202:5060</b> (Interactive Intelligence Customer Interaction Center IP address / FQDN and destination port)
Transport Type	<b>TCP</b>

**Figure 3-11: Configuring Proxy Address for Interactive Intelligence Customer Interaction Center**

The screenshot shows a dialog box titled "Edit Row" with a close button (X) in the top right corner. It contains three input fields: "Index" with the value "0", "Proxy Address" with the value "192.168.1.202:5060", and "Transport Type" with a dropdown menu showing "TCP". At the bottom right, there are two buttons: "Save" and "Cancel".

4. Configure a Proxy Set for the BT SIP Trunk:

Parameter	Value
Index	5
Name	BT SIP SBC
SBC IPv4 SIP Interface	SIPInterface_2
Proxy Keep-Alive	Using Options
Redundancy Mode	Homing
Proxy Hot Swap	Enable

Figure 3-12: Configuring Proxy Set for BT SIP Trunk

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. The window contains the following parameters and values:

- Index: 5
- SRD: DefaultSRD
- Name: BT SIP SBC
- Gateway IPv4 SIP Interface: None
- SBC IPv4 SIP Interface: SIPInterface\_2
- Proxy Keep-Alive: Using OPTIONS
- Proxy Keep-Alive Time [sec]: 60
- Redundancy Mode: Homing
- Proxy Load Balancing Method: Disable
- DNS Resolve Method: (empty dropdown)
- Proxy Hot Swap: Enable
- Keep-Alive Failure Responses: (empty text field)
- Classification Input: IP Address only
- TLS Context Name: None

At the bottom of the window are "Save" and "Cancel" buttons. Arrows on the left side of the window point to the Index, Name, SBC IPv4 SIP Interface, Proxy Keep-Alive, Redundancy Mode, Proxy Hot Swap, and Classification Input fields.

- a. Configure a Proxy Address Table for Proxy Set for the BT SIP Trunk:
- b. Navigate to **Configuration** tab > **VoIP** menu > **VoIP Network** > **Proxy Sets Table** > **Proxy Address Table**.

Parameter	Value
Index	<b>0</b>
Proxy Address	<b>192.65.221.5</b> (BT SIP Trunk IP address / FQDN and destination port)
Transport Type	<b>UDP</b>

**Figure 3-13: Configuring Proxy Address for BT SIP Trunk**

Edit Row
✕

Index

Proxy Address

Transport Type

The configured Proxy Sets are shown in the figure below:

**Figure 3-14: 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
1	ProxySet_1	DefaultSRD (#0	None	SIPInterface_1	60	Homing	Enable
2	ProxySet_2	DefaultSRD (#0	None	SIPInterface_2	60	Homing	Enable
3	ProxySet_3	DefaultSRD (#0	None	SIPInterface_2	60	Homing	Disable
4	ProxySet_4	DefaultSRD (#0	None	SIPInterface_1	60		Disable
5	BT SIP SBC	DefaultSRD (#0	None	SIPInterface_2	60	Homing	Enable

### 3.6 Step 6: Configure IP Profiles

This step 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, IP Profiles need to be configured for the following IP entities:

- Interactive Intelligence Customer Interaction Center
  - GSM GW
  - VoIPTalk
  - Remote Users
  - BT SIP trunk
- **To configure IP Profile for the Interactive Intelligence Customer Interaction Center:**
1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
  2. Click **Add**.
  3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	1
Name	CIC
Broken Connection Mode	Ignore

**Figure 3-15: Configuring IP Profile for Interactive Intelligence Customer Interaction Center – Common Tab**

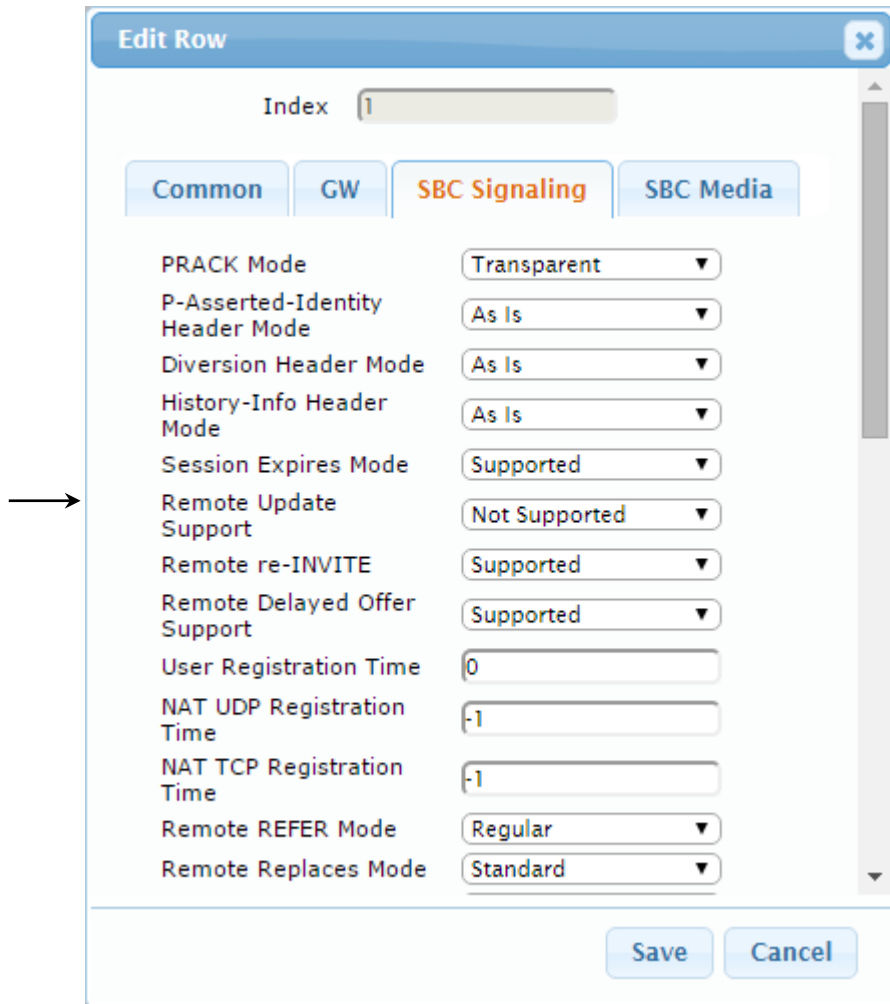
The screenshot shows the 'Edit Row' dialog box with the following configuration:

- Index:** 1
- Common Tab:** Selected
- Name:** CIC
- Dynamic Jitter Buffer Minimum Delay [msec]:** 10
- Dynamic Jitter Buffer Optimization Factor:** 10
- Jitter Buffer Max Delay [msec]:** 300
- RTP IP DiffServ:** 46
- Signaling DiffServ:** 40
- Silence Suppression:** Disable
- RTP Redundancy Depth:** 0
- Echo Canceled:** Line
- Broken Connection Mode:** Ignore
- Input Gain (-32 to 31 dB):** 0
- Voice Volume (-32 to 31 dB):** 0
- Media IP Version:** Only IPv4

- Click the **SBC Signaling** tab, and then configure the parameters as follows:

Parameter	Value
Remote Update Support	<b>Not Supported</b>

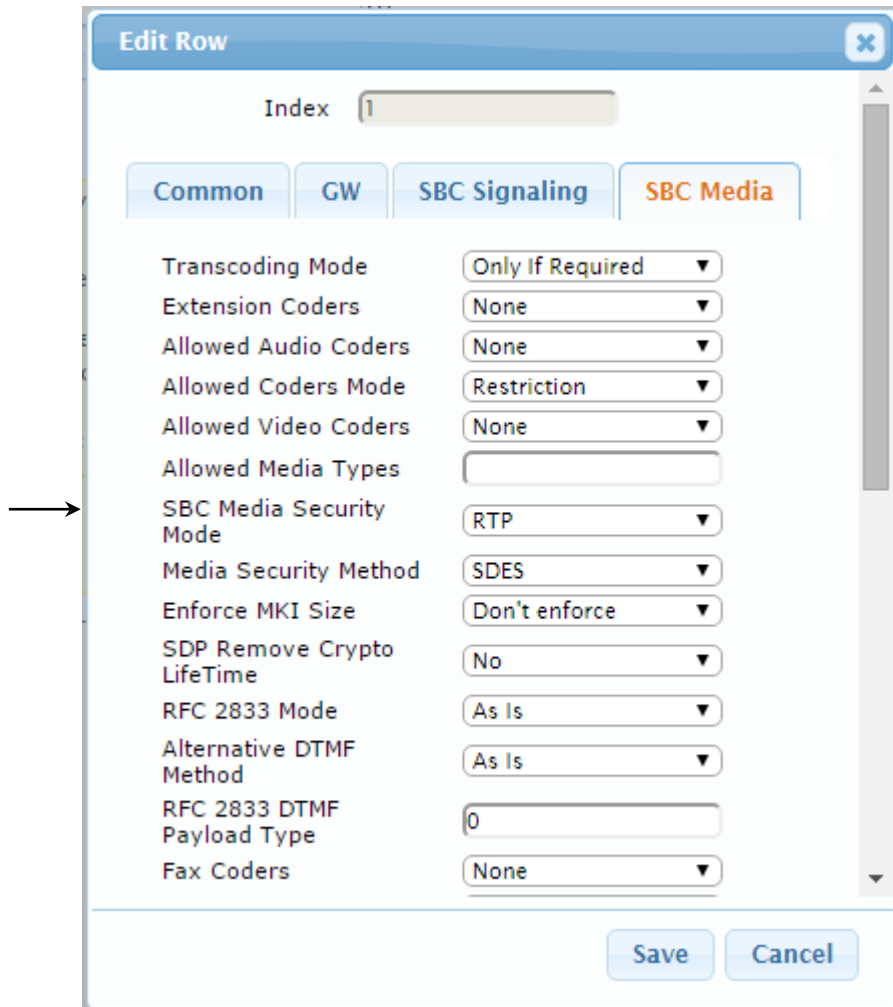
**Figure 3-16: Configuring IP Profile for Interactive Intelligence Customer Interaction Center – SBC Signaling Tab**



- Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
SBC Media Security Mode	RTP

**Figure 3-17: Configuring IP Profile for Interactive Intelligence Customer Interaction Center – SBC Media Tab**



➤ **To configure IP Profile for the GSM GW:**

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	<b>2</b>
Name	<b>GSM GW</b>
Broken Connection Mode	<b>Ignore</b>

4. Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
SBC Media Security Mode	<b>RTP</b>

➤ **To configure IP Profile for the VoIPTalk:**

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	<b>3</b>
Name	<b>VoIPTalk</b>

➤ **To configure IP Profile for the Remote Users:**

1. Open the IP Profile Settings page (**Configuration** tab > **VoIP** > **Coders and Profiles** > **IP Profile Settings**).
2. Click **Add**.
3. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	<b>4</b>
Name	<b>Remote Users</b>
Broken Connection Mode	<b>Ignore</b>

4. Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
SBC Media Security Mode	<b>RTP</b>

➤ **To configure an IP Profile for the BT SIP Trunk:**

1. Click **Add**.
2. Click the **Common** tab, and then configure the parameters as follows:

Parameter	Value
Index	5
Name	BT

**Figure 3-18: Configuring IP Profile for BT SIP Trunk – Common Tab**

**Edit Row** [Close]

Index:

**Common**
 GW
  SBC Signaling
  SBC Media

Name:

Dynamic Jitter Buffer Minimum Delay [msec]:

Dynamic Jitter Buffer Optimization Factor:

Jitter Buffer Max Delay [msec]:

RTP IP DiffServ:

Signaling DiffServ:

Silence Suppression:

RTP Redundancy Depth:

Echo Canceled:

Broken Connection Mode:

Input Gain (-32 to 31 dB):

Voice Volume (-32 to 31 dB):

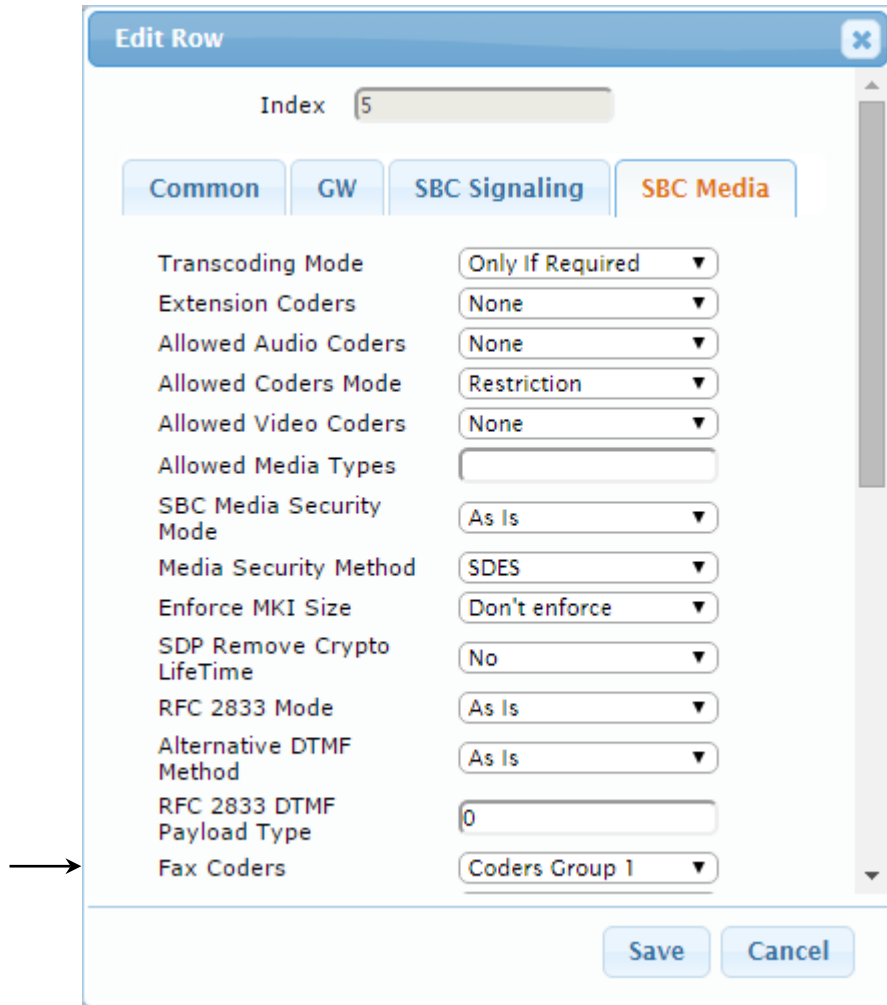
Media IP Version:



3. Click the **SBC Media** tab, and then configure the parameters as follows:

Parameter	Value
Fax Coders	Coders Group 1

Figure 3-19: Configuring IP Profile for BT SIP Trunk – SBC Media Tab



## 3.7 Step 7: Configure IP Groups

This step describes how to configure IP Groups. The IP Group represents an IP entity on the network with which the E-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 or Remote users). 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:

- Interactive Intelligence Customer Interaction Center located on LAN
- GSM GW located on WAN
- VoIPTalk located on WAN
- Remote Users located on WAN
- BT SIP Trunk located on WAN

➤ **To configure IP Groups:**

1. Open the IP Group Table page (**Configuration** tab > **VoIP** menu > **VoIP Network** > **IP Group Table**).
2. Add an IP Group for the Interactive Intelligence Customer Interaction Center.

Parameter	Value
Index	<b>1</b>
Name	<b>CIC</b>
Type	<b>Server</b>
Proxy Set	<b>ProxySet_1</b>
IP Profile	<b>CIC</b>
Media Realm	<b>realm0</b>
SIP Group Name	<b>217.33.37.220</b> (according to requirement)

3. Add an IP Group for the GSM GW.

Parameter	Value
Index	<b>2</b>
Name	<b>GSM GW</b>
Type	<b>Server</b>
Proxy Set	<b>ProxySet_2</b>
IP Profile	<b>GSM GW</b>
Media Realm	<b>realm1</b>
SIP Group Name	<b>WAN</b> (according to requirement)

## 4. Add an IP Group for the VoIPTalk.

Parameter	Value
Index	<b>3</b>
Name	<b>VoIPTalk</b>
Type	<b>Server</b>
Proxy Set	<b>ProxySet_3</b>
IP Profile	<b>VoIPTalk</b>
Media Realm	<b>realm1</b>
SIP Group Name	<b>voiptalk.org</b> (according to requirement)
Contact User	<b>voiptalk.org</b> (according to requirement)

## 5. Add an IP Group for the Remote Users.

Parameter	Value
Index	<b>4</b>
Name	<b>Remote Users</b>
Type	<b>User</b>
Proxy Set	<b>None</b>
IP Profile	<b>Remote Users</b>
Media Realm	<b>realmFEU</b>
Always Use Src Address	<b>Yes</b>
Classify By Proxy Set	<b>Disable</b>

## 6. Add an IP Group for the BT SIP Trunk.

Parameter	Value
Index	<b>5</b>
Name	<b>BT</b>
Type	<b>Server</b>
Proxy Set	<b>BT SIP SBC</b>
IP Profile	<b>BT</b>
Media Realm	<b>realm1</b>
SIP Group Name	<b>192.65.221.26</b> (according to requirement)

The configured IP Groups are shown in the figure below:

**Figure 3-20: 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 Manipulat Set
1	CIC	<input type="checkbox"/> DefaultS	Server	Not Configu	ProxySet_1	CIC	realm0	217.33.37..	Enable	-1	-1
2	GSM GW	<input checked="" type="checkbox"/> DefaultS	Server	Not Configu	ProxySet_2	GSM GW	realm1	WAN	Enable	-1	-1
3	VoipTalk	<input type="checkbox"/> DefaultS	Server	Not Configu	ProxySet_3	VoIPTalk	realm1	voiptalk.org	Enable	-1	-1
4	Remote Us	<input checked="" type="checkbox"/> DefaultS	User	Not Configu	None	Remote Users	realmFEU		Disable	-1	-1
5	BT	<input checked="" type="checkbox"/> DefaultS	Server	Not Configu	BT SIP SBC	BT	realm1	192.65.221	Enable	-1	-1

Page  of 1
 
View 1 - 5 of 5

### 3.8 Step 8: Configure Coders

This step describes how to configure coders (termed *Coder Group*). As BT SIP Trunk supports the T.38 fax coder while the Interactive Intelligence Customer Interaction Center may restrict operation with only fax over G.711 coder, you need to add a Fax Coder Group with the T.38 coder for the BT SIP Trunk.

Note that the Coder Group ID for this entity was assigned to the BT SIP Trunk IP Profile in the previous step (see Section 3.6 on page 28).

➤ **To configure coders:**

1. Open the Coder Group Settings (**Configuration** tab > **VoIP** menu > **Coders and Profiles** > **Coders Group Settings**).
2. Configure a Coder Group for the BT SIP Trunk:

Parameter	Value
Coder Group ID	1
Coder Name	<ul style="list-style-type: none"> <li>▪ G.711 U-law</li> <li>▪ G.711 A-law</li> <li>▪ G.729</li> <li>▪ T.38</li> </ul>

**Figure 3-21: Configuring Coder Group for BT SIP Trunk**

▼					
Coder Group ID				1 ▼	
Coder Name	Packetization Time	Rate	Payload Type	Silence Suppression	Coder Specific
G.711A-law ▼	20 ▼	64 ▼	8	Disabled ▼	
G.711U-law ▼	20 ▼	64 ▼	0	Disabled ▼	
G.729 ▼	20 ▼	8 ▼	18	Disabled ▼	
T.38 ▼	N/A ▼	N/A ▼	N/A	N/A ▼	
▼	▼	▼		▼	

## 3.9 Step 9: Configure Maximum IP Media Channels

This step describes how to configure the maximum number of required IP media channels. The number of media channels represents the number of DSP channels that the E-SBC allocates to call sessions.



**Note:** This step is required **only** if transcoding is required.

➤ **To configure the maximum number of IP media channels:**

1. Open the IP Media Settings page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Advanced Parameters**).

**Figure 3-22: Configuring Number of Media Channels**

Number of Media Channels	30
--------------------------	----

2. In the 'Number of Media Channels' field, enter the number of media channels according to your environments transcoding calls (e.g., **30**).
3. Click **Submit**.
4. Reset the E-SBC with a burn to flash for your settings to take effect (see Section [03.13](#) on page [50](#)).

## 3.10 Step 10: Configure IP-to-IP Call Routing Rules

This step 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 E-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. The routing rules use the configured IP Groups to denote the source and destination of the call. As configured in Section 3.7 on page 27, IP Group 1 represents Interactive Intelligence Customer Interaction Center, and IP Group 5 represents BT SIP Trunk.

For the interoperability test topology, the following IP-to-IP routing rules need to be configured to route calls between Interactive Intelligence Customer Interaction Center (LAN) and BT SIP Trunk (WAN):

- Terminate SIP OPTIONS messages on the E-SBC that are received from both LAN and WAN
- Calls from Interactive Intelligence Customer Interaction Center to BT SIP Trunk
- Calls from BT SIP Trunk to Interactive Intelligence Customer Interaction Center

- **To configure IP-to-IP routing rules:**
- 1. Open the IP-to-IP Routing Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **IP-to-IP Routing Table**).
- 2. Configure a rule to terminate SIP OPTIONS messages received from the LAN:
  - a. Click **Add**.
  - b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	<b>0</b>
Name	<b>Terminate OPTIONS</b> (arbitrary descriptive name)
Source IP Group	<b>Any</b>
Request Type	<b>OPTIONS</b>

**Figure 3-23: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Rule Tab**

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

- Index:** 0
- Routing Policy:** Default\_SBCRouting
- Rule Tab:** Selected
- Name:** Terminate OPTIONS
- Alternative Route Options:** Route Row
- Source IP Group:** Any
- Request Type:** OPTIONS
- Source Username Prefix:** \*
- Source Host:** \*
- Destination Username Prefix:** \*
- Destination Host:** \*
- Message Condition:** None
- Call Trigger:** Any
- ReRoute IP Group:** Any

Buttons: [Classic View](#), **Save**, **Cancel**



c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	<b>Dest Address</b>
Destination Address	<b>internal</b>

**Figure 3-24: Configuring IP-to-IP Routing Rule for Terminating SIP OPTIONS – Action Tab**

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

- Index: 0
- Routing Policy: Default\_SBCRouting
- Tab: Action
- Destination Type: Dest Address
- Destination IP Group: None
- Destination SIP Interface: None
- Destination Address: internal
- Destination Port: 0
- Destination Transport Type: (empty dropdown)
- Call Setup Rules Set ID: -1
- Group Policy: None
- Cost Group: None

Buttons: Add, Cancel, Classic View

3. Configure a rule to route calls from Interactive Intelligence Customer Interaction Center to BT SIP Trunk:
  - a. Click **Add**.
  - b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	1
Name	CIC to BT (arbitrary descriptive name)
Source IP Group	CIC

Figure 3-25: Configuring IP-to-IP Routing Rule for CIC to BT – Rule tab

The screenshot shows the 'Edit Row' configuration window for a routing rule. The 'Rule' tab is selected. The configuration parameters are as follows:

- Index: 1
- Routing Policy: Default\_SBCRouting
- Name: CIC to BT
- Alternative Route Options: Route Row
- Source IP Group: CIC
- Request Type: All
- Source Username Prefix: \*
- Source Host: \*
- Destination Username Prefix: \*
- Destination Host: \*
- Message Condition: None
- Call Trigger: Any
- ReRoute IP Group: Any

Buttons for 'Save' and 'Cancel' are visible at the bottom. A 'Classic View' link is also present.

c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	IP Group
Destination IP Group	BT
Destination SIP Interface	SIPInterface_2

Figure 3-26: Configuring IP-to-IP Routing Rule for CIC to BT – Action tab

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. Below the title bar, there are two input fields: "Index" with the value "1" and "Routing Policy" with a dropdown menu showing "Default\_SBCRouting". Below these are two tabs: "Rule" and "Action", with "Action" being the active tab. The "Action" tab contains several configuration fields:

- Destination Type: dropdown menu showing "IP Group"
- Destination IP Group: dropdown menu showing "BT"
- Destination SIP Interface: dropdown menu showing "SIPInterface\_2"
- Destination Address: empty text input field
- Destination Port: text input field with "0"
- Destination Transport Type: dropdown menu
- Call Setup Rules Set ID: text input field with "-1"
- Group Policy: dropdown menu showing "None"
- Cost Group: dropdown menu showing "None"

At the bottom right of the window, there is a link labeled "Classic View". At the very bottom, there are "Save" and "Cancel" buttons. Three arrows on the left side of the window point to the "Destination Type", "Destination IP Group", and "Destination SIP Interface" fields.

4. To configure rule to route calls from BT SIP Trunk to Interactive Intelligence Customer Interaction Center:
  - a. Click **Add**.
  - b. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	2
Name	BT to CIC (arbitrary descriptive name)
Source IP Group	BT

Figure 3-27: Configuring IP-to-IP Routing Rule for BT to CIC – Rule tab

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

- Index: 2
- Routing Policy: Default\_SBCRouting
- Tab: Rule
- Name: BT to CIC
- Alternative Route Options: Route Row
- Source IP Group: BT
- Request Type: All
- Source Username Prefix: \*
- Source Host: \*
- Destination Username Prefix: \*
- Destination Host: \*
- Message Condition: None
- Call Trigger: Any
- ReRoute IP Group: Any

Buttons: Save, Cancel

c. Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Destination Type	IP Group
Destination IP Group	CIC
Destination SIP Interface	SIPInterface_1

Figure 3-28: Configuring IP-to-IP Routing Rule for BT to CIC – Action tab

The screenshot shows a configuration window titled "Edit Row" with a close button (X) in the top right corner. Below the title bar, there are two input fields: "Index" with the value "2" and "Routing Policy" with a dropdown menu showing "Default\_SBCRouting". Below these are two tabs: "Rule" and "Action", with "Action" being the active tab. The "Action" tab contains several configuration fields:

- Destination Type: dropdown menu showing "IP Group"
- Destination IP Group: dropdown menu showing "CIC"
- Destination SIP Interface: dropdown menu showing "SIPInterface\_1"
- Destination Address: empty text input field
- Destination Port: text input field with "0"
- Destination Transport Type: dropdown menu
- Call Setup Rules Set ID: text input field with "-1"
- Group Policy: dropdown menu showing "None"
- Cost Group: dropdown menu showing "None"

At the bottom right of the configuration area, there is a link labeled "Classic View". At the very bottom of the window, there are two buttons: "Save" and "Cancel". On the left side of the window, three horizontal arrows point to the "Destination Type", "Destination IP Group", and "Destination SIP Interface" fields respectively.

The configured routing rules are shown in the figure below:

**Figure 3-29: Configured IP-to-IP Routing Rules in IP-to-IP Routing Table**

Inde:	Name	Routing Policy	Alternati Route Options	Source IP Group	Request Type	Source Usernam Prefix	Destinati Usernam Prefix	Destinati Type	Destinati IP Group	Destination SIP Interface	Destinat Address
0	Terminate	Default_SB	Route Row	Any	OPTIONS	*	*	Dest Adre	None	None	internal
1	CIC to BT	Default_SB	Route Row	CIC	All	*	*	IP Group	BT	SIPInterface_2	
2	BT to CIC	Default_SB	Route Row	BT	All	*	*	IP Group	CIC	SIPInterface_1	

Page 1 of 1 | 10 | View 1 - 3 of 3



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

### 3.11 Step 11: Configure Registration Accounts

This step describes how to configure SIP registration accounts. This is required so that the E-SBC can register with the VoIPTalk on behalf of Interactive Intelligence Customer Interaction Center. The VoIPTalk service requires authentication to provide service.

In the interoperability test topology, the Served IP Group is Interactive Intelligence Customer Interaction Center IP Group and the Serving IP Group is VoIPTalk IP Group.

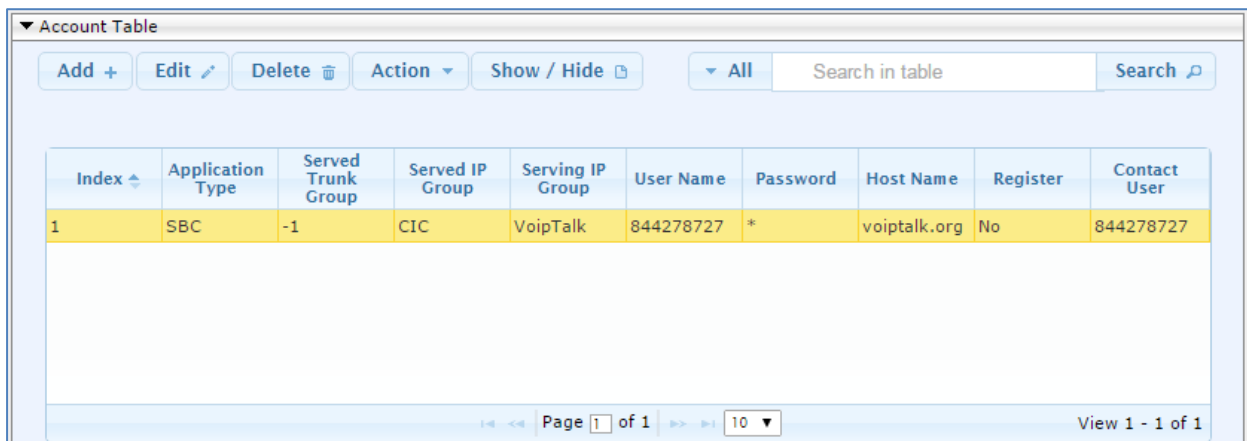
➤ **To configure a registration account:**

1. Open the Account Table page (**Configuration** tab > **VoIP** menu > **SIP Definitions** > **Account Table**).
2. Enter an index number (e.g., "1"), and then click **Add**.
3. Configure the account according to the provided information from , for example:

Parameter	Value
Application Type	<b>SBC</b>
Served IP Group	<b>CIC</b>
Serving IP Group	<b>VoipTalk</b>
Username	<b>844278727</b> (as provided by customer)
Password	as provided by customer
Host Name	<b>voiptalk.org</b>
Register	<b>No</b>
Contact User	<b>844278727</b> (as provided by customer)

4. Click **Apply**.

**Figure 3-30: Configuring SIP Registration Account**



### 3.12 Step 12: Configure Classification Table

This section describes how to configure the E-SBC Classification Table. In the current interoperability test topology, it's necessary to allow messages to be received from different entities. The Classification Table does this.

➤ **To configure Classification Table:**

1. Open the Classification Table page (**Configuration** tab > **VoIP** menu > **SBC** > **Routing SBC** > **Classification Table**).
2. Click **Add**.
3. Click the **Rule** tab, and then configure the parameters as follows:

Parameter	Value
Index	<b>0</b>
Name	<b>Allow remote users</b> (arbitrary descriptive name)
Source SIP Interface	<b>SIPInterface_4</b>
Source Transport Type	<b>UDP</b>

**Figure 3-31: Classification Table Page – Rule Tab**

The screenshot shows a web-based configuration window titled "Edit Row". At the top, there are two tabs: "Rule" (selected) and "Action". Below the tabs, the following fields are visible:

- Index: 0
- SRD: DefaultSRD
- Name: Allow remote users
- Source SIP Interface: SIPInterface\_4
- Source IP Address: (empty)
- Source Transport Type: UDP
- Source Port: 0
- Source Username Prefix: \*
- Source Host: \*
- Destination Username Prefix: \*
- Destination Host: \*
- Message Condition: None

At the bottom right, there is a link for "Classic View". At the bottom center, there are "Save" and "Cancel" buttons.



- Click the **Action** tab, and then configure the parameters as follows:

Parameter	Value
Action Type	<b>Allow</b>
Source IP Group	<b>Remote Users</b>
IP Profile	<b>Remote Users</b>

Figure 3-32: Classification Table Page – Action Tab

- Click **Save**.
- Click **Submit**.

Figure 3-33: Example of Classification Table

Index	Name	SRD	Source SIP Interface	Source Username Prefix	Source Host	Destination Username Prefix	Destination Host	Action Type	Source IP Group
0	Allow remote	DefaultSRD	SIPInterface_*	*	*	*	*	Allow	Remote User

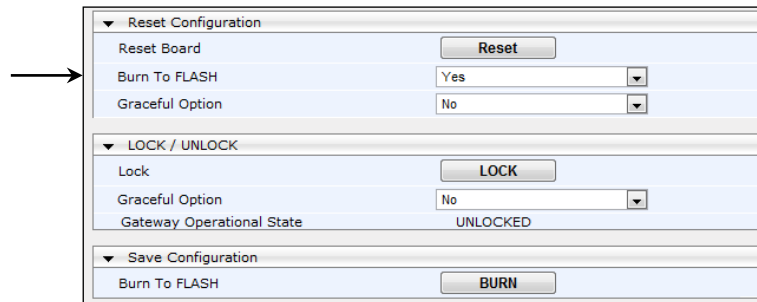
### 3.13 Step 13: Reset the E-SBC

After you have completed the configuration of the E-SBC described in this chapter, save ("burn") the configuration to the E-SBC's flash memory with a reset for the settings to take effect.

➤ **To save the configuration to flash memory:**

1. Open the Maintenance Actions page (**Maintenance** tab > **Maintenance** menu > **Maintenance Actions**).

**Figure 3-34: Resetting the E-SBC**



▼ Reset Configuration	
Reset Board	<input type="button" value="Reset"/>
Burn To FLASH	Yes ▼
Graceful Option	No ▼
▼ LOCK / UNLOCK	
Lock	<input type="button" value="LOCK"/>
Graceful Option	No ▼
Gateway Operational State	UNLOCKED
▼ Save Configuration	
Burn To FLASH	<input type="button" value="BURN"/>

2. Ensure that the 'Burn to FLASH' field is set to **Yes** (default).
3. Click the **Reset** button.

## A AudioCodes INI File

The *ini* configuration file of the E-SBC, corresponding to the Web-based configuration as described in Section 3 on page 13, is shown below:



**Note:** To load and save an ini file, use the Configuration File page (**Maintenance** tab > **Software Update** menu > **Configuration File**).

```

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

;Board: Mediant 800B
;HW Board Type: 69  FK Board Type: 72
;Serial Number: 7637055
;Slot Number: 1
;Software Version: 7.00A.013.006
;DSP Software Version: 5014AE3_R => 700.32
;Board IP Address: 192.168.1.212
;Board Subnet Mask: 255.255.0.0
;Board Default Gateway: 192.168.1.210
;Ram size: 496M  Flash size: 64M  Core speed: 500Mhz
;Num of DSP Cores: 3  Num DSP Channels: 90
;Num of physical LAN ports: 12
;Profile: NONE
;;Key features;;Board Type: Mediant 800B ;BRITrunks=1 ;Security: IPSEC
MediaEncryption StrongEncryption EncryptControlProtocol ;DATA features:
;Channel Type: RTP DspCh=90 ;HA ;Coders: G723 G729 GSM-FR G727 ;PSTN
Protocols: ISDN IUA=2 CAS ;DSP Voice features: IpmDetector ;IP Media:
VXML ;Control Protocols: SIP SBC=100 MSFT FEU=50 ;Default
features;;Coders: G711 G726;

;----- HW components-----
;
; Slot # : Module type : # of ports
;-----
;      1 : Empty
;      2 : Empty
;      3 : Empty
;-----

[SYSTEM Params]

;NTPServerIP_abs is hidden but has non-default value
TelnetServerEnable = 0
;VpFileLastUpdateTime is hidden but has non-default value
NTPServerIP = '192.168.1.202'
NTPSecondaryServerIP = '192.168.1.201'
;PM_gwINVITEDialogs is hidden but has non-default value
;PM_gwSUBSCRIBEDialogs is hidden but has non-default value
;PM_gwSBCRegisteredUsers is hidden but has non-default value
;PM_gwSBCMediaLegs is hidden but has non-default value
;PM_gwSBCTranscodingSessions is hidden but has non-default value

```

```
[BSP Params]

PCMLawSelect = 3
INIFileVersion = 20908
UdpPortSpacing = 10
EnterCpuOverloadPercent = 99
ExitCpuOverloadPercent = 95

[Analog Params]

[ControlProtocols Params]

AdminStateLockControl = 0

[MGCP Params]

[MEGACO Params]

EP_Num_0 = 0
EP_Num_1 = 1
EP_Num_2 = 1
EP_Num_3 = 0
EP_Num_4 = 0

[PSTN Params]

[SS7 Params]

[Voice Engine Params]

FaxRelayMaxRate = 3
FaxRelayECMEnable = 0
NatMode = 0
CallProgressTonesFilename = 'usa_tones_13.dat'

[WEB Params]

LogoWidth = '145'
HTTPSCipherString = 'RC4:EXP'

[SIP Params]

MEDIACHANNELS = 30
GWDEBUGLEVEL = 1
;ISPRACKREQUIRED is hidden but has non-default value
SIPSESSIONEXPIRES = 900
MINSE = 600
ISFAXUSED = 1
ENABLESBCAPPLICATION = 1
MSLDAPPRIMARYKEY = 'telephoneNumber'
ENERGYDETECTORCMD = 587202560
```

```
ANSWERDETECTORCMD = 10486144
;GWAPPCONFIGURATIONVERSION is hidden but has non-default value

[SCTP Params]

[IPsec Params]

[Audio Staging Params]

[SNMP Params]

[ PhysicalPortsTable ]

FORMAT PhysicalPortsTable_Index = PhysicalPortsTable_Port,
PhysicalPortsTable_Mode, PhysicalPortsTable_SpeedDuplex,
PhysicalPortsTable_PortDescription, PhysicalPortsTable_GroupMember,
PhysicalPortsTable_GroupStatus;
PhysicalPortsTable 0 = "GE_4_1", 1, 4, "LAN Port#1", "GROUP_1", "Active";
PhysicalPortsTable 1 = "GE_4_2", 1, 4, "LAN Port#2", "GROUP_1",
"Redundant";
PhysicalPortsTable 2 = "GE_4_3", 1, 4, "WAN Port#1", "GROUP_2", "Active";
PhysicalPortsTable 3 = "GE_4_4", 1, 4, "WAN Port#2", "GROUP_2",
"Redundant";
PhysicalPortsTable 4 = "FE_5_1", 1, 4, "User Port #4", "GROUP_3",
"Active";
PhysicalPortsTable 5 = "FE_5_2", 1, 4, "User Port #5", "GROUP_3",
"Redundant";
PhysicalPortsTable 6 = "FE_5_3", 1, 4, "User Port #6", "GROUP_4",
"Active";
PhysicalPortsTable 7 = "FE_5_4", 1, 4, "User Port #7", "GROUP_4",
"Redundant";
PhysicalPortsTable 8 = "FE_5_5", 1, 4, "User Port #8", "GROUP_5",
"Active";
PhysicalPortsTable 9 = "FE_5_6", 1, 4, "User Port #9", "GROUP_5",
"Redundant";
PhysicalPortsTable 10 = "FE_5_7", 1, 4, "User Port #10", "GROUP_6",
"Active";
PhysicalPortsTable 11 = "FE_5_8", 1, 4, "User Port #11", "GROUP_6",
"Redundant";

[ \PhysicalPortsTable ]

[ EtherGroupTable ]

FORMAT EtherGroupTable_Index = EtherGroupTable_Group,
EtherGroupTable_Mode, EtherGroupTable_Member1, EtherGroupTable_Member2;
EtherGroupTable 0 = "GROUP_1", 2, "GE_4_1", "GE_4_2";
EtherGroupTable 1 = "GROUP_2", 2, "GE_4_3", "GE_4_4";
EtherGroupTable 2 = "GROUP_3", 2, "FE_5_1", "FE_5_2";
EtherGroupTable 3 = "GROUP_4", 2, "FE_5_3", "FE_5_4";
EtherGroupTable 4 = "GROUP_5", 2, "FE_5_5", "FE_5_6";
EtherGroupTable 5 = "GROUP_6", 2, "FE_5_7", "FE_5_8";
EtherGroupTable 6 = "GROUP_7", 0, "", "";
EtherGroupTable 7 = "GROUP_8", 0, "", "";
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EtherGroupTable 8 = "GROUP_9", 0, "", "";
EtherGroupTable 9 = "GROUP_10", 0, "", "";
EtherGroupTable 10 = "GROUP_11", 0, "", "";
EtherGroupTable 11 = "GROUP_12", 0, "", "";

[ \EtherGroupTable ]

[ DeviceTable ]

FORMAT DeviceTable_Index = DeviceTable_VlanID,
DeviceTable_UnderlyingInterface, DeviceTable_DeviceName,
DeviceTable_Tagging;
DeviceTable 0 = 1, "GROUP_1", "NET1_VLAN", 0;
DeviceTable 1 = 2, "GROUP_2", "NET2_VLAN", 0;

[ \DeviceTable ]

[ InterfaceTable ]

FORMAT InterfaceTable_Index = InterfaceTable_ApplicationTypes,
InterfaceTable_InterfaceMode, InterfaceTable_IPAddress,
InterfaceTable_PrefixLength, InterfaceTable_Gateway,
InterfaceTable_InterfaceName, InterfaceTable_PrimaryDNSServerIPAddress,
InterfaceTable_SecondaryDNSServerIPAddress,
InterfaceTable_UnderlyingDevice;
InterfaceTable 0 = 6, 10, 192.168.1.212, 16, 192.168.1.210, "NET1",
192.168.1.201, 0.0.0.0, "NET1_VLAN";
InterfaceTable 1 = 5, 10, 217.33.37.220, 25, 217.33.37.193, "NET2",
8.8.8.8, 0.0.0.0, "NET2_VLAN";

[ \InterfaceTable ]

[ DspTemplates ]

;
; *** TABLE DspTemplates ***
; This table contains hidden elements and will not be exposed.
; This table exists on board and will be saved during restarts.
;

[ \DspTemplates ]

[ WebUsers ]

FORMAT WebUsers_Index = WebUsers_Username, WebUsers_Password,
WebUsers_Status, WebUsers_PwAgeInterval, WebUsers_SessionLimit,
WebUsers_SessionTimeout, WebUsers_BlockTime, WebUsers_UserLevel,
WebUsers_PwNonce;
WebUsers 0 = "Admin",
"$1$!i+jo6+zp86b0paekpqP8/6z9+aqqr5OQkMCRwZWSm8/Jm5XMn8qD1dCKhoKPhNvdiovYj
YqIpfjx86Kh8fSqq/8=", 1, 0, 2, 15, 60, 200,
"4d7d628b8d92881b3e517bdaedec492a";
WebUsers 1 = "User",
"$1$RHMmdyR+eSlzeX18KWF1MWNtMDdnPW05b2w4OmlXVAZWAFBXAg1cDglYD1pbQBAXR0JEQ
    
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hFLHUgfSE0fG+OysbQ=", 3, 0, 2, 15, 60, 50,
"5939b7ddb0d3c232369a7525561b87f1";

[ \WebUsers ]

[ TLSContexts ]

FORMAT TLSContexts_Index = TLSContexts_Name, TLSContexts_TLSVersion,
TLSContexts_ServerCipherString, TLSContexts_ClientCipherString,
TLSContexts_OcspEnable, TLSContexts_OcspServerPrimary,
TLSContexts_OcspServerSecondary, TLSContexts_OcspServerPort,
TLSContexts_OcspDefaultResponse;
TLSContexts 0 = "default", 0, "RC4:EXP", "ALL:!ADH", 0, , , 2560, 0;

[ \TLSContexts ]

[ IpProfile ]

FORMAT IpProfile_Index = IpProfile_ProfileName, IpProfile_IpPreference,
IpProfile_CodersGroupID, IpProfile_IsFaxUsed,
IpProfile_JitterBufMinDelay, IpProfile_JitterBufOptFactor,
IpProfile_IPDiffServ, IpProfile_SigIPDiffServ, IpProfile_SCE,
IpProfile_RTPRedundancyDepth, IpProfile_RemoteBaseUDPPort,
IpProfile_CNGmode, IpProfile_VxxTransportType, IpProfile_NSEMode,
IpProfile_IsDTMFUsed, IpProfile_PlayRBTone2IP,
IpProfile_EnableEarlyMedia, IpProfile_ProgressIndicator2IP,
IpProfile_EnableEchoCanceller, IpProfile_CopyDest2RedirectNumber,
IpProfile_MediaSecurityBehaviour, IpProfile_CallLimit,
IpProfile_DisconnectOnBrokenConnection, IpProfile_FirstTxDtmfOption,
IpProfile_SecondTxDtmfOption, IpProfile_RxDTMFOption,
IpProfile_EnableHold, IpProfile_InputGain, IpProfile_VoiceVolume,
IpProfile_AddIEInSetup, IpProfile_SBCExtensionCodersGroupID,
IpProfile_MediaIPVersionPreference, IpProfile_TranscodingMode,
IpProfile_SBCAllowedMediaTypes, IpProfile_SBCAllowedCodersGroupID,
IpProfile_SBCAllowedVideoCodersGroupID, IpProfile_SBCAllowedCodersMode,
IpProfile_SBCMediaSecurityBehaviour, IpProfile_SBCRFC2833Behavior,
IpProfile_SBCAlternativeDTMFMethod, IpProfile_SBCAssertIdentity,
IpProfile_AMDSensitivityParameterSuit, IpProfile_AMDSensitivityLevel,
IpProfile_AMDMaxGreetingTime, IpProfile_AMDMaxPostSilenceGreetingTime,
IpProfile_SBCDiversionsMode, IpProfile_SBCHistoryInfoMode,
IpProfile_EnableQSIGTunneling, IpProfile_SBCFaxCodersGroupID,
IpProfile_SBCFaxBehavior, IpProfile_SBCFaxOfferMode,
IpProfile_SBCFaxAnswerMode, IpProfile_SbcPrackMode,
IpProfile_SBCSessionExpiresMode, IpProfile_SBCRemoteUpdateSupport,
IpProfile_SBCRemoteReinviteSupport,
IpProfile_SBCRemoteDelayedOfferSupport, IpProfile_SBCRemoteReferBehavior,
IpProfile_SBCRemote3xxBehavior, IpProfile_SBCRemoteMultiple18xSupport,
IpProfile_SBCRemoteEarlyMediaResponseType,
IpProfile_SBCRemoteEarlyMediaSupport, IpProfile_EnableSymmetricMKI,
IpProfile_MKISize, IpProfile_SBCEnforceMKISize,
IpProfile_SBCRemoteEarlyMediaRTP, IpProfile_SBCRemoteSupportsRFC3960,
IpProfile_SBCRemoteCanPlayRingback, IpProfile_EnableEarlyl83,
IpProfile_EarlyAnswerTimeout, IpProfile_SBC2833DTMFPayloadType,
IpProfile_SBCUserRegistrationTime, IpProfile_ResetSRTPStateUponRekey,
IpProfile_AmdMode, IpProfile_SBCReliableHeldToneSource,
IpProfile_GenerateSRTPKeys, IpProfile_SBCPlayHeldTone,
IpProfile_SBCRemoteHoldFormat, IpProfile_SBCRemoteReplacesBehavior,
IpProfile_SBCSDPptimeAnswer, IpProfile_SBCPreferredPTime,
IpProfile_SBCUseSilenceSupp, IpProfile_SBCRTPRedundancyBehavior,
IpProfile_SBCPlayRBTToTransferee, IpProfile_SBCRTCPMode,
IpProfile_SBCJitterCompensation,
IpProfile_SBCRemoteRenegotiateOnFaxDetection,

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IpProfile_JitterBufMaxDelay,
IpProfile_SBCUserBehindUdpNATRegistrationTime,
IpProfile_SBCUserBehindTcpNATRegistrationTime,
IpProfile_SBCSDPHandleRTCPAttribute,
IpProfile_SBCRemoveCryptoLifetimeInSDP, IpProfile_SBCIceMode,
IpProfile_SBCRTCPMux, IpProfile_SBCMediaSecurityMethod,
IpProfile_SBCHandleXDetect, IpProfile_SBCRTCPFeedback,
IpProfile_SBCRemoteRepresentationMode, IpProfile_SBCKeepVIAHeaders,
IpProfile_SBCKeepRoutingHeaders, IpProfile_SBCKeepUserAgentHeader,
IpProfile_SBCRemoteMultipleEarlyDialogs,
IpProfile_SBCRemoteMultipleAnswersMode, IpProfile_SBCDirectMediaTag,
IpProfile_SBCAdaptRFC2833BWTToVoiceCoderBW;
IpProfile 1 = "CIC", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 2, 0,
0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 3, 0, 2, 1, 0, 0, 1, 0, 1,
0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 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;
IpProfile 2 = "GSM GW", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0,
0, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 2,
0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1, 0,
1, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 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;
IpProfile 3 = "VoIPTalk", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0,
0, 0, -1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0,
0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0, 1,
0, 1, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 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;
IpProfile 4 = "Remote Users", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0,
0, 0, 0, -1, 1, 0, 0, -1, 0, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1,
0, 2, 0, 0, 0, 0, 8, 300, 400, 0, 0, 0, -1, 0, 0, 1, 3, 0, 2, 2, 1, 0, 0,
1, 0, 1, 0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 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;
IpProfile 5 = "BT", 1, 0, 0, 10, 10, 46, 40, 0, 0, 0, 0, 2, 0, 0, 0, 0,
-1, 1, 0, 0, -1, 1, 4, -1, 1, 1, 0, 0, "", -1, 0, 0, "", -1, -1, 0, 0, 0,
0, 0, 0, 8, 300, 400, 0, 0, 0, 1, 0, 0, 1, 3, 3, 2, 2, 1, 0, 0, 1, 0, 1,
0, 0, 0, 0, 1, 0, 0, 0, 0, 0, 0, 1, 0, 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;

[ \IpProfile ]

[ CpMediaRealm ]

FORMAT CpMediaRealm_Index = CpMediaRealm_MediaRealmName,
CpMediaRealm_IPv4IF, CpMediaRealm_IPv6IF, CpMediaRealm_PortRangeStart,
CpMediaRealm_MediaSessionLeg, CpMediaRealm_PortRangeEnd,
CpMediaRealm_IsDefault, CpMediaRealm_QoeProfile, CpMediaRealm_BWProfile;
CpMediaRealm 0 = "realm0", "NET1", "", 6000, 100, 6990, 0, "", "";
CpMediaRealm 1 = "realm1", "NET2", "", 7000, 100, 7990, 0, "", "";
CpMediaRealm 2 = "realmFEU", "NET2", "", 9000, 100, 9990, 0, "", "";

[ \CpMediaRealm ]

[ SBCRoutingPolicy ]

FORMAT SBCRoutingPolicy_Index = SBCRoutingPolicy_Name,
SBCRoutingPolicy_LCREnable, SBCRoutingPolicy_LCRAverageCallLength,
SBCRoutingPolicy_LCRDefaultCost, SBCRoutingPolicy_LdapServerGroupName;
SBCRoutingPolicy 0 = "Default_SBCRoutingPolicy", 0, 1, 0, "";
    
```



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[ \SBCRoutingPolicy ]

[ SRD ]

FORMAT SRD_Index = SRD_Name, SRD_BlockUnRegUsers, SRD_MaxNumOfRegUsers,
SRD_EnableUnAuthenticatedRegistrations, SRD_SharingPolicy,
SRD_UsedByRoutingServer, SRD_SBCOperationMode,
SRD_SBCRegisteredUsersClassificationMethod, SRD_SBCRoutingPolicyName;
SRD 0 = "DefaultSRD", 0, -1, 1, 0, 0, 0, -1, "Default_SBCRoutingPolicy";

[ \SRD ]

[ SIPInterface ]

FORMAT SIPInterface_Index = SIPInterface_InterfaceName,
SIPInterface_NetworkInterface, SIPInterface_ApplicationType,
SIPInterface_UDPPort, SIPInterface_TCPSPort, SIPInterface_TLSPort,
SIPInterface_SRDName, SIPInterface_MessagePolicyName,
SIPInterface_TLSContext, SIPInterface_TLSMutualAuthentication,
SIPInterface_TCPKeepaliveEnable,
SIPInterface_ClassificationFailureResponseType,
SIPInterface_PreClassificationManSet, SIPInterface_EncapsulatingProtocol,
SIPInterface_MediaRealm, SIPInterface_SBCDirectMedia,
SIPInterface_BlockUnRegUsers, SIPInterface_MaxNumOfRegUsers,
SIPInterface_EnableUnAuthenticatedRegistrations,
SIPInterface_UsedByRoutingServer;
SIPInterface 1 = "SIPInterface_1", "NET1", 2, 0, 5060, 0, "DefaultSRD",
", "default", -1, 0, 500, -1, 0, "realm0", 0, -1, -1, -1, 0;
SIPInterface 2 = "SIPInterface_2", "NET2", 2, 5060, 0, 0, "DefaultSRD",
", "default", -1, 0, 500, -1, 0, "realm1", 0, -1, -1, -1, 0;
SIPInterface 4 = "SIPInterface_4", "NET2", 2, 5070, 0, 0, "DefaultSRD",
", "default", -1, 0, 500, -1, 0, "realmFEU", 0, -1, -1, -1, 0;

[ \SIPInterface ]

[ ProxySet ]

FORMAT ProxySet_Index = ProxySet_ProxyName,
ProxySet_EnableProxyKeepAlive, ProxySet_ProxyKeepAliveTime,
ProxySet_ProxyLoadBalancingMethod, ProxySet_IsProxyHotSwap,
ProxySet_SRDName, ProxySet_ClassificationInput, ProxySet_TLSContextName,
ProxySet_ProxyRedundancyMode, ProxySet_DNSResolveMethod,
ProxySet_KeepAliveFailureResp, ProxySet_GWIPv4SIPInterfaceName,
ProxySet_SBCIPv4SIPInterfaceName, ProxySet_SASIPv4SIPInterfaceName,
ProxySet_GWIPv6SIPInterfaceName, ProxySet_SBCIPv6SIPInterfaceName,
ProxySet_SASIPv6SIPInterfaceName;
ProxySet 1 = "ProxySet_1", 0, 60, 0, 1, "DefaultSRD", 0, "", 1, -1, "",
", "SIPInterface_1", "", "", "", "";
ProxySet 2 = "ProxySet_2", 0, 60, 0, 1, "DefaultSRD", 0, "", 1, -1, "",
", "SIPInterface_2", "", "", "", "";
ProxySet 3 = "ProxySet_3", 0, 60, 0, 0, "DefaultSRD", 0, "", 1, -1, "",
", "SIPInterface_2", "", "", "", "";
ProxySet 4 = "ProxySet_4", 0, 60, 0, 0, "DefaultSRD", 0, "", -1, -1, "",
", "SIPInterface_1", "", "", "", "";
ProxySet 5 = "BT SIP SBC", 1, 60, 0, 1, "DefaultSRD", 0, "", 1, -1, "",
", "SIPInterface_2", "", "", "", "";

[ \ProxySet ]

```

```

[ IPGroup ]

FORMAT IPGroup_Index = IPGroup_Type, IPGroup_Name, IPGroup_ProxySetName,
IPGroup_SIPGroupName, IPGroup_ContactUser, IPGroup_SipReRoutingMode,
IPGroup_AlwaysUseRouteTable, IPGroup_SRDName, IPGroup_MediaRealm,
IPGroup_ClassifyByProxySet, IPGroup_ProfileName,
IPGroup_MaxNumOfRegUsers, IPGroup_InboundManSet, IPGroup_OutboundManSet,
IPGroup_RegistrationMode, IPGroup_AuthenticationMode, IPGroup_MethodList,
IPGroup_EnableSBCCClientForking, IPGroup_SourceUriInput,
IPGroup_DestUriInput, IPGroup_ContactName, IPGroup_Username,
IPGroup_Password, IPGroup_UIFormat, IPGroup_QOEProfile,
IPGroup_BWProfile, IPGroup_MediaEnhancementProfile,
IPGroup_AlwaysUseSourceAddr, IPGroup_MsgManUserDef1,
IPGroup_MsgManUserDef2, IPGroup_SIPConnect, IPGroup_SBCPSAPMode,
IPGroup_DTLSContext, IPGroup_CreatedByRoutingServer,
IPGroup_UsedByRoutingServer, IPGroup_SBCOperationMode,
IPGroup_SBCRouteUsingRequestURIPort;

IPGroup 1 = 0, "CIC", "ProxySet_1", "217.33.37.220", "", -1, 0,
"DefaultSRD", "realm0", 1, "CIC", -1, -1, -1, 0, 0, "", 0, -1, -1, "",
"", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;

IPGroup 2 = 0, "GSM GW", "ProxySet_2", "WAN", "", -1, 0, "DefaultSRD",
"realm1", 1, "GSM GW", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "",
"$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;

IPGroup 3 = 0, "VoipTalk", "ProxySet_3", "voiptalk.org", "voiptalk.org",
-1, 0, "DefaultSRD", "realm1", 1, "VoIPTalk", -1, -1, -1, 0, 0, "", 0, -
1, -1, "", "", "$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1,
0;

IPGroup 4 = 1, "Remote Users", "", "", "", -1, 0, "DefaultSRD",
"realmFEU", 0, "Remote Users", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "",
"$1$gQ==", 0, "", "1", "", "", 0, 0, "", 0, 0, -1, 0;

IPGroup 5 = 0, "BT", "BT SIP SBC", "192.65.221.26", "", -1, 0,
"DefaultSRD", "realm1", 1, "BT", -1, -1, -1, 0, 0, "", 0, -1, -1, "", "",
"$1$gQ==", 0, "", "", "", 0, "", "", 0, 0, "", 0, 0, -1, 0;

[ \IPGroup ]

[ ProxyIp ]

FORMAT ProxyIp_Index = ProxyIp_ProxySetId, ProxyIp_ProxyIpIndex,
ProxyIp_IpAddress, ProxyIp_TransportType;
ProxyIp 0 = "4", 1, "192.168.1.202:8060", 1;
ProxyIp 1 = "1", 0, "192.168.1.202:5060", 1;
ProxyIp 2 = "5", 1, "192.65.221.5", 0;
ProxyIp 3 = "2", 0, "192.168.1.211:5060", 0;
ProxyIp 4 = "3", 4, "77.240.48.94", 0;

[ \ProxyIp ]

[ Account ]

FORMAT Account_Index = Account_ServedTrunkGroup,
Account_ServedIPGroupName, Account_ServingIPGroupName, Account_Username,
Account_Password, Account_HostName, Account_Register,
Account_ContactUser, Account_ApplicationType;
Account 1 = -1, "CIC", "VoipTalk", "844278727", "$1$QBQgKy8mPBZ0PCw=",
"voiptalk.org", 0, "844278727", 2;
    
```

```

[ \Account ]

[ IP2IPRouting ]

FORMAT IP2IPRouting_Index = IP2IPRouting_RouteName,
IP2IPRouting_RoutingPolicyName, IP2IPRouting_SrcIPGroupName,
IP2IPRouting_SrcUsernamePrefix, IP2IPRouting_SrcHost,
IP2IPRouting_DestUsernamePrefix, IP2IPRouting_DestHost,
IP2IPRouting_RequestType, IP2IPRouting_MessageConditionName,
IP2IPRouting_ReRouteIPGroupName, IP2IPRouting_Trigger,
IP2IPRouting_CallSetupRulesSetId, IP2IPRouting_DestType,
IP2IPRouting_DestIPGroupName, IP2IPRouting_DestSIPInterfaceName,
IP2IPRouting_DestAddress, IP2IPRouting_DestPort,
IP2IPRouting_DestTransportType, IP2IPRouting_AltRouteOptions,
IP2IPRouting_GroupPolicy, IP2IPRouting_CostGroup;
IP2IPRouting 0 = "Terminate OPTIONS", "Default_SBCRoutingPolicy", "Any",
"*, *, *, *, 6, "", "Any", 0, -1, 1, "", "", "internal", 0, -1, 0,
0, "";
IP2IPRouting 1 = "CIC to BT", "Default_SBCRoutingPolicy", "CIC", "*",
"*, *, *, 0, "", "Any", 0, -1, 0, "BT", "SIPInterface_2", "", 0, -1,
0, 0, "";
IP2IPRouting 2 = "BT to CIC", "Default_SBCRoutingPolicy", "BT", "*", "*",
"*, *, 0, "", "Any", 0, -1, 0, "CIC", "SIPInterface_1", "", 0, -1, 0,
0, "";

[ \IP2IPRouting ]

[ Classification ]

FORMAT Classification_Index = Classification_ClassificationName,
Classification_MessageConditionName, Classification_SRDName,
Classification_SrcSIPInterfaceName, Classification_SrcAddress,
Classification_SrcPort, Classification_SrcTransportType,
Classification_SrcUsernamePrefix, Classification_SrcHost,
Classification_DestUsernamePrefix, Classification_DestHost,
Classification_ActionType, Classification_SrcIPGroupName,
Classification_DestRoutingPolicy, Classification_IpProfileName;
Classification 0 = "Allow remote users", "", "DefaultSRD",
"SIPInterface_4", "", 0, 0, "*", "*", "*", "*", 1, "Remote Users", "",
"Remote Users";
Classification 1 = "Allow CIC access", "", "DefaultSRD",
"SIPInterface_1", "", 0, -1, "*", "*", "*", "*", 1, "CIC",
"Default_SBCRoutingPolicy", "CIC";
Classification 2 = "Allow VoIPTalk Access", "", "DefaultSRD",
"SIPInterface_2", "", 0, 0, "*", "*", "*", "*", 1, "VoipTalk", "",
"VoIPTalk";
Classification 3 = "Allow BT Access IP1", "", "DefaultSRD",
"SIPInterface_2", "192.65.221.23", 5060, 0, "*", "*", "*", "*", 1, "BT",
"Default_SBCRoutingPolicy", "BT";
Classification 4 = "Allow BT Access IP2", "", "DefaultSRD",
"SIPInterface_2", "192.65.221.26", 5060, 0, "*", "*", "*", "*", 1, "BT",
"Default_SBCRoutingPolicy", "";

[ \Classification ]

[ CodersGroup0 ]

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```
FORMAT CodersGroup0_Index = CodersGroup0_Name, CodersGroup0_pTime,
CodersGroup0_rate, CodersGroup0_PayloadType, CodersGroup0_Sce,
CodersGroup0_CoderSpecific;
CodersGroup0 0 = "g711Alaw64k", 20, 0, -1, 0, "";
CodersGroup0 1 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup0 2 = "g729", 20, 0, -1, 0, "";
CodersGroup0 3 = "t38fax", 255, 255, -1, 255, "";

[ \CodersGroup0 ]

[ CodersGroup1 ]

FORMAT CodersGroup1_Index = CodersGroup1_Name, CodersGroup1_pTime,
CodersGroup1_rate, CodersGroup1_PayloadType, CodersGroup1_Sce,
CodersGroup1_CoderSpecific;
CodersGroup1 0 = "g711Alaw64k", 20, 0, -1, 0, "";
CodersGroup1 1 = "g711Ulaw64k", 20, 0, -1, 0, "";
CodersGroup1 2 = "g729", 20, 0, -1, 0, "";
CodersGroup1 3 = "t38fax", 255, 255, -1, 255, "";

[ \CodersGroup1 ]

[ GwRoutingPolicy ]

FORMAT GwRoutingPolicy_Index = GwRoutingPolicy_Name,
GwRoutingPolicy_LCREnable, GwRoutingPolicy_LCRAverageCallLength,
GwRoutingPolicy_LCRDefaultCost, GwRoutingPolicy_LdapServerGroupName;
GwRoutingPolicy 0 = "GwRoutingPolicy", 0, 1, 0, "";

[ \GwRoutingPolicy ]

[ ResourcePriorityNetworkDomains ]

FORMAT ResourcePriorityNetworkDomains_Index =
ResourcePriorityNetworkDomains_Name,
ResourcePriorityNetworkDomains_Ip2TelInterworking;
ResourcePriorityNetworkDomains 1 = "dsn", 1;
ResourcePriorityNetworkDomains 2 = "dod", 1;
ResourcePriorityNetworkDomains 3 = "drsn", 1;
ResourcePriorityNetworkDomains 5 = "uc", 1;
ResourcePriorityNetworkDomains 7 = "cuc", 1;

[ \ResourcePriorityNetworkDomains ]
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