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# Ribbon SBC 1000/2000 V9.0.7 IOT Vodafone SIP Trunk: Interoperability Guide

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## Interoperable Vendors

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# Copyright

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## Document Overview

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This document outlines Avaya's configuration best practices involving Ribbon SBC 1000/2000 when deployed with Cisco Unified Communication Manager 12.5. This document also provides the configuration snapshot of the interoperability performed between Ribbon 1000/2000 and Cisco Unified Communication Manager 12.5.

## Scope

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This document provides configuration best practices for deploying Ribbon's SBC 1000/2000 Cisco Unified Communication Manager 12.5. These are configuration best practices and customers may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design which best meets their requirements.

## Non-Goals

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This document is provided as a starting point to build the SBC configurations in consultation with network design and deployment engineers. It is not intended to provide detailed configurations that will meet the requirements of every customer.

## Audience

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This technical document is intended for telecommunications engineers with the purpose of configuring the Ribbon 1000/2000 with the Cisco Unified Communication Manager 12.5.

Steps will require navigating the third-party product as well as the Ribbon product using graphical user interface (GUI) or command line interface (CLI). An understanding of the basic concepts of TCP/UDP/TLS, IP/Routing, and SIP/RTP/SRTP is needed to complete the configuration and any necessary troubleshooting.

## Pre-Requisites

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The interoperability test described in this document requires no special licensing.

## Product and Device Details

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	Equipment/Product	Software Version
<b>Ribbon Communications</b>	SBC 2000	V9.0.7
<b>Third-Party Products</b>	Cisco Unified Communication Manager	V12.5
	Cisco IP Communicator	V8.6.2.0
	MicroSIP	V3.20.7

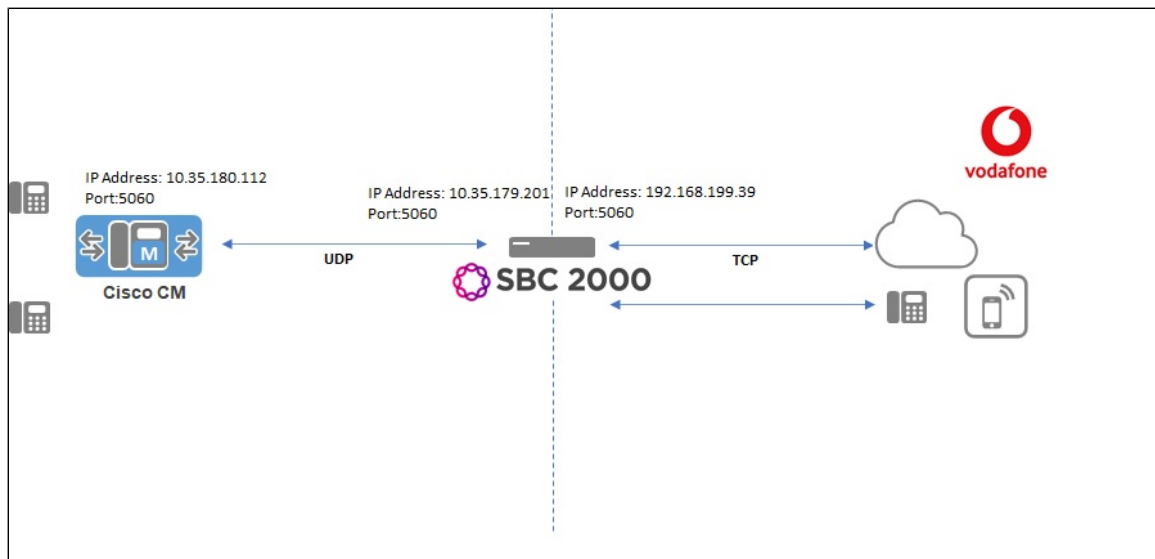
## Network Topology Diagram

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### Deployment Topology

The deployment topology diagram is depicted below.

**Figure 1:** Deployment Topology



## Section-A: SBC 1000/2000 Configuration

The following configuration steps provide an example of how to configure the Ribbon SBC 1000/2000 to interoperate with the Cisco Unified Communication Manager and Vodafone SIP Trunk:

1. [SIP Profile](#)
2. [SIP Server](#)
3. [Media System](#)
4. [Media Profiles](#)
5. [Media List](#)
6. [Signaling Groups](#)
7. [Transformation](#)
8. [Call Routing Table](#)

### 1. SIP Profile

SIP Profiles control how the Ribbon SBC 1000/2000 communicates with SIP devices. The profiles control important characteristics such as session timers, SIP Header customization, SIP timers, MIME payloads, and option tags.

Select **Settings > SIP > SIP Profiles** to access the SIP Profile screen.

The following figures show the default SIP profile used for this configuration of the Ribbon 1000/2000.

**Figure 2:** Vodafone SIP Profile

▼
Vodafone SIP Profile

Description:

**Session Timer**  
 Session Timer:

**Header Customization**  
 FQDN in From Header:   
 FQDN in Contact Header:   
 Send Assert Header:   
 SBC Edge Diagnostics Header:   
 Trusted Interface:   
 UA Header:   
 Calling Info Source:   
 Diversion Header Selection:   
 Record Route Header:

**MIME Payloads**  
 ELIN Identifier:   
 PIDF-LO Passthrough:   
 Unknown Subtype Passthrough:

**Options Tags**  
 100rel:   
 Path:   
 Update:

**Timers**  
 Transport Timeout Timer:  ms (5000..32000)  
 Maximum Retransmissions:   
 Redundancy Retry Timer:  ms (5000..180000)

---

**RFC Timers**  
 Timer T1:  ms (100..10000)  
 Timer T2:  ms (1000..80000)(> = T1)  
 Timer T4:  ms (1000..100000)  
 Timer D:  ms (5000..640000)  
 Timer B: 32000 ms  
 Timer F: 32000 ms  
 Timer H: 32000 ms (64\*TimerT1)  
 Timer J:  ms (4000..640000)

**SDP Customization**  
 Send Number of Audio Channels:   
 Connection Info in Media Section:   
 Origin Field Username:  default: SBC  
 Session Name:  default: VoipCall  
 Digit Transmission Preference:   
 SDP Handling Preference:

Figure 3: CUCM SIP Profile

CUCM 12.0.1

<p><b>Session Timer</b></p> <p>Session Timer: <input type="text" value="Enable"/> <input type="button" value="v"/></p> <p>Minimum Acceptable Timer: <input type="text" value="600"/> * secs [90..7200]</p> <p>Offered Session Timer: <input type="text" value="3600"/> * secs [90..7200]</p> <p>Terminate On Refresh Failure: <input type="text" value="False"/> <input type="button" value="v"/></p>	<p><b>MIME Payloads</b></p> <p>ELIN Identifier: <input type="text" value="LOC"/> <input type="button" value="v"/></p> <p>PIDF-LO Passthrough: <input type="text" value="Enable"/> <input type="button" value="v"/></p> <p>Unknown Subtype Passthrough: <input type="text" value="Disable"/> <input type="button" value="v"/></p>
<p><b>Header Customization</b></p> <p>FQDN in From Header: <input type="text" value="Disable"/> <input type="button" value="v"/></p> <p>FQDN in Contact Header: <input type="text" value="Disable"/> <input type="button" value="v"/></p> <p>Send Assert Header: <input type="text" value="Trusted Only"/> <input type="button" value="v"/></p> <p>SBC Edge Diagnostics Header: <input type="text" value="Enable"/> <input type="button" value="v"/></p> <p>Trusted Interface: <input type="text" value="Enable"/> <input type="button" value="v"/></p> <p>UA Header: <input type="text" value="Ribbon SBC Edge"/></p> <p>Calling Info Source: <input type="text" value="RFC Standard"/> <input type="button" value="v"/></p> <p>Diversion Header Selection: <input type="text" value="Last"/> <input type="button" value="v"/></p> <p>Record Route Header: <input type="text" value="RFC 3261 Standard"/> <input type="button" value="v"/></p>	<p><b>Options Tags</b></p> <p>100rel: <input type="text" value="Supported"/> <input type="button" value="v"/></p> <p>Path: <input type="text" value="Not Present"/> <input type="button" value="v"/></p> <p>Timer: <input type="text" value="Supported"/> <input type="button" value="v"/></p> <p>Update: <input type="text" value="Supported"/> <input type="button" value="v"/></p>
<p><b>Timers</b></p> <p>Transport Timeout Timer: <input type="text" value="5000"/> ms [5000..32000]</p> <p>Maximum Retransmissions: <input type="text" value="RFC Stand"/> <input type="button" value="v"/></p> <p>Redundancy Retry Timer: <input type="text" value="180000"/> ms [5000..180000]</p> <p><b>RFC Timers</b></p> <p>Timer T1: <input type="text" value="500"/> ms [100..10000]</p> <p>Timer T2: <input type="text" value="4000"/> ms [1000..80000](&gt;= T1)</p> <p>Timer T4: <input type="text" value="5000"/> ms [1000..100000]</p> <p>Timer D: <input type="text" value="32000"/> ms [5000..640000]</p> <p>Timer B: 32000 ms</p> <p>Timer F: 32000 ms</p> <p>Timer H: 32000 ms (64*TimerT1)</p> <p>Timer J: <input type="text" value="4000"/> ms [4000..640000]</p>	<p><b>SDP Customization</b></p> <p>Send Number of Audio Channels: <input type="text" value="False"/> <input type="button" value="v"/></p> <p>Connection Info in Media Section: <input type="text" value="True"/> <input type="button" value="v"/></p> <p>Origin Field Username: <input type="text" value="SBC"/> <small>default: SBC</small></p> <p>Session Name: <input type="text" value="VoipCall"/> <small>default: VoipCall</small></p> <p>Digit Transmission Preference: <input type="text" value="RFC 2833/Voic"/> <input type="button" value="v"/></p> <p>SDP Handling Preference: <input type="text" value="Legacy Audio/f"/> <input type="button" value="v"/></p>

## 2. SIP Server

SIP Server Tables contain information about the SIP devices connected to the Ribbon SBC 1000/2000.

Select **Settings > SIP > SIP Server Tables** to access the SIP Server Tables screen.

The entries in the tables provide information about the IP addresses, ports, and protocols used to communicate with each SIP server. The entries also contain links to counters that are useful for troubleshooting, as shown in the following figures.

**Figure 4:** Vodafone SIP Servers

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	None
Priority	1		
Host FQDN/IP	X.X.X.X *		
Host IP Version	IPv4		
Port	5060 * [1..65535]		
Protocol	TCP *		
Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None +	Reuse	True
Contact Registrant Table	None +	Sockets	4
Session URI Validation	Liberal	Reuse Timeout	Forever
<b>Apply</b>			

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	None
Priority	1		
Host FQDN/IP	X.X.X.X *		
Host IP Version	IPv4		
Port	5060 * [1..65535]		
Protocol	TCP *		
Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None +	Reuse	True
Contact Registrant Table	None +	Sockets	4
Session URI Validation	Liberal	Reuse Timeout	Forever
<b>Apply</b>			

Figure 5: CUCM SIP Servers

### Server Host

Server Lookup **IP/FQDN**

Priority

Host FQDN/IP  \*

Port  \* [1..65535]

Protocol  \*

### Transport

Monitor

### Remote Authorization and Contacts

Remote Authorization Table  +

Contact Registrant Table  +

Session URI Validation

### 3. Media System

The Media System Configuration contains system-wide settings for the Media System. Configuring the media system means setting the number of RTP/RTCP port pairs and the starting port.

Select **Settings > Media > Media System Configuration** to access the Media System configuration screen.

Figure 6: Media System

### Media System Configuration

Upload Music File

#### Port Range

Start Port  \* [1024..32767]

Number of Port Pairs  \* [1..4800]

Regular Call Media Port Range **16384-17583**

ICE Call Media Port Range **Not activated**

#### Music on Hold

Music on Hold Source

Current Music File **Not Installed**

Echo Canceller Type Option

Echo Cancel NLP Option

Send STUN Packets

### 4. Media Profiles

Media Profiles specify the individual voice and fax compression codecs and their associated settings for inclusion into a Media List. Different codecs provide varying levels of compression, allowing the reduction of bandwidth requirements at the expense of voice quality.

Select **Settings > Media > Media Profiles**.

The following figures illustrate possible media profiles of the voice codecs used for the SBC 1000/2000. The examples are for reference only.

Figure 7: Vodafone Media Profile



**Voice Codec Configuration**

Description:

Codec:

Payload Size:  ms

**Voice Codec Configuration**

Description:

Codec:

Payload Size:  ms

**Figure 8:** CUCM Media Profile

**Voice Codec Configuration**

Description:

Codec:

Payload Size:  ms

**Voice Codec Configuration**

Description:

Codec:

Payload Size:  ms

## 5. Media List

The Media List shows the selected voice and fax compression codecs and their associated settings.

Select **Settings > Media > Media List** to access the Media List configuration screen.

**Figure 9:** Vodafone Media List

Description:

Media Profiles List:
 

- Vodafone G711A
- Vodafone G711U

 Up, Down, Add/Edit, Remove

SDES-SRTP Profile:  *Associated SIP SG Listen Ports should be TLS only.* +

DTLS-SRTP Profile:  +

Media DSCP:  \* [0..63]

RTCP Mode:

Dead Call Detection:

Silence Suppression:

Gain Control	Digit Relay
Receive Gain: <input type="text" value="0"/> [-14..+6] dB Transmit Gain: <input type="text" value="0"/> [-14..+6] dB	Digit (DTMF) Relay Type: <input type="text" value="RFC 2833"/> Digit Relay Payload Type: <input type="text" value="101"/> [96..127]

**Passthrough/Tone Detection**

Modem Passthrough:

Fax Passthrough:

CNG Tone Detection:

Fax Tone Detection:

DTMF Signal to Noise:  [-3..+6] dB

DTMF Minimum Level:  [-48..-14] dBm0

Figure 10: CUCM Media List

Description:

Media Profiles List:

- CUCM G711A
- CUCM G.711 U

Up, Down, Add/Edit, Remove

SDES-SRTP Profile:  Associated SIP SG Listen Ports should be TLS only. +

DTLS-SRTP Profile:  +

Media DSCP:  \* [0..63]

RTCP Mode:

Dead Call Detection:

Silence Suppression:

**Gain Control**

Receive Gain:  [-14..+6] dB

Transmit Gain:  [-14..+6] dB

**Digit Relay**

Digit (DTMF) Relay Type:

Digit Relay Payload Type:  [96..127]

**Passthrough/Tone Detection**

Modem Passthrough:

Fax Passthrough:

CNG Tone Detection:

Fax Tone Detection:

DTMF Signal to Noise:  [-3..+6] dB

DTMF Minimum Level:  [-48..-14] dBm0

## 6. Signaling Groups

Signaling Groups allow telephony channels to be grouped together for the purposes of routing and shared configurations. These groups are location where calls are routed to, as well as the location from which Call Routes are selected. Tone Tables and Action Sets are also selected here. In the case of SIP, Signaling Groups will specify protocol settings and links to server, media and mapping tables.

Select **Settings > Signaling Groups** to access the Signaling Groups configuration screens.

**Figure 11:** Vodafone Media Signaling Group

Description:

Admin State:

Service Status: Up

---

#### SIP Channels and Routing

Action Set Table:  +

Call Routing Table:  +

No. of Channels:  \* [1..960]

SIP Profile:  +

SIP Mode:

Agent Type:

#### Media Information

Supported Audio/Fax Modes:   
  
  \*

Supported Video/Application Modes:

Media List ID:  +

---

Interop Mode:

SIP Server Table:  +

Load Balancing:

Channel Hunting:

Notify Lync CAC Profile:

Challenge Request:

Outbound Proxy IP/FQDN:

Outbound Proxy Port:  [1..65535]

No Channel Available Override:

Call Setup Response Timer:  [180..750] secs

Call Proceeding Timer:  [24..750] secs

Play Ringback:

Tone Table:  +

Play Congestion Tone:

Early 183:

Allow Refresh SDP:

Music on Hold:

RTCP Multiplexing:

Media Codec Latch:

#### Mapping Tables

---

QoE Reporting:

Use Register as Keep Alive:

Forked Call Answered Too Soon:

SIP To Q.850 Override Table:  +

Q.850 To SIP Override Table:  +

Pass-thru Peer SIP Response Code:

---

#### SIP IP Details

Teams Local Media Optimization:

Signaling/Media Source IP:

Signaling DSCP:  \* [0..63]

---

#### NAT Traversal

ICE Support:

---

#### Static NAT - Outbound

Outbound NAT Traversal:

---

#### Static NAT - Inbound

Detection:

---

#### Listen Ports

Total 1 SIP Listen Port Row

	Port	Protocol	TLS Profile ID
<input type="checkbox"/>	5060	TCP	N/A

#### Federated IP/FQDN

Total 2 SIP Federated IP Rows

	IP/FQDN	Netmask/Prefix
<input type="checkbox"/>	[REDACTED]	255.255.255.255
<input type="checkbox"/>	[REDACTED]	255.255.255.255

Figure 12: CUCM Media Signaling Group

Description:

Admin State:

Service Status: Up

---

**SIP Channels and Routing**

Action Set Table:  +

Call Routing Table:  +

No. of Channels:  \* [1..960]

SIP Profile:  +

SIP Mode:

Agent Type:

**Media Information**

Supported Audio/Fax Modes:  Add/Edit Remove

Supported Video/Application Modes:

Media List ID:  +

---

Interop Mode:

SIP Server Table:  +

Load Balancing:

Channel Hunting:

Notify Lync CAC Profile:

Challenge Request:

Outbound Proxy IP/FQDN:

Outbound Proxy Port:  [1..65535]

No Channel Available Override:

Play Ringback:

Tone Table:  +

Play Congestion Tone:

Early 183:

Allow Refresh SDP:

Music on Hold:

RTCP Multiplexing:

Media Codec Latch:

---

Call Setup Response Timer:  [180..750] secs

Call Proceeding Timer:  [24..750] secs

QoE Reporting:

Use Register as Keep Alive:

Forked Call Answered Too Soon:

**Mapping Tables**

SIP To Q.850 Override Table:  +

Q.850 To SIP Override Table:  +

Pass-thru Peer SIP Response Code:

---

**SIP IP Details**

Teams Local Media Optimization:

Signaling/Media Source IP:

Signaling DSCP:  \* [0..63]

**NAT Traversal**

ICE Support:

**Static NAT - Outbound**

Outbound NAT Traversal:

**Static NAT - Inbound**

Detection:

---

**Listen Ports**

Total 2 SIP Listen Port Rows

Port	Protocol	TLS Profile ID
<input type="checkbox"/> 5060	UDP	N/A
<input type="checkbox"/> 5060	TCP	N/A

**Federated IP/FQDN**

Total 1 SIP Federated IP Row

IP/FQDN	Netmask/Prefix
<input type="checkbox"/> 10.35.180.112	255.255.255.255

## 7. Transformation

Transformation Tables facilitate the conversion of names, numbers and other fields when routing a call. For example, transformations can convert a public PSTN number into a private extension number, or into a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table. In addition, Transformation tables will be configurable as a reusable pool by Action Set Table.

Select **Settings > Transformation** to access the Transformation configuration screen.

**Figure 13:** Vodafone Media Transformation

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="\+{.*}"/>	Value <input type="text" value="\1"/>

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Calling Address/Number"/>	Type <input type="text" value="Calling Address/Number"/>
Value <input type="text" value="\+{.*}"/>	Value <input type="text" value="\1"/>

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Calling Address/Number"/>	Type <input type="text" value="Host Name"/>
Value <input type="text" value="anonymous"/>	Value <input type="text" value="muesx001.ngn.vodafone.de"/>

Figure 14: CUCM Media Transformation

Description

Admin State Enabled ▼

Match Type Optional (Match One) ▼

**Input Field**

Type Called Address/Number ▼

Value

**Output Field**

Type Called Address/Number ▼

Value

Description

Admin State Enabled ▼

Match Type Optional (Match One) ▼

**Input Field**

Type Called Address/Number ▼

Value

**Output Field**

Type Called Address/Number ▼

Value

Description

Admin State Enabled ▼

Match Type Optional (Match One) ▼

**Input Field**

Type Calling Address/Number ▼

Value

**Output Field**

Type Calling Address/Number ▼

Value

## 8. Call Routing Table

Call Routing allows calls to be carried between signaling groups, thus allowing calls to be carried between ports, and between protocols (like ISDN to SIP). Call Routing Tables define routes. The use of Call Routing Tables allows for the flexible configuration of calls to be carried, and also how the calls are translated. These tables are one of the central connection points of the system, linking Transformation Tables, Message translations, Cause Code Reroutes, Media Lists and the three types of Signaling Groups (ISDN, SIP and CAS).

Select **Settings > Call Routing Table** to access the Call Routing Table configuration screen.

**Figure 15:** Vodafone Media Call Routing

**Route Details**

Description

Admin State  ▾

Route Priority  ▾

Call Priority  ▾

Number/Name Transformation Table  ▾ +

Time of Day Restriction  ▾ +

**Destination Information**

Destination Type  ▾

Message Translation Table  ▾ +

Cause Code Reroutes  ▾ +

Cancel Others upon Forwarding  ▾

Fork Call  ▾

Destination Signaling Groups

(SIP) CUCM 12.5

Enable Maximum Call Duration  ▾

**Media**

Audio/Fax Stream Mode  ▾

Video/Application Stream Mode

Media Transcoding  ▾

Media List  ▾ +

**Quality of Service**

Quality Metrics Number of Calls  [1..100]

Quality Metrics Time Before Retry  [1-60] min.

Min. ASR Threshold  % [0..100]

Enable Min MOS Threshold  ▾

Enable Max. R/T Delay  ▾

Max. R/T Delay  ms [1..65535]

Enable Max. Jitter  ▾

Max. Jitter  ms [1..3000]

**Figure 16: CUCM Media Call Routing**



Route Details

Description

Admin State  ▼

Route Priority  ▼

Call Priority  ▼

Number/Name Transformation Table  ▼ +

Time of Day Restriction  ▼ +

---

Destination Information

Destination Type  ▼

Message Translation Table  ▼ +

Cause Code Reroutes  ▼ +

Cancel Others upon Forwarding  ▼

Fork Call  ▼

Destination Signaling Groups

Enable Maximum Call Duration  ▼

---

Media

Audio/Fax Stream Mode  ▼

Video/Application Stream Mode  ▼

Media Transcoding  ▼

Media List  ▼ +

Quality of Service

Quality Metrics Number of Calls  [1..100]

Quality Metrics Time Before Retry  [1-60] min.

Min. ASR Threshold  % [0..100]

Enable Min MOS Threshold  ▼

Enable Max. R/T Delay  ▼

Enable Max. Jitter  ▼

## Section-B: Cisco Unified Communication Manager 12.5 Configuration

### CUCM Configuration

---

The following new configurations are included in this section:

1. [Security Profile](#)
2. [SIP Profile](#)
3. [SIP Trunk](#)
4. [Route Group](#)
5. [Route List](#)
6. [Route Pattern](#)

#### 1. Security Profile

Select **System > Security > SIP Trunk Security Profile**

**Figure 17:** SIP Trunk Security Profile

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

**Status**  
Status: Ready

**SIP Trunk Security Profile Information**

Name\* Non Secure SIP Trunk Profile

Description Non Secure SIP Trunk Profile authenticated by null String

Device Security Mode Non Secure ▾

Incoming Transport Type\* TCP+UDP ▾

Outgoing Transport Type UDP ▾

Enable Digest Authentication

Nonce Validity Time (mins)\* 600

X.509 Subject Name

Incoming Port\* 5060

Enable Application level authorization

Accept presence subscription

Accept out-of-dialog refer\*\*

Accept unsolicited notification

Accept replaces header

Transmit security status

Allow charging header

SIP V.150 Outbound SDP Offer Filtering\* Use Default Filter ▾

Save Delete Copy Reset Apply Config Add New

## 2. SIP Profile

Select **Device > Device Settings > SIP Profile**

**Figure 18:** SIP Profile

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### SIP Profile Configuration

Save Delete Copy Reset Apply Config Add New

**Status**

Status: Ready  
All SIP devices using this profile must be restarted before any changes will take affect.

**SIP Profile Information**

Name\* SIP OPTIONS Profile  
 Description Default SIP Profile  
 Default MTP Telephony Event Payload Type\* 101  
 Early Offer for G.Clear Calls\* Disabled  
 User-Agent and Server header information\* Send Unified CM Version Information as User-Ager  
 Version in User Agent and Server Header\* Major And Minor  
 Dial String Interpretation\* Phone number consists of characters 0-9, \*, #, an  
 Confidential Access Level Headers\* Disabled

Redirect by Application  
 Disable Early Media on 180  
 Outgoing T.38 INVITE include audio mline  
 Offer valid IP and Send/Receive mode only for T.38 Fax Relay  
 Use Fully Qualified Domain Name in SIP Requests  
 Assured Services SIP conformance  
 Enable External QoS\*\*

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites\* TIAS and AS  
 SDP Transparency Profile Pass all unknown SDP attributes  
 Accept Audio Codec Preferences in Received Offer\* Default

Require SDP Inactive Exchange for Mid-Call Media Change  
 Allow RR/RS bandwidth modifier (RFC 3556)

Figure 19: SIP Profile1

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### SIP Profile Configuration

Save 
 Delete 
 Copy 
 Reset 
 Apply Config 
 Add New

**Parameters used in Phone**

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video
Start Media Port*	16384
Stop Media Port*	32766
DSCP for Audio Calls	Use System Default ▾
DSCP for Video Calls	Use System Default ▾
DSCP for Audio Portion of Video Calls	Use System Default ▾
DSCP for TelePresence Calls	Use System Default ▾
DSCP for Audio Portion of TelePresence Calls	Use System Default ▾
Call Pickup URI*	x-cisco-serviceuri-pickup
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup
Call Pickup Group URI*	x-cisco-serviceuri-gpickup
Meet Me Service URI*	x-cisco-serviceuri-meetme
User Info*	None ▾
DTMF DB Level*	Nominal ▾
Call Hold Ring Back*	Off ▾
Anonymous Call Block*	Off ▾
Caller ID Blocking*	Off ▾
Do Not Disturb Control*	User ▾
Telnet Level for 7940 and 7960*	Disabled ▾
Resource Priority Namespace	< None > ▾
Timer Keep Alive Expires (seconds)*	120
Timer Subscribe Expires (seconds)*	120
Timer Subscribe Delta (seconds)*	5

**Figure 20: SIP Profile2**

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### SIP Profile Configuration

Save 
 Delete 
 Copy 
 Reset 
 Apply Config 
 Add New

Maximum Redirections\*   
 Off Hook To First Digit Timer (milliseconds)\*   
 Call Forward URI\*   
 Speed Dial (Abbreviated Dial) URI\*

Conference Join Enabled  
 RFC 2543 Hold  
 Semi Attended Transfer  
 Enable VAD  
 Stutter Message Waiting  
 MLPP User Authorization

#### Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value	
1	<input type="text"/>	<input type="text"/>	<input type="button" value="+"/> <input type="button" value="-"/>

#### Incoming Requests FROM URI Settings

Caller ID DN   
 Caller Name

#### Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on\*   
 Resource Priority Namespace List   
 SIP Rel1XX Options\*   
 Video Call Traffic Class\*   
 Calling Line Identification Presentation\*   
 Session Refresh Method\*   
 Early Offer support for voice and video calls\*

Enable ANAT  
 Deliver Conference Bridge Identifier  
 Allow Passthrough of Configured Line Device Caller Information  
 Reject Anonymous Incoming Calls

Figure 21: SIP Profile3

**Cisco Unified CM Administration**  
For Cisco Unified Communications Solutions

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾ Help ▾

### SIP Profile Configuration

Reroute Incoming Request to new Trunk based on\*

Resource Priority Namespace List

SIP Rel1XX Options\*

Video Call Traffic Class\*

Calling Line Identification Presentation\*

Session Refresh Method\*

Early Offer support for voice and video calls\*

Enable ANAT  
 Deliver Conference Bridge Identifier  
 Allow Passthrough of Configured Line Device Caller Information  
 Reject Anonymous Incoming Calls  
 Reject Anonymous Outgoing Calls  
 Send ILS Learned Destination Route String  
 Connect Inbound Call before Playing Queuing Announcement

#### SIP OPTIONS Ping

Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

Ping Interval for In-service and Partially In-service Trunks (seconds)\*

Ping Interval for Out-of-service Trunks (seconds)\*

Ping Retry Timer (milliseconds)\*

Ping Retry Count\*

#### SDP Information

Send send-receive SDP in mid-call INVITE  
 Allow Presentation Sharing using BFCP  
 Allow iX Application Media  
 Allow multiple codecs in answer SDP

### 3. SIP Trunk

Select Device > Trunk > Add New

Figure 22: SIP Trunk

**Status**

*i* Status: Ready

---

**SIP Trunk Status**

**Service Status:** Full Service

**Duration:** Time In Full Service: 0 day 23 hours 14 minutes

---

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	<input type="text" value="SBC_Rewa"/>
Description	<input type="text" value="SBC_Rewa"/>
Device Pool*	<input type="text" value="711_DP"/>
Common Device Configuration	<input type="text" value="&lt; None &gt;"/>
Call Classification*	<input type="text" value="Use System Default"/>
Media Resource Group List	<input type="text" value="&lt; None &gt;"/>
Location*	<input type="text" value="Hub_None"/>
AAR Group	<input type="text" value="&lt; None &gt;"/>
Tunneled Protocol*	<input type="text" value="None"/>
QSIG Variant*	<input type="text" value="No Changes"/>
ASN.1 ROSE OID Encoding*	<input type="text" value="No Changes"/>
Packet Capture Mode*	<input type="text" value="None"/>
Packet Capture Duration	<input type="text" value="0"/>

Media Termination Point Required  
 Retry Video Call as Audio  
 Path Replacement Support  
 Transmit UTF-8 for Calling Party Name

Figure 23: SIP Trunk 1

Transmit UTF-8 Names in QSIG APDU

Unattended Port

SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.

Consider Traffic on This Trunk Secure\* When using both sRTP and TLS

Route Class Signaling Enabled\* Default

Use Trusted Relay Point\* Default

PSTN Access

Run On All Active Unified CM Nodes

**Intercompany Media Engine (IME)**

E.164 Transformation Profile < None >

**MLPP and Confidential Access Level Information**

MLPP Domain < None >

Confidential Access Mode < None >

Confidential Access Level < None >

**Call Routing Information**

Remote-Party-Id

Asserted-Identity

Asserted-Type\* Default

SIP Privacy\* Default

Trust Received Identity\* Trust All (Default)

**Inbound Calls**

Significant Digits\* All

Connected Line ID Presentation\* Default

Connected Name Presentation\* Default

Calling Search Space < None >

**Figure 24: SIP Trunk 2**

Adm Calling Search Space < None >

Prefix DN

Redirecting Diversion Header Delivery - Inbound

**Incoming Calling Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Incoming Called Party Settings**

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

[Clear Prefix Settings](#) [Default Prefix Settings](#)

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Connected Party Settings**

Connected Party Transformation CSS < None >

Use Device Pool Connected Party Transformation CSS

**Outbound Calls**

Called Party Transformation CSS < None >

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS < None >

Use Device Pool Calling Party Transformation CSS

Calling Party Selection\* Originator

Calling Line ID Presentation\* Default

**Figure 25: SIP Trunk 3**

Calling Name Presentation\* Default

Calling and Connected Party Info Format\* Deliver DN only in connected party

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS < None >

Use Device Pool Redirecting Party Transformation CSS

**Presentation Information**

Anonymous Presentation

Presentation Number

Presentation Name

Send Presentation Name and Number only in the FROM header and not in the other identity headers

**SIP Information**

**Destination**

Destination Address is an SRV

1*	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
	10.35.179.201		5060	up		Time Up: 0 day 23 hours 14 minutes

MTP Preferred Originating Codec\* 711ulaw

BLF Presence Group\* Standard Presence group

SIP Trunk Security Profile\* Non Secure SIP Trunk Profile

Routing Calling Search Space < None >

Out-Of-Dialing Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* SIP OPTIONS Profile [View Details](#)

DTMF Signaling Method\* No Preference

**Figure 26: SIP Trunk 4**

**Normalization Script**

Normalization Script < None >

Enable Trace

1	Parameter Name	Parameter Value

**Recording Information**

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation < None >

Geolocation Filter < None >

Send Geolocation Information

## 4. Route Group

Select **Call Routing > Route/Hunt > Route Group > Add New**

**Figure 27: Route Group**

**Route Group Configuration**

Save ~~Delete~~ + Add New

**Status**  
 Status: Ready

**Route Group Information**  
 Route Group Name\* Rewa\_SBC  
 Distribution Algorithm\* Circular

**Route Group Member Information**

**Find Devices to Add to Route Group**  
 Device Name contains  Find  
 Available Devices\*\*  
 CUBE  
 CharterSpectrum  
 EM2900  
 OrangeSBCLite  
 Plusnet  
 Port(s)  
 All  
 Add to Route Group

**Current Route Group Members**  
 Selected Devices (ordered by priority)\* SBC\_Rewa (All Ports)   
 Reverse Order of Selected Devices  
 Removed Devices\*\*\*

**Route Group Members**  
 SBC\_Rewa

## 5. Route List

Select **Call Routing > Route/Hunt > Route List > Add New**

**Figure 28: Route List**

**Route List Configuration**

Save ~~Delete~~ Copy Reset Apply Config + Add New

**Status**  
 Status: Ready

**Route List Information**  
 Registration: Registered with Cisco Unified Communications Manager 10.35.180.112  
 IPv4 Address: 10.35.180.112  
 Device is trusted  
 Name\* Rewa\_SBC  
 Description  
 Cisco Unified Communications Manager Group\* UCM\_UCMG  
 Enable this Route List (change effective on Save; no reset required)  
 Run On All Active Unified CM Nodes

**Route List Member Information**  
 Selected Groups\*\* Rewa\_SBC   
 Add Route Group  
 Removed Groups\*\*\*

**Route List Details**  
 Rewa\_SBC

Save Delete Copy Reset Apply Config Add New



## 6. Route Pattern

Select **Call Routing > Route/Hunt > Route Pattern > Add New**

Figure 29: Route Pattern

**Route Pattern Configuration**

Save ✖ Delete 📄 Copy ➕ Add New

**Status**  
📌 Status: Ready

**Pattern Definition**  
Route Pattern\*   
Route Partition   
Description   
Numbering Plan   
Route Filter   
MLPP Precedence\*   
 Apply Call Blocking Percentage  
Resource Priority Namespace Network Domain   
Route Class\*   
Gateway/Route List\*  [\(Edit\)](#)  
Route Option  
 Route this pattern  
 Block this pattern   
Call Classification\*   
External Call Control Profile   
 Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority  
 Require Forced Authorization Code  
Authorization Level\*   
 Require Client Matter Code

**Calling Party Transformations**  
 Use Calling Party's External Phone Number Mask  
Calling Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Calling Line ID Presentation\*   
Calling Name Presentation\*   
Calling Party Number Type\*   
Calling Party Numbering Plan\*

**Connected Party Transformations**  
Connected Line ID Presentation\*   
Connected Name Presentation\*

**Called Party Transformations**  
Discard Digits   
Called Party Transform Mask   
Prefix Digits (Outgoing Calls)   
Called Party Number Type\*   
Called Party Numbering Plan\*

**ISDN Network-Specific Facilities Information Element**  
Network Service Protocol   
Carrier Identification Code   
Network Service  Service Parameter Name  Service Parameter Value

Save Delete Copy Add New

## Supplementary Services and Features Coverage

The following checklist identifies the set of supplementary services/features supported through the configuration defined in this Interoperability document.

Sr. No.	Supplementary Services/Features	Supported

1.1	Outgoing call CLIP	✓
1.2	Incoming call CLIP	✓
1.3	Emergency call 113 (Vodafone emergency number) CLIP	✓
1.4	Outgoing existing call: A hangs up	✓
1.5	Outgoing existing call: B hangs up	✓
1.6	Incoming existing call: A hangs up	✓
1.7	Incoming existing call: B hangs up	✓
1.8	Outgoing call: B rejects	✓
1.9	Outgoing call: B does not answer	✓
1.10	Outgoing call: B busy	✓
1.11	Outgoing call: B DND	✓
1.12	Outgoing call: A hangs up	✓
1.13	Incoming call: B rejects	✓
1.14	Incoming call: B does not answer	✓
1.15	Incoming call: B busy	✓
1.16	Incoming call: B DND	✓
1.17	Incoming call: A hangs up	✓
1.18	Outgoing call: A puts B on hold	✓
1.19	Outgoing call: A retrieves B	✓
1.20	Outgoing call: B puts A on hold	✓
1.21	Outgoing call: B retrieves A	✓
1.22	Incoming call: A puts B on hold	✓
1.23	Incoming call: A retrieves B	✓
1.24	Incoming call: B puts A on hold	✓
1.25	Incoming call: B retrieves A	✓
1.26	Call forwarding	✓
1.27	Conference Call	✓
1.28	Outgoing call CLIR	✓
1.29	Incoming call CLIR	✓
1.30	Emergency call 113 (Vodafone emergency number) CLIR	✓
1.31	Outgoing call CLIP No Screening	✓
1.32	Emergency 113 (Vodafone emergency number) CLIP No Scening	✓

1.33	DTMF outgoing call: from A to B	✓
1.34	DTMF outgoing call: from B to A	✓
1.35	DTMF incoming call: from A to B	✓
1.36	DTMF incoming call: from B to A	✓
1.37	COLP No Screening	✓
1.39	COLR	✓

#### Legend

Supported	✓
Not Supported	✗

## Caveats

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The following items should be noted in relation to this Interop document. These are either limitations, untested elements, or useful information pertaining to the Interoperability.

- Not Caveats.

## Support

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For any support related queries about this guide, please contact your local Ribbon representative, or use the details below:

- Sales and Support: 1-833-742-2661
- Other Queries: 1-877-412-8867
- Website: <https://ribboncommunications.com/about-us>

## References

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For detailed information about Ribbon products & solutions, please visit:

<https://ribboncommunications.com/products>

## Conclusion

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This Interoperability document describes a successful configuration and interop involving Ribbon SBC 1000/2000 and Cisco Unified Communication Manager 12.5.

All features and capabilities tested are detailed within this document. Any limitations, notes or observations are also recorded to provide the reader with an accurate understanding of what has been covered, and what has not.

Configuration guidance is provided to enable the reader to replicate the same base setup - there may be additional configuration changes required that are specific to the exact deployment environment.

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