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# Ribbon SBC 1000/2000 V8.0.2 IOT Cisco Unified Communication Manager PlusNet SIP Trunk TCP Application Note

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# Overview

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This Application Note is a configuration guide for the Ribbon SBC (Session Border Controller) 1000/2000 when connecting to Cisco Unified Communication Manager (CUCM) and PlusNet SIP Trunk.

The configuration guide supports features outlined in the Microsoft Technet web page:

- For additional information on Cisco Platform, visit <http://www.cisco.com>.
- For additional information on Ribbon SBC 1000/2000, visit <https://ribboncommunications.com/>.


## Introduction


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Interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC 1000/2000 and Cisco CUCM.

### Audience

This technical document provides telecommunications engineers with information for configuring both the Ribbon SBC and the third-party product. Procedures in this document require navigating third-party equipment as well as applying Ribbon SBC Command Line Interface (CLI) commands. To complete the configuration and perform any troubleshooting, the engineer performing the procedures must understand the basic concepts of TCP /UDP, IP/Routing, and SIP/RTP.

 This Application Note is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this document are subject to change without notice. All statements, information, and recommendations contained in this document are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information contained here.

 The links are only internal to Ribbon partners and employees. They do not work outside of the Ribbon Network.

### Requirements

The following table lists the hardware and software used in the reference configuration.

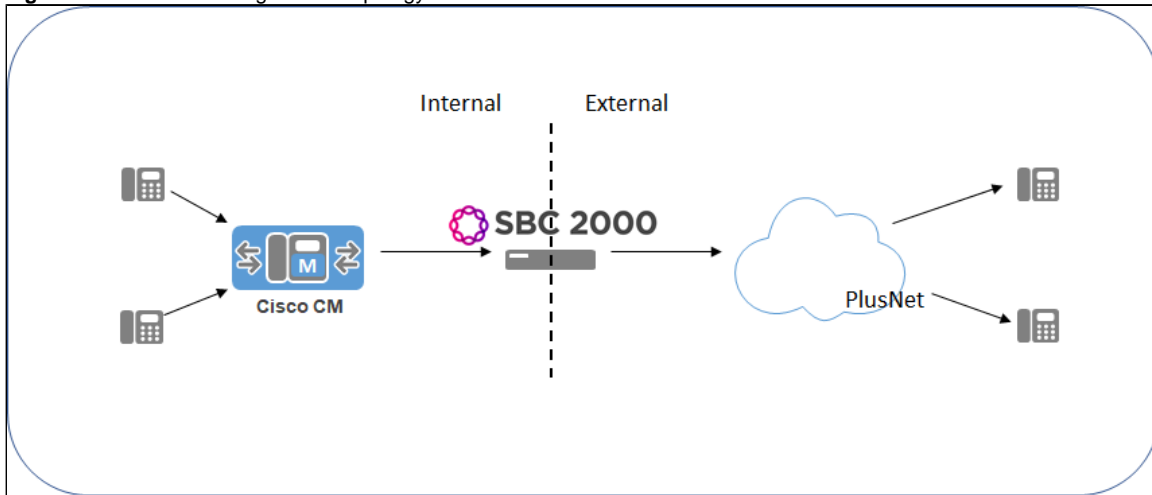
**Table 1:** Test Equipment and Software

Vendor	Equipment	Software Version
Ribbon Networks	SBC 2000	V8.0.2
Third-party Vendor		
Cisco	Cisco Unified CM Administration	12.0.1.21900
Cisco	Cisco SIP Phone 7841	sip78xx.11-7-1-17
VentaFax	Fax Machine VentaFax	7.6.243.616

## Reference Configuration

The following figure serves as a topology for the reference configuration. The figure shows the connectivity between third-party equipment and the Ribbon SBC 1000/2000.

**Figure 1:** Reference Configuration Topology



## Support

For questions about information in this document, contact Ribbon Support in either of the following ways:

- Global Support Assistance Center +1-978-614-8589 or +1-888-391-3434 (English language Support)
- Web: <https://ribboncommunications.com/services/ribbon-support-portal-login>

## Verify License

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

## CUCM 12.0.1 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 2:** SIP Trunk Security Profile



## SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

### Status

Status: Ready

### SIP Trunk Security Profile Information

Name *	<input type="text" value="Non Secure SIP Trunk Profile"/>
Description	<input type="text" value="Non Secure SIP Trunk Profile authenticated by null String"/>
Device Security Mode	<input type="text" value="Non Secure"/>
Incoming Transport Type*	<input type="text" value="TCP+UDP"/>
Outgoing Transport Type	<input type="text" value="UDP"/>
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	<input type="text" value="600"/>
X.509 Subject Name	<div style="border: 1px solid #ccc; height: 80px;"></div>
Incoming Port*	<input type="text" value="5060"/>
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input checked="" type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	<input type="text" value="Use Default Filter"/>

## 2. SIP Profile

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

Figure 3: SIP Profile

The screenshot shows the Cisco Unified CM Administration interface for SIP Profile Configuration. The page title is "Cisco Unified CM Administration" with the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "SIP Profile Configuration" and includes a toolbar with Save, Delete, Copy, Reset, Apply Config, and Add New buttons. The "Status" section shows "Status: Ready" and a warning that all SIP devices using this profile must be restarted. The "SIP Profile Information" section contains fields for Name (SIP OPTIONS Profile), Description (Default SIP Profile), and Default MTP Telephony Event Payload Type (101). It also includes several dropdown menus for 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), and Confidential Access Level Headers (Disabled). A list of checkboxes includes 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, and Enable External QoS\*\*. The "SDP Information" section includes dropdown menus for SDP Session-level Bandwidth Modifier (TJAS and AS), SDP Transparency Profile (Pass all unknown SDP attributes), and Accept Audio Codec Preferences in Received Offer (Default). It also includes checkboxes for Require SDP Inactive Exchange for Mid-Call Media Change and Allow RR/RS bandwidth modifier (RFC 3556).

Figure 4: 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 5: 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"/>		

**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 6: 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 7:** SIP Trunk



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

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

### Trunk Configuration

Save Delete Reset Add New

**Status**  
Status: Ready

**SIP Trunk Status**  
Service Status: Full Service  
Duration: Time In Full Service: 28 days 23 hours 36 minutes

**Device Information**

Product: SIP Trunk  
Device Protocol: SIP  
Trunk Service Type: None(Default)  
Device Name\*: Rewa  
Description: Rewa  
Device Pool\*: Sonus\_DP  
Common Device Configuration: < None >  
Call Classification\*: Use System Default  
Media Resource Group List: < None >  
Location\*: Hub\_None  
AAR Group: < None >  
Tunneled Protocol\*: None  
QSIG Variant\*: No Changes  
ASN.1 ROSE OID Encoding\*: No Changes  
Packet Capture Mode\*: None  
Packet Capture Duration: 0

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

**Figure 8: SIP Trunk1**

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

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

### Trunk Configuration

Save Delete Reset Add New

**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 >  
AAR 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	Skip Digits	Calling Search Space	Use Device Pool COS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**Figure 9: SIP Trunk2**

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

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

### Trunk Configuration

Save Delete Reset Add New

**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  
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

**Caller Information**  
Caller ID DN  
Caller Name  
 Maintain Original Caller ID DN and Caller Name in Identity Headers

**Figure 10: SIP Trunk3**

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

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

### Trunk Configuration

Save Delete Reset Add New

**SIP Information**

**Destination**  
 Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1*	10.35.179.136		5060	up		Time Up: 0 day 23 hours 36 minutes

MTP Preferred Originating Codec\* 711ulaw  
 BLP Presence Group\* Standard Presence group  
 SIP Trunk Security Profile\* Non Secure SIP Trunk Profile  
 Rerouting Calling Search Space < None >  
 Out-Of-Dialog Refer Calling Search Space < None >  
 SUBSCRIBE Calling Search Space < None >  
 SIP Profile\* Standard SIP Profile - OPTIONS Enable [View Details](#)  
 DTMF Signaling Method\* No Preference

**Normalization Script**  
Normalization Script < None >  
 Enable Trace

Parameter Name	Parameter Value
1	

**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

Save Delete Reset Add New

## 4. Route Group

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

Figure 11: Route Group

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

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

### 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  
Rewa  
USTX-SBCLAB01  
Port(s) All  
Add to Route Group

**Current Route Group Members**  
Selected Devices (ordered by priority)\* Rewa (All Ports)   
Removed Devices\*\*\*

**Route Group Members**  
 [Rewa](#)

Save Delete Add New

## 5. Route List

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

Figure 12: Route List

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

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

### Route List Configuration

Save Delete Copy Reset Apply Config Add New

---

#### Status

**i** Status: Ready

---

#### Route List Information

Registration: Registered with Cisco Unified Communications Manager UCM12.vo.sonusnet.com  
IPv4 Address: 10.35.180.111  
 Device is trusted  
Name\*   
Description   
Cisco Unified Communications Manager Group\*   
 Enable this Route List (change effective on Save; no reset required)  
 Run On All Active Unified CM Nodes

---

#### Route List Member Information

Selected Groups\*\*    
Removed Groups\*\*\*

---

#### Route List Details

Rewa\_SBC

Save Delete Copy Reset Apply Config Add New

**i** \*- indicates required item.  
**i** \*\*Ordered by highest priority  
**i** \*\*\*Will be removed from Route List when you click Save

## 6. Route Pattern

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

Figure 13: Route Pattern

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

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

### 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\*

Figure 14: Route Pattern1

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

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

### Route Pattern Configuration

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
<input type="text" value="-- Not Selected --"/> ▾	<input type="text" value="&lt; Not Exist &gt;"/>	<input type="text"/>

## Ribbon SBC 1000/2000 Configuration

The following configuration steps provide an example of how to configure the Ribbon SBC 1000/2000 to interoperate with Skype 2015 and Virgin Media SIP Trunk:

1. [SIP Profile](#)
2. [SIP Server](#)
3. [Media System](#)
4. [Media Profiles](#)
5. [Media List](#)
6. [Remote Authorization Tables](#)
7. [Signaling Groups](#)
8. [Transformation](#)
9. [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 the Ribbon 1000/2000 used for this configuration effort.

**Figure 15: PlusNet SIP Profile**

The screenshot displays the configuration interface for the PlusNet SIP Profile, organized into several sections:

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

**Figure 16: CUCM 12.0.1 SIP Profile**

Description:

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

Figure 17: Fax SIP Profile



▼
Tenor-Fax

Description

#### Session Timer

Session Timer  ▼

Minimum Acceptable Timer  \* secs (90..7200)

Offered Session Timer  \* secs (90..7200)

Terminate On Refresh Failure  ▼

#### MIME Payloads

ELIN Identifier  ▼

PIDF-LO Passthrough  ▼

Unknown Subtype Passthrough  ▼

#### 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  ▼

#### Options Tags

100rel  ▼

Path  ▼

Timer  ▼

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)( $\geq T1$ )

Timer T4  ms (1000..100000)

Timer D  ms (5000..640000)

Timer B  ms

Timer F  ms

Timer H  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  ▼

## 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 18: PlusNet SIP Servers**

The screenshot shows the configuration page for a PlusNet SIP server. At the top, there is a table with columns: Host / Domain, Server Lookup, Port, Protocol, Display Counters, and Primary Key. The row for 'sipconnect01.ipfonie.de' is selected, showing IP/FQDN, 5060, TCP, Counters, and 2. Below the table are four configuration panels: Server Host, Transport, Remote Authorization and Contacts, and Connection Reuse. The Server Host panel includes fields for Server Lookup (IP/FQDN), Priority (1), Host FQDN/IP (x.x.x.x), Host IP Version (IPv4), Port (5060), and Protocol (TCP). The Transport panel has a Monitor dropdown set to None. The Remote Authorization and Contacts panel includes Remote Authorization Table (PlusNet), Contact Registrant Table (None), Retry Non-State Nonce (True), Authorization on Refresh (True), and Session URI Validation (Liberal). The Connection Reuse panel includes Reuse (True), Sockets (4), and Reuse Timeout (Forever).

**Figure 19: CUCM SIP Server**

The screenshot shows the configuration page for a CUCM SIP server. At the top, there is a table with columns: Host / Domain, Server Lookup, Port, Protocol, Display Counters, and Primary Key. The row for '10.35.180.112' is selected, showing IP/FQDN, 5060, UDP, Counters, and 1. Below the table are three configuration panels: Server Host, Transport, and Remote Authorization and Contacts. The Server Host panel includes fields for Server Lookup (IP/FQDN), Priority (1), Host FQDN/IP (10.35.180.112), Port (5060), and Protocol (UDP). The Transport panel has a Monitor dropdown set to None. The Remote Authorization and Contacts panel includes Remote Authorization Table (None), Contact Registrant Table (None), and Session URI Validation (Liberal). An Apply button is located at the bottom right.

**Figure 20: Fax SIP Server**

### 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  ▼

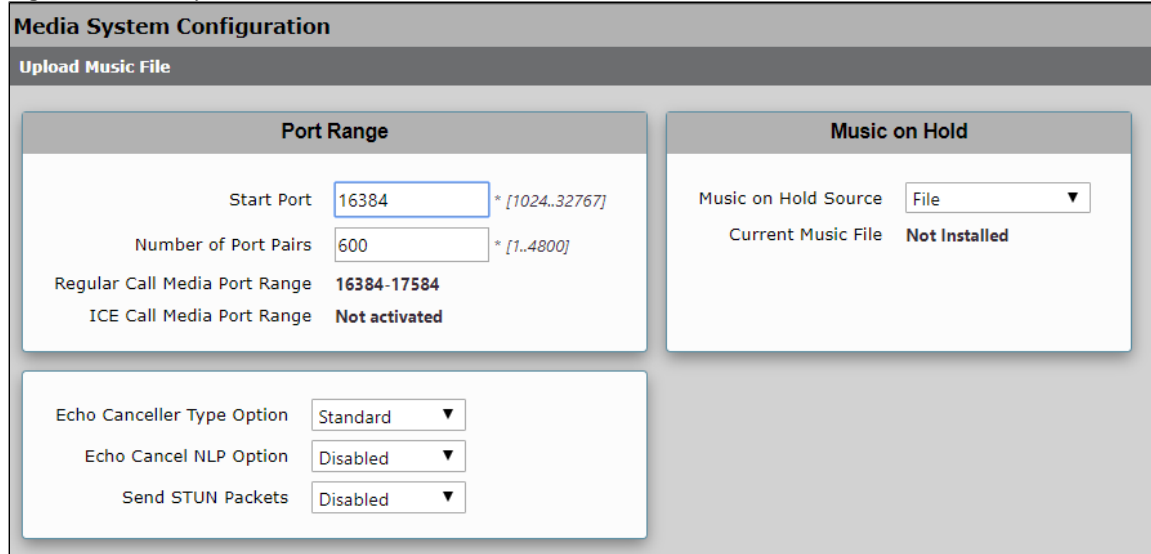
Session URI Validation  ▼

### 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 21: Media System



Port Range		Music on Hold	
Start Port	16384 * [1024..32767]	Music on Hold Source	File ▼
Number of Port Pairs	600 * [1..4800]	Current Music File	Not Installed
Regular Call Media Port Range	16384-17584		
ICE Call Media Port Range	Not activated		

Echo Cancellor Type Option	Standard ▼
Echo Cancel NLP Option	Disabled ▼
Send STUN Packets	Disabled ▼

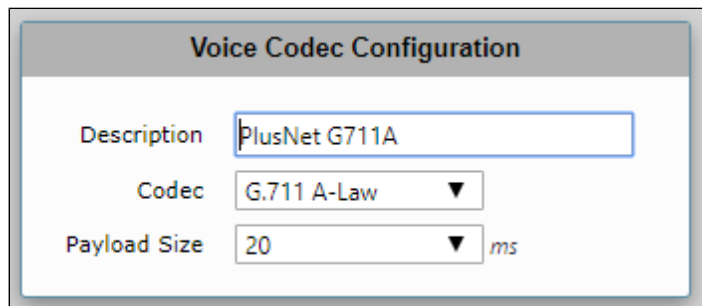
### 4. Media Profiles

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.

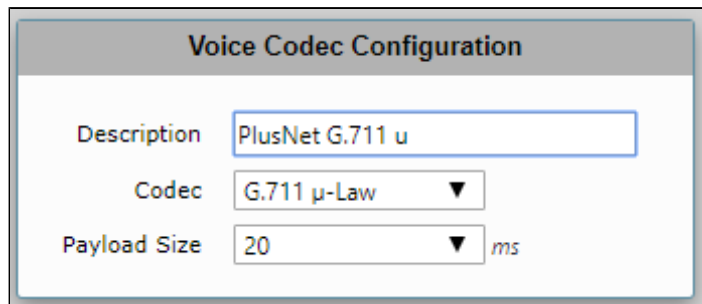
Select **Settings > Media > Media Profiles**.

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

Figure 22: PlusNet Media Profile



Voice Codec Configuration	
Description	PlusNet G711A
Codec	G.711 A-Law ▼
Payload Size	20 ▼ ms



Voice Codec Configuration	
Description	PlusNet G.711 u
Codec	G.711 μ-Law ▼
Payload Size	20 ▼ ms

**Voice Codec Configuration**

Description

Codec

Rate 64000 *b/s*

Payload Size 20 *ms*

**Voice Codec Configuration**

Description

Codec

Payload Size  *ms*

**Fax Codec Configuration**

Description

Codec T.38 Fax

Maximum Rate  *b/s*

Signaling Packet Redundancy  [*0..7*]

Payload Packet Redundancy  [*0..3*]

Error Correction Mode

Training Confirmation Procedure

Fallback to Passthrough

Super G3 to G3 Fallback

**Figure 23:** CUCM Media Profile

**Voice Codec Configuration**

Description

Codec

Payload Size  *ms*

**Figure 24:** Fax Media Profile

### Voice Codec Configuration

Description

Codec

Payload Size  ms

### Voice Codec Configuration

Description

Codec

Payload Size  ms

### Fax Codec Configuration

Description

Codec **T.38 Fax**

Maximum Rate  b/s

Signaling Packet Redundancy  [0..7]

Payload Packet Redundancy  [0..3]

Error Correction Mode

Training Confirmation Procedure

Fallback to Passthrough

Super G3 to G3 Fallback

## 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 25: PlusNet Media List

The screenshot shows the 'PlusNet Media List' configuration window. At the top, the 'Description' field contains 'PlusNet Media List'. Below it is a 'Media Profiles List' containing two entries: 'PlusNet G711A' and 'T.38 PlusNet'. To the right of this list are four buttons: 'Up', 'Down', 'Add/Edit', and 'Remove'. Below the list, there are several configuration fields: 'SDES-SRTP Profile' is set to 'None' with a note 'Associated SIP SG Listen Ports should be TLS only.'; 'DTLS-SRTP Profile' is also set to 'None'; 'Media DSCP' is set to '46' with a note '\* [0..63]'; 'RTCP Mode' is set to 'RTCP'; 'Dead Call Detection' is set to 'Disabled'; and 'Silence Suppression' is set to 'Disabled'.

The screenshot shows the 'CUCM Media List' configuration window. It is divided into three main sections: 'Gain Control', 'Digit Relay', and 'Passthrough/Tone Detection'.  
The 'Gain Control' section has two fields: 'Receive Gain' and 'Transmit Gain', both set to '0' with a range of '[-14..+6] dB'.  
The 'Digit Relay' section has two fields: 'Digit (DTMF) Relay Type' set to 'RFC 2833' and 'Digit Relay Payload Type' set to '101' with a range of '[96..127]'.  
The 'Passthrough/Tone Detection' section has six fields: 'Modem Passthrough' (Enabled), 'Fax Passthrough' (Disabled), 'CNG Tone Detection' (Disabled), 'Fax Tone Detection' (Enabled), 'DTMF Signal to Noise' set to '0' with a range of '[-3..+6] dB', and 'DTMF Minimum Level' set to '-38' with a range of '[-48..-14] dBm0'.

Figure 26: CUCM Media List

**CUCM Media List**

Description: CUCM Media List

Media Profiles List: CUCM G711A

Up, Down, Add/Edit, Remove

SDES-SRTP Profile: None *Associated SIP SG Listen Ports should be TLS only.*

DTLS-SRTP Profile: None

Media DSCP: 46 \* [0..63]

RTCP Mode: RTCP

Dead Call Detection: Disabled

Silence Suppression: Enabled

**CUCM Media List**

**Gain Control**

Receive Gain: 0 [-14..+6] dB

Transmit Gain: 0 [-14..+6] dB

**Digit Relay**

Digit (DTMF) Relay Type: RFC 2833

Digit Relay Payload Type: 101 [96..127]

**Passthrough/Tone Detection**

Modem Passthrough: Enabled

Fax Passthrough: Enabled

CNG Tone Detection: Disabled

Fax Tone Detection: Enabled

DTMF Signal to Noise: 0 [-3..+6] dB

DTMF Minimum Level: -38 [-48..-14] dBm0

Figure 27: Fax Media List



Description

Media Profiles List

- Tenor G729
- Tenor G711A
- Tenor T.38

\*

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"/> <input type="button" value="v"/> Digit Relay Payload Type <input type="text" value="101"/> [96..127]
Passthrough/Tone Detection	
Modem Passthrough <input type="text" value="Disabled"/> <input type="button" value="v"/>	
Fax Passthrough <input type="text" value="Disabled"/> <input type="button" value="v"/>	
CNG Tone Detection <input type="text" value="Enabled"/> <input type="button" value="v"/>	
Fax Tone Detection <input type="text" value="Enabled"/> <input type="button" value="v"/>	
DTMF Signal to Noise <input type="text" value="0"/> [-3..+6] dB	
DTMF Minimum Level <input type="text" value="-38"/> [-48..-14] dBm0	

## 6. Remote Authorization Tables

Remote Authorization Tables and their entries contain information used to respond to request message challenges by an upstream server. The Remote Authorization Tables on this page appear as options in Creating and Modifying Entries in the SIP Servers (For additional information about Remote Authorization Tables, see the Ribbon online SBC 1000/2000 documentation).

Select **Settings > SIP > Remote Authorization Tables** to access the Remote Authorization Tables configuration screen.

Figure 28: Remote Authorization Table

Realm	<input type="text" value="ipfonie.de"/>
Authentication ID	<input type="text" value="107530375765"/> *
Password Setting	<input type="text" value="Use Current"/> ▼
From URI User Match	<input type="text" value="Regex"/> ▼
Match Regex	<input type="text" value=".*"/>

## 7. Signaling Groups

Signaling Groups allow telephony channels to be grouped for routing and shared configuration. These groups are the entity to which calls are routed, and the location from which Call Routes are selected. These are also the location from which Tone Tables and Action Sets are selected. 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 29: PlusNet Signaling Group

Description	<input type="text" value="To/From PlusNet"/>
Admin State	<input type="text" value="Enabled"/> ▼
Service Status	Up

SIP Channels and Routing	
Action Set Table	<input type="text" value="None"/> ▼
Call Routing Table	<input type="text" value="From PlusNet"/> ▼
No. of Channels	<input type="text" value="60"/> * [1..960]
SIP Profile	<input type="text" value="PlusNet SIP Profile"/> ▼
SIP Mode	<input type="text" value="Basic Call"/> ▼
Agent Type	<input type="text" value="Back-to-Back User Agent"/> ▼
Interop Mode	<input type="text" value="Standard"/> ▼

Media Information	
Supported Audio/Fax Modes	<input type="text" value="DSP"/> ▲ <input type="text" value="Proxy"/> ▼ <input type="text" value="Direct"/> ▼ <input type="button" value="Add/Edit"/> * <input type="button" value="Remove"/>
Supported Video/Application Modes	Disabled
Media List ID	<input type="text" value="PlusNet Media List"/> ▼
Play Ringback	<input type="text" value="Auto on 180"/> ▼

SIP Server Table	Test_PlusNet	▼
Load Balancing	Priority: Register All	▼
Channel Hunting	Most Idle	▼
Notify Lync CAC Profile	Disable	▼
Challenge Request	Disable	▼
Outbound Proxy IP/FQDN		
Outbound Proxy Port	5060	[1..65535]
No Channel Available Override	34: No Circuit/Channel Available	▼
Call Setup Response Timer	255	[180..750] secs
Call Proceeding Timer	180	[24..750] secs
QoE Reporting	Disabled	▼
Use Register as Keep Alive	Enable	▼

Tone Table	Default Tone Table	▼
Play Congestion Tone	Disable	▼
Early 183	Disable	▼
Allow Refresh SDP	Enable	▼
Music on Hold	Disabled	▼
RTCP Multiplexing	Disable	▼

Mapping Tables	
SIP To Q.850 Override Table	Default (RFC4497) ▼
Q.850 To SIP Override Table	Default (RFC4497) ▼
Pass-thru Peer SIP Response Code	Enable ▼

SIP IP Details	
Signaling/Media Source IP	Ethernet 3 IP (216.110.2.220) ▼
Signaling DSCP	40 [0..63]
NAT Traversal	
ICE Support	Disabled ▼
Static NAT - Outbound	
Outbound NAT Traversal	None ▼
Static NAT - Inbound	
Detection	Disabled ▼

Listen Ports	Federated IP/FQDN						
<p>Total 1 SIP Listen Port Row</p> <table border="1"> <thead> <tr> <th>Port</th> <th>Protocol</th> <th>TLS Profile ID</th> </tr> </thead> <tbody> <tr> <td>5060</td> <td>TCP</td> <td>N/A</td> </tr> </tbody> </table>	Port	Protocol	TLS Profile ID	5060	TCP	N/A	<p>Total 0 SIP Federated IP Rows</p> <p>-- Table is empty --</p>
Port	Protocol	TLS Profile ID					
5060	TCP	N/A					
<p>Message Manipulation Disabled ▼</p>							

Figure 30: CUCM Signaling Group

Description	CUCM 12.0.1
Admin State	Enabled ▼
Service Status	Up

SIP Channels and Routing	
Action Set Table	None ▼
Call Routing Table	From_CUCM 14.0.1 ▼
No. of Channels	60 [1..960]
SIP Profile	CUCM 12.0.1 ▼
SIP Mode	Basic Call ▼
Agent Type	Back-to-Back User Agent ▼
Interop Mode	Standard ▼

Media Information		
Supported Audio/Fax Modes	DSP Proxy Direct	<input type="button" value="Add/Edit"/> <input type="button" value="Remove"/>
Supported Video/Application Modes	Disabled	
Media List ID	Default Media List ▼	
Play Ringback	Auto on 180 ▼	

SIP Server Table	CUCM 12.0.1
Load Balancing	Round Robin
Channel Hunting	Most Idle
Notify Lync CAC Profile	Disable
Challenge Request	Disable
Outbound Proxy IP/FQDN	
Outbound Proxy Port	5060 [1..65535]
No Channel Available Override	34: No Circuit/Channel Available
Call Setup Response Timer	255 [180..750] secs
Call Proceeding Timer	180 [24..750] secs
QoE Reporting	Disabled
Use Register as Keep Alive	Enable

Tone Table	Default Tone Table
Play Congestion Tone	Disable
Early 183	Disable
Allow Refresh SDP	Enable
Music on Hold	Disabled
RTCP Multiplexing	Disable

Mapping Tables	
SIP To Q.850 Override Table	Default (RFC4497)
Q.850 To SIP Override Table	Default (RFC4497)
Pass-thru Peer SIP Response Code	Enable

<table border="1"> <thead> <tr> <th colspan="2">SIP IP Details</th> </tr> </thead> <tbody> <tr> <td>Signaling/Media Source IP</td> <td>Ethernet 1 IP (10.35.179.136)</td> </tr> <tr> <td>Signaling DSCP</td> <td>40 [0..63]</td> </tr> <tr> <td colspan="2">NAT Traversal</td> </tr> <tr> <td>ICE Support</td> <td>Disabled</td> </tr> <tr> <td colspan="2">Static NAT - Outbound</td> </tr> <tr> <td>Outbound NAT Traversal</td> <td>None</td> </tr> <tr> <td colspan="2">Static NAT - Inbound</td> </tr> <tr> <td>Detection</td> <td>Disabled</td> </tr> </tbody> </table>	SIP IP Details		Signaling/Media Source IP	Ethernet 1 IP (10.35.179.136)	Signaling DSCP	40 [0..63]	NAT Traversal		ICE Support	Disabled	Static NAT - Outbound		Outbound NAT Traversal	None	Static NAT - Inbound		Detection	Disabled
SIP IP Details																		
Signaling/Media Source IP	Ethernet 1 IP (10.35.179.136)																	
Signaling DSCP	40 [0..63]																	
NAT Traversal																		
ICE Support	Disabled																	
Static NAT - Outbound																		
Outbound NAT Traversal	None																	
Static NAT - Inbound																		
Detection	Disabled																	

Listen Ports			
Total 2 SIP Listen Port Rows			
<input type="checkbox"/>	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		
<input type="checkbox"/>	IP/FQDN	Netmask/Prefix
<input type="checkbox"/>	10.35.180.111	255.255.255.255

Figure 31: Fax Signaling Group

Description	To/From Tenor-Fax
Admin State	Enabled
Service Status	Up

SIP Channels and Routing	
Action Set Table	None
Call Routing Table	From Tenor-Fax
No. of Channels	60 [1..960]
SIP Profile	Tenor-Fax
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
Interop Mode	Standard

Media Information	
Supported Audio/Fax Modes	DSP Proxy Direct
Supported Video/Application Modes	Disabled
Media List ID	Tenor-Fax Media List
Play Ringback	Auto on 180

SIP Server Table	Tenor_Fax	Tone Table	Default Tone Table
Load Balancing	Round Robin	Play Congestion Tone	Disable
Channel Hunting	Most Idle	Early 183	Disable
Notify Lync CAC Profile	Disable	Allow Refresh SDP	Enable
Challenge Request	Disable	Music on Hold	Disabled
Outbound Proxy IP/FQDN		RTCP Multiplexing	Disable
Outbound Proxy Port	5060 [1..65535]	<b>Mapping Tables</b>	
No Channel Available Override	34: No Circuit/Channel Available	SIP To Q.850 Override Table	Default (RFC4497)
Call Setup Response Timer	255 [180..750] secs	Q.850 To SIP Override Table	Default (RFC4497)
Call Proceeding Timer	180 [24..750] secs	Pass-thru Peer SIP Response Code	Enable
QoE Reporting	Disabled		
Use Register as Keep Alive	Enable		
Forked Call Answered Too Soon	Disable		

<b>SIP IP Details</b>	
Signaling/Media Source IP	Ethernet 1 IP (10.35.179.136)
Signaling DSCP	40 * [0..63]
<b>NAT Traversal</b>	
ICE Support	Disabled
<b>Static NAT - Outbound</b>	
Outbound NAT Traversal	None
<b>Static NAT - Inbound</b>	
Detection	Disabled

Listen Ports			
Total 2 SIP Listen Port Rows			
Port	Protocol	TLS Profile ID	
5060	UDP	N/A	
5060	TCP	N/A	

Federated IP/FQDN		
Total 1 SIP Federated IP Row		
IP/FQDN	Netmask/Prefix	
10.35.137.106	255.255.255.255	

## 8. 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 a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table.

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

**Figure 32: PlusNet Transformation**

From PlusNet								January 02, 2020 11:26:53
Total 6 Transformation Entry Rows								Filter...
Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key	
▶	Called Address/Number	\+(4932211057459)	Called Address/Number	\1	Optional (Match One)	Entry ID 4	4	
▶	Called Address/Number	\+(496029995243)	Called Address/Number	\1	Optional (Match One)	Entry ID 6	6	
▶	Calling Address/Number	(*)	Calling Address/Number	\1	Optional (Match One)	From_PlusNet	2	
▶	Calling Address/Number	\+(4991147726289)	Calling Address/Number	\1	Optional (Match One)	Entry ID 5	5	
▶	Called Address/Number	\+(*)	Called Address/Number	\1	Optional (Match One)	From_PlusNet	1	
▶	Called Address/Number	\+(4991147726289)	Called Address/Number	\1	Optional (Match One)	4991147726289	3	

**Figure 33: CUCM Transformation**

From_CUCM 14.0.1								January 02, 2020 11:28:04
Total 2 Transformation Entry Rows								
Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key	
▶	Called Address/Number	(*)	Called Address/Number	+1	Optional (Match One)	From_CUCM	1	
▶	Calling Address/Number	(*)	Calling Address/Number	+1	Optional (Match One)	From_CUCM	2	

**Figure 34: Fax-Tenor Transformation**

From Tenor-Fax								January 02, 2020 11:28:59
Total 5 Transformation Entry Rows								
Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key	
▶	Calling Address/Number	(4932211057459)	Calling Address/Number	+1	Optional (Match One)	Entry ID 3	3	
▶	Called Address/Number	(496029995243)	Called Address/Number	+1	Optional (Match One)	Entry ID 5	5	
▶	Called Address/Number	(499113929210)	Called Address/Number	+1	Optional (Match One)	From_Tenr-Fax	1	
▶	Called Address/Number	(4932166985357)	Called Address/Number	+1	Optional (Match One)	Entry ID 4	4	
▶	Called Address/Number	1(*)	Called Address/Number	+1\1	Optional (Match One)	TO US FAX	2	



## 9. 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 flexible configuration of which calls will 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 35:** PlusNet Call Routing

The screenshot displays the PlusNet Call Routing configuration interface, organized into four main sections:

- Route Details:** Includes fields for Description (To CUCM 14.0.1), Admin State (Enabled), Route Priority (1), Call Priority (Normal), Number/Name Transformation Table (From PlusNet), and Time of Day Restriction (None).
- Destination Information:** Includes Destination Type (Normal), Message Translation Table (None), Cause Code Reroutes (None), Cancel Others upon Forwarding (Disabled), Fork Call (No), Destination Signaling Groups (a list containing (SIP) CUCM 12.5 and (SIP) CUCM 12.0.1 with Up, Down, Add/Edit, and Remove buttons), and Enable Maximum Call Duration (Disabled).
- Media:** Includes Audio/Fax Stream Mode (DSP), Video/Application Stream Mode (Disabled), Media Transcoding (Enabled), and Media List (None).
- Quality of Service:** Includes Quality Metrics Number of Calls (10), Quality Metrics Time Before Retry (10), Min. ASR Threshold (0), Enable Min MOS Threshold (Disable), Enable Max. R/T Delay (Enable), Max. R/T Delay (65535), Enable Max. Jitter (Enable), and Max. Jitter (3000).

**Figure 36:** CUCM 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**

**Quality of Service**

Audio/Fax Stream Mode  ▼

Video/Application Stream Mode  ▼

Media Transcoding  ▼

Media List  ▼

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 37: Fax 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) To/From PlusNet
-----------------------

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]

# Interoperability Test Results

The following table provides test results for interoperability compliance testing between Ribbon SBC 1000/2000 and CUCM

**Table 2:** Interoperability Compliance Test Results

Test Number	Test Scenario	Setup / Test Results	Status	Comment
4.1	Registration and authentication (registration mode)	The PBX is able to execute the correct resolution of the DNS SRV record	Pass	
4.3	Basic call	With the basic call tests, the standard call scenarios and the CLIP/CLIR features are tested. <ul style="list-style-type: none"> <li>Only en-bloc dialing is supported, overlap sending is not possible.</li> </ul>	Pass	
4.3.1	Normal call	Outgoing call from PBX to PSTN <ul style="list-style-type: none"> <li>En-bloc dialing</li> <li>Local area call (without area code); area code must be set by the PBX</li> <li>Setting of the correct calling number with all available telephone number blocks</li> <li>If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the user's location. This is very important in case of emergency calls, because Plusnet uses the number of the PAI header to route the emergency call to the proper emergency call center.</li> </ul>	Pass	
4.3.1.1	Normal call	The geographic number in the PAI header corresponds to the location 11 of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.1.2	Normal call	Display of A-number in B-party CLIP (national PSTN)	Pass	
4.3.1.3	Normal call	Display of A-number in B-party CLIP (international PSTN)	Pass	
4.3.1.4	Normal call	Display of A-number in B-party CLIP (mobile)	Pass	
4.3.1.5	Normal call	Call to mobile Outgoing call to mobile => mobile phone turned off	Pass	
4.3.1.6	Normal call	Suppression of A-number => CUR	Pass	
4.3.1.7	Normal call	Outgoing call from analog extension	Pass	
4.3.1.8	Normal call	Outgoing call (> 5 min.) => PSTN	Pass	Hold connection for 5 minutes => RTP still correct? Yes
4.3.2	Normal call	Incoming call from PSTN (national) => PBX <ul style="list-style-type: none"> <li>Test all available telephone number blocks</li> </ul>	Pass	
4.3.2.1	Normal call	Display of A-number => CLIP	Pass	
4.3.2.2	Normal call	Incoming call from mobile => PBX Display of A-number => CLIP	Pass	
4.3.2.3	Normal call	Suppression of A-number => CLIR	Pass	
4.3.3	Normal call	Two simultaneous outgoing/incoming calls	Pass	
4.3.4	Normal call	Enabled feature DND (do not disturb)	Pass	

4.3.5	Normal call	Test call with codec G.711	Pass	
4.3.6	Normal call	Test call with codec G.722 (only SIP <=> SIP)	Fail	PlusNet didn't support G.722
4.3.7	Normal call	Test call with codec G.729	Pass	
4.3.2.8	Clip No Screening	Outgoing call from PBX to PSTN <ul style="list-style-type: none"> <li>• With feature Clip No Screening</li> <li>• Test with several different A-numbers</li> <li>• If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the users location. This is very important in case of emergency calls, because PlusNet uses the number</li> </ul>	Pass	
4.3.2.8.1	Clip No Screening	Despite Clip No Screening the geographic number in the PAI header corresponds to the location of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.2.8.2	Clip No Screening	Display of A-number (NoSClip) at B-party (PSTN)	Pass	
4.3.2.8.3	Clip No Screening	Display of A-number (NoSClip) at B-party (international PSTN; depending on the destination carrier, the NoSClip telephone number may not be displayed in this case!)	Pass	
4.3.2.8.3	Clip No Screening	Call made from a PSTN line to an IP-PBX line with call forward to a line within the same IP-PBX, Answer Call. <ul style="list-style-type: none"> <li>• Either party terminates call.</li> </ul>	Pass	Does the IP-PBX has configuration settings to send SIP status 181 messages to the soft switch? Yes
4.3.2.8.4	Clip No Screening	Display of A-number (NoSClip) at B-party (mobile)	Pass	
4.3.3.9	Special call situations	Outgoing call PBX => PSTN <ul style="list-style-type: none"> <li>• Call is rejected by B-party</li> </ul>	Pass	
4.3.3.10	Special call situations	Outgoing call PBX=> PSTN <ul style="list-style-type: none"> <li>• B-party does not answer; clearing after timer</li> </ul>	Pass	
4.3.3.11	Special call situations	Outgoing call PBX => PSTN <ul style="list-style-type: none"> <li>• B-party busy; busy tone</li> </ul>	Pass	
4.3.3.12	Special call situations	Outgoing call PBX=> PSTN <ul style="list-style-type: none"> <li>• A-party hangs up before call is established (cancel)</li> </ul>	Pass	
4.3.3.13	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• Call is rejected by PBX party</li> </ul>	Pass	
4.3.3.14	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• PBX party does not answer; clearing after timer</li> </ul>	Pass	
4.3.3.15	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• PBX party busy; busy tone</li> </ul>	Pass	

4.3.3.16	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>A-party hangs up before call is established (cancel)</li> </ul>	Pass	
4.3.4.17	Call clearing	Incoming / outgoing call; clearing after established call. Correct clearing on both sides <ul style="list-style-type: none"> <li>PBX party hangs up</li> <li>PSTN party hangs up</li> </ul>	Pass	
4.3.4.18	Call clearing	Interrupting the network connection of the SIP terminal device during a call <ul style="list-style-type: none"> <li>Call should be cleared correctly</li> </ul>	Pass	
4.4.19	Hold	PBX => PSTN and PSTN => PBX <ul style="list-style-type: none"> <li>Test call in both directions</li> </ul>	Pass	
4.4.19.1	Hold	Putting an external call on hold in the PBX	Pass	
4.4.19.2	Hold	If applicable, MoH (music on hold) at A-party (PSTN)	Pass	
4.4.19.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.19.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.4.20	Hold	PBX => PSTN and PSTN => PBX <ul style="list-style-type: none"> <li>Test call in both directions</li> </ul>	Pass	
4.4.20.1	Hold	Putting an external call on hold in the PSTN	Pass	
4.4.20.2	Hold	If applicable, MoH (music on hold) at A-party (PBX)	Pass	
4.4.20.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.20.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.5.21	Call transfer	Internal call is transferred to external party: internal => PBX => external	Pass	
4.5.21.1	Call transfer	Call transfer from PBX party => PSTN party with announcement (attendant transfer)	Pass	
4.5.21.2	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer)	Pass	
4.5.21.3	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) <ul style="list-style-type: none"> <li>Call is rejected or not answered</li> </ul>	Pass	
4.5.21.4	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) <ul style="list-style-type: none"> <li>PSTN party busy</li> </ul>	Pass	
4.5.22	Call transfer	Transferred call from external party to PBX: PSTN => PBX	Pass	
4.5.22.1	Call transfer	Call transfer from PSTN => PBX party with announcement (attendant transfer)	Pass	
4.5.22.2	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer)	Pass	
4.5.22.3	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) <ul style="list-style-type: none"> <li>Call is rejected or not answered</li> </ul>	Pass	
4.5.22.4	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) <ul style="list-style-type: none"> <li>PBX party busy</li> </ul>	Pass	
4.5.23	Call transfer	Call from external party transferred to another external party: external => PBX => external	Pass	
4.5.23.1	Call transfer	PSTN => PBX party => PSTN with announcement (attendant transfer)	Pass	
4.5.23.2	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer)	Pass	

4.5.23.3	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) <ul style="list-style-type: none"> <li>• Call is rejected or not answered</li> </ul>	Pass	
4.5.23.4	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) <ul style="list-style-type: none"> <li>• PSTN party busy</li> </ul>	Pass	
4.6.24	Call diversion	PBX party CFU to external party (PSTN)	Pass	
4.6.24.1	Call diversion	Internal call (CFU) => PSTN	Pass	
4.6.24.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.3	Call diversion	A-party clears in ring phase	Pass	
4.6.24.4	Call diversion	External => PBX party (CFU) => PSTN	Pass	
4.6.24.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.6	Call diversion	A-party clears in ring phase	Pass	
4.6.25	Call diversion	PBX party CFNR to external party	Pass	
4.6.25.1	Call diversion	Internal call (CFNR) => PSTN	Pass	
4.6.25.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.3	Call diversion	A-party clears in ring phase	Pass	
4.6.25.4	Call diversion	External => PBX party (CFNR) => PSTN	Pass	
4.6.25.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.6	Call diversion	A-party clears in ring phase	Pass	
4.6.26	Call diversion	PBX party CFB to external party	Pass	
4.6.26.1	Call diversion	Internal call (CFB) => PSTN	Pass	
4.6.26.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.3	Call diversion	A-party clears in ring phase	Pass	
4.6.26.4	Call diversion	External => PBX party (CFB) => PSTN	Pass	
4.6.26.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.6	Call diversion	A-party clears in ring phase	Pass	
4.6.27	Call diversion	External party (PSTN, mobile, etc.) CFU to PBX party:	Pass	
4.6.27.1	Call diversion	External (CFU) => PBX party	Pass	

4.6.27.2	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.3	Call diversion	A-party clears in ring phase	Pass	
4.6.27.4	Call diversion	External (CFNR) => PBX party	Pass	
4.6.27.5	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.6	Call diversion	A-party clears in ring phase	Pass	
4.6.28	Call diversion	External party (PSTN, mobile, etc.) CFB to PBX party:	Pass	
4.6.28.1	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.28.2	Call diversion	A-party clears in ring phase	Pass	
4.6.29	Call diversion	Call deflection: diversion during the ring phase	Pass	
4.6.29.1.1	Call diversion	Internal call PBX party CD => PBX party	Pass	
4.6.29.1.2	Call diversion	PBX party busy	Pass	
4.6.29.1.3	Call diversion	PBX party does not answer	Pass	
4.6.29.1.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.2.1	Call diversion	External call to PBX party CD => PBX party	Pass	
4.6.29.2.2	Call diversion	PBX party busy	Pass	
4.6.29.2.3	Call diversion	PBX party does not answer	Pass	
4.6.29.2.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.3.1	Call diversion	External call to PBX party CD => external PSTN party	Pass	
4.6.29.3.2	Call diversion	PSTN party busy	Pass	
4.6.29.3.3	Call diversion	PSTN party does not answer	Pass	
4.6.29.3.4	Call diversion	A-party clears in ring phase	Pass	
4.7.30	Call waiting	Incoming call during active internal call	Pass	
4.7.30.1	Call waiting	Call waiting tone	Pass	
4.7.30.2	Call waiting	Display of number of waiting party	Pass	
4.7.30.3	Call waiting	Acceptance of waiting call	Pass	
4.7.30.4	Call waiting	Putting the waiting call on hold	Pass	
4.7.30.5	Call waiting	Retrieve of waiting call	Pass	
4.7.30.6	Call waiting	Holding the 2nd call	Pass	
4.7.30.7	Call waiting	Terminating the active call	Pass	
4.7.30.8	Call waiting	Disregarding the waiting call	Pass	

4.7.30.9	Call waiting	Rejecting the waiting call	Pass	
4.8.31	3-party conference	Establishing a conference according to the operating instructions of the PBX Internal - internal - external	Pass	
4.8.31.1	3-party conference	Selecting a party (internal or external); 3rd party is put on hold <ul style="list-style-type: none"> <li>Switching to 3rd party; 2nd party is put on hold</li> <li>Reactivating the conference</li> <li>Clearing one party (internal or external)</li> <li>Terminating the conference</li> </ul>	Pass	
4.8.32	3-party conference	Establishing a conference according to the operating instructions of the PBX internal - external - external	Pass	
4.8.32.1	3-party conference	Selecting a party (external); 3rd party is put on hold <ul style="list-style-type: none"> <li>Switching to 3rd party; 2nd party is put on hold</li> <li>Reactivating the conference</li> <li>Clearing one party (external)</li> <li>Terminating the conference</li> </ul>	Pass	
4.9.33	Pick up	Picking up a call from another extension of the PBX	Pass	
4.10.34	Call list	Entries in the call list <ul style="list-style-type: none"> <li>Incoming from PSTN</li> <li>Incoming from mobile</li> <li>Incoming CLIR</li> <li>Dialing prefix for outside line in the call list</li> </ul>	Pass	
4.10.35	Call list	Call-back from call list <ul style="list-style-type: none"> <li>To PSTN</li> <li>To mobile</li> </ul>	Pass	
4.11.36	DTMF	DTMF support (G.711) <ul style="list-style-type: none"> <li>PSTN =&gt; PBX (SIP terminal device)</li> <li>PSTN =&gt; PBX (analog or system terminal device)</li> <li>PBX (SIP terminal device) =&gt; PSTN</li> <li>PBX (analog or system terminal device) =&gt; PSTN</li> </ul>	Pass	
4.11.37	DTMF	DTMF support (G.729) <ul style="list-style-type: none"> <li>- PSTN =&gt; PBX (SIP terminal device)</li> <li>- PSTN =&gt; PBX (analog or system terminal device)</li> <li>- PBX (SIP terminal device) =&gt; PSTN</li> <li>- PBX (analog or system terminal device) =&gt; PSTN</li> </ul>	Pass	
4.12.38	Fax	Fax reception (G.711 only)	Pass	
4.12.38.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed <ul style="list-style-type: none"> <li>One-page fax</li> <li>Multi-page fax (at least 5 pages)</li> </ul>	Pass	
4.12.39	Fax	Fax sending (G.711 only)	Pass	
4.12.39.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed <ul style="list-style-type: none"> <li>One-page fax</li> <li>Multi-page fax (at least 5 pages)</li> </ul>	Pass	
4.12.40	Fax	Fax reception via T.38	Pass	
4.12.40.1	Fax	Re-invite to T.38 by PBX or network <ul style="list-style-type: none"> <li>- One-page fax</li> <li>- Multi-page fax (at least 5 pages)</li> </ul>	Pass	
4.12.41	Fax	Fax sending via T.38 (not possible in conjunction with the encryption option)	Pass	

4.12.41.1	Fax	<p>Re-invite to T.38 by PBX or network</p> <ul style="list-style-type: none"> <li>• T.38-only invites are not supported</li> <li>• One-page fax</li> <li>• Multi-page fax (at least 5 pages)</li> </ul>	Pass	
5.42	Redundancy	<p>Test of redundancy:</p> <ul style="list-style-type: none"> <li>• Only if redundant connection is possible on the PBX side</li> <li>• This requires at least two PBX servers to be online on the SIP trunk in registration mode, or exactly two PBX servers in peering mode.</li> </ul>	Pass	
5.42.1	Redundancy	<ol style="list-style-type: none"> <li>1. Register all available PBX systems the Plusnet SBC.</li> <li>2. Calls from Plusnet =&gt; PBX are routed by round robin procedure</li> <li>3. Deliberately de-register one PBX system =&gt; no more calls are routed to this PBX</li> <li>4. and/or disconnect the PBX system from LAN =&gt; after the register expire period, no more calls are routed to this PBX.</li> <li>5. Calls are only routed to the remaining PBX systems.</li> </ol>	Pass	

## Conclusion

---

This Application Notes document describes the steps required to configure the Ribbon SBC 1000/2000 to successfully interoperate with the Cisco CUCM and PlusNet SIP Trunk. All feature and serviceability test cases have been completed. The majority of test cases passed with noted exceptions and observations provided in [Interoperability Test Results](#).

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# Overview

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This Application Note is a configuration guide for the Ribbon SBC (Session Border Controller) 1000/2000 when connecting to Cisco Unified Communication Manager (CUCM) and PlusNet SIP Trunk.

The configuration guide supports features outlined in the Microsoft Technet web page:

- For additional information on Cisco Platform, visit <http://www.cisco.com>.
- For additional information on Ribbon SBC 1000/2000, visit <https://ribboncommunications.com/>.


## Introduction


---

Interoperability compliance testing focuses on verifying inbound and outbound call flows between Ribbon SBC 1000/2000 and Cisco CUCM.

### Audience

This technical document provides telecommunications engineers with information for configuring both the Ribbon SBC and the third-party product. Procedures in this document require navigating third-party equipment as well as applying Ribbon SBC Command Line Interface (CLI) commands. To complete the configuration and perform any troubleshooting, the engineer performing the procedures must understand the basic concepts of TCP /UDP, IP/Routing, and SIP/RTP.

 This Application Note is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this document are subject to change without notice. All statements, information, and recommendations contained in this document are believed to be accurate but are presented without warranty of any kind, express or implied, and are provided "AS IS". Users must take full responsibility for the application of the specifications and information contained here.

 The links are only internal to Ribbon partners and employees. They do not work outside of the Ribbon Network.

### Requirements

The following table lists the hardware and software used in the reference configuration.

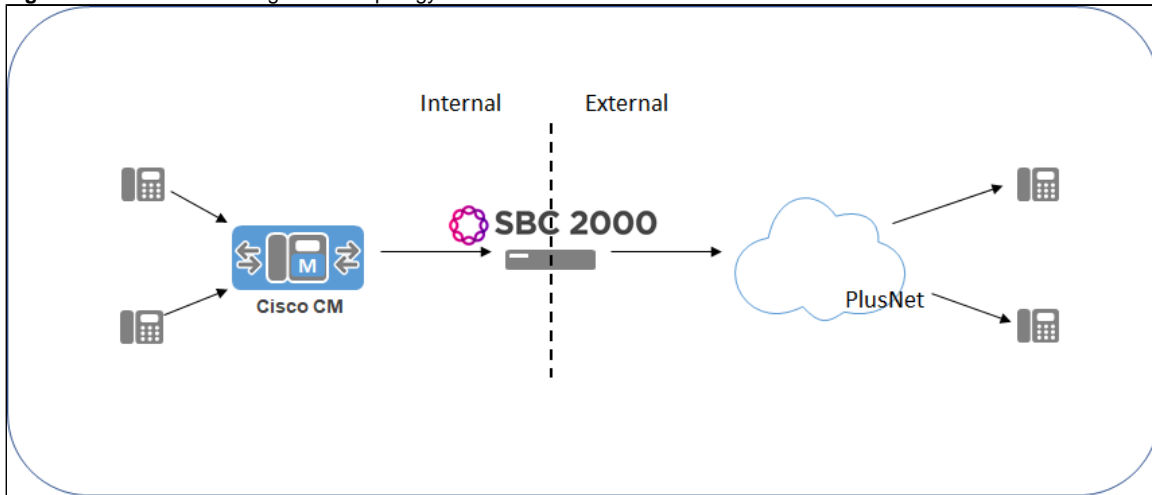
**Table 3:** Test Equipment and Software

Vendor	Equipment	Software Version
Ribbon Networks	SBC 2000	V8.0.2
Third-party Vendor		
Cisco	Cisco Unified CM Administration	12.0.1.21900
Cisco	Cisco SIP Phone 7841	sip78xx.11-7-1-17
VentaFax	Fax Machine VentaFax	7.6.243.616

## Reference Configuration

The following figure serves as a topology for the reference configuration. The figure shows the connectivity between third-party equipment and the Ribbon SBC 1000/2000.

**Figure 38:** Reference Configuration Topology



## Support

For questions about information in this document, contact Ribbon Support in either of the following ways:

- Global Support Assistance Center +1-978-614-8589 or +1-888-391-3434 (English language Support)
- Web: <https://ribboncommunications.com/services/ribbon-support-portal-login>

## Verify License

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

## CUCM 12.0.1 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 39:** SIP Trunk Security Profile



## SIP Trunk Security Profile Configuration

Save Delete Copy Reset Apply Config Add New

### Status

Status: Ready

### SIP Trunk Security Profile Information

Name *	<input type="text" value="Non Secure SIP Trunk Profile"/>
Description	<input type="text" value="Non Secure SIP Trunk Profile authenticated by null String"/>
Device Security Mode	<input type="text" value="Non Secure"/>
Incoming Transport Type*	<input type="text" value="TCP+UDP"/>
Outgoing Transport Type	<input type="text" value="UDP"/>
<input type="checkbox"/> Enable Digest Authentication	
Nonce Validity Time (mins)*	<input type="text" value="600"/>
X.509 Subject Name	<div style="border: 1px solid #ccc; height: 80px;"></div>
Incoming Port*	<input type="text" value="5060"/>
<input type="checkbox"/> Enable Application level authorization	
<input type="checkbox"/> Accept presence subscription	
<input checked="" type="checkbox"/> Accept out-of-dialog refer**	
<input checked="" type="checkbox"/> Accept unsolicited notification	
<input checked="" type="checkbox"/> Accept replaces header	
<input checked="" type="checkbox"/> Transmit security status	
<input type="checkbox"/> Allow charging header	
SIP V.150 Outbound SDP Offer Filtering*	<input type="text" value="Use Default Filter"/>

## 2. SIP Profile

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

Figure 40: SIP Profile

The screenshot shows the Cisco Unified CM Administration interface for SIP Profile Configuration. The page title is "Cisco Unified CM Administration" with the subtitle "For Cisco Unified Communications Solutions". The navigation menu includes System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. The main content area is titled "SIP Profile Configuration" and includes a toolbar with Save, Delete, Copy, Reset, Apply Config, and Add New buttons. The "Status" section shows "Status: Ready" and a warning that all SIP devices using this profile must be restarted. The "SIP Profile Information" section contains the following fields and options:

- 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\*\*:

The "SDP Information" section contains the following fields and options:

- 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 41: 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 42: 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"/>		

**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 43: 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 44:** SIP Trunk



**Cisco Unified CM Administration**  
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### Trunk Configuration

Save Delete Reset Add New

**Status**  
Status: Ready

**SIP Trunk Status**  
Service Status: Full Service  
Duration: Time In Full Service: 28 days 23 hours 36 minutes

**Device Information**

Product: SIP Trunk  
Device Protocol: SIP  
Trunk Service Type: None(Default)  
Device Name\*: Rewa  
Description: Rewa  
Device Pool\*: Sonus\_DP  
Common Device Configuration: < None >  
Call Classification\*: Use System Default  
Media Resource Group List: < None >  
Location\*: Hub\_None  
AAR Group: < None >  
Tunneled Protocol\*: None  
QSIG Variant\*: No Changes  
ASN.1 ROSE OID Encoding\*: No Changes  
Packet Capture Mode\*: None  
Packet Capture Duration: 0

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

Figure 45: SIP Trunk1

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

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

### Trunk Configuration

Save Delete Reset Add New

**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 >  
AAR 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	Skip Digits	Calling Search Space	Use Device Pool COS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

Figure 46: SIP Trunk2

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

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

### Trunk Configuration

Save Delete Reset Add New

**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  
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

**Caller Information**  
Caller ID DN  
Caller Name  
 Maintain Original Caller ID DN and Caller Name in Identity Headers

**Figure 47: SIP Trunk3**

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

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

### Trunk Configuration

Save Delete Reset Add New

**SIP Information**

**Destination**  
 Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port	Status	Status Reason	Duration
1*	10.35.179.136		5060	up		Time Up: 0 day 23 hours 36 minutes

MTP Preferred Originating Codec\* 711ulaw  
 BLF Presence Group\* Standard Presence group  
 SIP Trunk Security Profile\* Non Secure SIP Trunk Profile  
 Rerouting Calling Search Space < None >  
 Out-Of-Dialog Refer Calling Search Space < None >  
 SUBSCRIBE Calling Search Space < None >  
 SIP Profile\* Standard SIP Profile - OPTIONS Enable [View Details](#)  
 DTMF Signaling Method\* No Preference

**Normalization Script**  
Normalization Script < None >  
 Enable Trace

	Parameter Name	Parameter Value
1		

**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

Save Delete Reset Add New

## 4. Route Group

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

Figure 48: Route Group

The screenshot displays the Cisco Unified CM Administration web interface for configuring a Route Group. The page title is "Route Group Configuration". At the top, there is a navigation menu with options: System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. Below the navigation menu, there are three buttons: Save, Delete, and Add New. The main configuration area is divided into several sections:

- Status:** Shows "Status: Ready".
- Route Group Information:** Includes "Route Group Name\*" (Rewa\_SBC) and "Distribution Algorithm\*" (Circular).
- Route Group Member Information:** Contains a "Find Devices to Add to Route Group" section with a search box, a "Find" button, and a list of available devices (CUBE, Rewa, USTX-SBCLAB01). It also has a "Port(s)" dropdown set to "All" and an "Add to Route Group" button.
- Current Route Group Members:** Shows "Selected Devices (ordered by priority)\*" (Rewa (All Ports)) and "Removed Devices\*\*\*". A "Reverse Order of Selected Devices" button is present.
- Route Group Members:** A section at the bottom with a "Save" button and a link to "Rewa".

At the very bottom of the page, there are three buttons: Save, Delete, and Add New.

## 5. Route List

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

Figure 49: Route List

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

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

### Route List Configuration

Save Delete Copy Reset Apply Config Add New

---

**Status**

Status: Ready

---

**Route List Information**

Registration: Registered with Cisco Unified Communications Manager UCM12.vo.sonusnet.com  
IPv4 Address: 10.35.180.111

Device is trusted

Name\*

Description

Cisco Unified Communications Manager Group\*

Enable this Route List (change effective on Save; no reset required)

Run On All Active Unified CM Nodes

---


**Route List Member Information**

Selected Groups\*\*

Removed Groups\*\*\*

---

**Route List Details**

 Rewa\_SBC

Save Delete Copy Reset Apply Config Add New

**i** \*- indicates required item.  
**i** \*\*Ordered by highest priority  
**i** \*\*\*Will be removed from Route List when you click Save

## 6. Route Pattern

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

Figure 50: Route Pattern

The screenshot shows the 'Route Pattern Configuration' page in Cisco Unified CM Administration. The page is titled 'Route Pattern Configuration' and includes a navigation menu at the top with options like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, Bulk Administration, and Help. Below the navigation menu, there are icons for Save, Delete, Copy, and Add New. The main configuration area is divided into three sections: Status, Pattern Definition, and Calling Party Transformations.

**Status:** Status: Ready

**Pattern Definition:**

- Route Pattern\*: 49!
- Route Partition: < None >
- Description: PlusNet
- Numbering Plan: -- Not Selected --
- Route Filter: < None >
- MLPP Precedence\*: Default
- Apply Call Blocking Percentage
- Resource Priority Namespace Network Domain: < None >
- Route Class\*: Default
- Gateway/Route List\*: Rewa\_SBC (Edit)
- Route Option:
  - Route this pattern
  - Block this pattern No Error
- Call Classification\*: OffNet
- External Call Control Profile: < None >
- Allow Device Override  Provide Outside Dial Tone  Allow Overlap Sending  Urgent Priority
- Require Forced Authorization Code
- Authorization Level\*: 0
- Require Client Matter Code

**Calling Party Transformations:**

- Use Calling Party's External Phone Number Mask
- Calling Party Transform Mask: [Empty]
- Prefix Digits (Outgoing Calls): [Empty]
- Calling Line ID Presentation\*: Default
- Calling Name Presentation\*: Default
- Calling Party Number Type\*: Cisco CallManager
- Calling Party Numbering Plan\*: Cisco CallManager

Figure 51: Route Pattern1

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

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

### Route Pattern Configuration

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
<input type="text" value="-- Not Selected --"/> ▾	<input type="text" value="&lt; Not Exist &gt;"/>	<input type="text"/>

## Ribbon SBC 1000/2000 Configuration

The following configuration steps provide an example of how to configure the Ribbon SBC 1000/2000 to interoperate with Skype 2015 and Virgin Media SIP Trunk:

1. [SIP Profile](#)
2. [SIP Server](#)
3. [Media System](#)
4. [Media Profiles](#)
5. [Media List](#)
6. [Remote Authorization Tables](#)
7. [Signaling Groups](#)
8. [Transformation](#)
9. [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 the Ribbon 1000/2000 used for this configuration effort.

**Figure 52: PlusNet SIP Profile**

The screenshot displays the configuration for the PlusNet SIP Profile, organized into six main sections:

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

**Figure 53: CUCM 12.0.1 SIP Profile**

Description:

Session Timer		MIME Payloads	
Session Timer	<input type="text" value="Enable"/>	ELIN Identifier	<input type="text" value="LOC"/>
Minimum Acceptable Timer	<input type="text" value="600"/> * secs [90..7200]	PIDF-LO Passthrough	<input type="text" value="Enable"/>
Offered Session Timer	<input type="text" value="3600"/> * secs [90..7200]	Unknown Subtype Passthrough	<input type="text" value="Disable"/>
Terminate On Refresh Failure	<input type="text" value="False"/>		

Header Customization		Options Tags	
FQDN in From Header	<input type="text" value="Disable"/>	100rel	<input type="text" value="Supported"/>
FQDN in Contact Header	<input type="text" value="Disable"/>	Path	<input type="text" value="Not Present"/>
Send Assert Header	<input type="text" value="Trusted Onl"/>	Timer	<input type="text" value="Supported"/>
SBC Edge Diagnostics Header	<input type="text" value="Enable"/>	Update	<input type="text" value="Supported"/>
Trusted Interface	<input type="text" value="Enable"/>		
UA Header	<input type="text" value="Ribbon SBC Edge"/>		
Calling Info Source	<input type="text" value="RFC Standard"/>		
Diversion Header Selection	<input type="text" value="Last"/>		
Record Route Header	<input type="text" value="RFC 3261 Standard"/>		

Timers		SDP Customization	
Transport Timeout Timer	<input type="text" value="5000"/> ms [5000..32000]	Send Number of Audio Channels	<input type="text" value="False"/>
Maximum Retransmissions	<input type="text" value="RFC Standa"/>	Connection Info in Media Section	<input type="text" value="True"/>
Redundancy Retry Timer	<input type="text" value="180000"/> ms [5000..180000]	Origin Field Username	<input type="text" value="SBC"/> <small>default: SBC</small>
<b>RFC Timers</b>		Session Name	<input type="text" value="VoipCall"/> <small>default: VoipCall</small>
Timer T1	<input type="text" value="500"/> ms [100..10000]	Digit Transmission Preference	<input type="text" value="RFC 2833/Voice"/>
Timer T2	<input type="text" value="4000"/> ms [1000..80000](>= T1)	SDP Handling Preference	<input type="text" value="Legacy Audio/F"/>
Timer T4	<input type="text" value="5000"/> ms [1000..100000]		
Timer D	<input type="text" value="32000"/> ms [5000..640000]		
Timer B	<input type="text" value="32000"/> ms		
Timer F	<input type="text" value="32000"/> ms		
Timer H	<input type="text" value="32000"/> ms (64*TimerT1)		
Timer J	<input type="text" value="4000"/> ms [4000..640000]		

Figure 54: Fax SIP Profile



▼
Tenor-Fax

Description

#### Session Timer

Session Timer  ▼

Minimum Acceptable Timer  \* secs (90..7200)

Offered Session Timer  \* secs (90..7200)

Terminate On Refresh Failure  ▼

#### MIME Payloads

ELIN Identifier  ▼

PIDF-LO Passthrough  ▼

Unknown Subtype Passthrough  ▼

#### 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  ▼

#### Options Tags

100rel  ▼

Path  ▼

Timer  ▼

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)( $\geq T1$ )

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  ▼

## 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 55: PlusNet SIP Servers**

The screenshot shows the PlusNet SIP Server configuration interface. At the top, it displays 'PlusNet' and the date 'January 02, 2020'. Below this is a table with columns: Host / Domain, Server Lookup, Port, Protocol, Display Counters, and Primary Key. The table contains one entry: sipconnect01.ipfonie.de, IP/FQDN, 5060, TCP, Counters, and 2. Below the table are four configuration panels: Server Host, Transport, Remote Authorization and Contacts, and Connection Reuse. The Server Host panel includes fields for Server Lookup (IP/FQDN), Priority (1), Host FQDN/IP (x.x.x.x), Host IP Version (IPv4), Port (5060), and Protocol (TCP). The Transport panel has a Monitor dropdown set to None. The Remote Authorization and Contacts panel includes Remote Authorization Table (PlusNet), Contact Registrant Table (None), Retry Non-State Nonce (True), Authorization on Refresh (True), and Session URI Validation (Liberal). The Connection Reuse panel includes Reuse (True), Sockets (4), and Reuse Timeout (Forever).

**Figure 56: CUCM SIP Server**

The screenshot shows the CUCM SIP Server configuration interface. At the top, it displays 'CUCM 12.05' and the date 'January 02, 2020'. Below this is a table with columns: Host / Domain, Server Lookup, Port, Protocol, Display Counters, and Primary Key. The table contains one entry: 10.35.180.112, IP/FQDN, 5060, UDP, Counters, and 1. Below the table are three configuration panels: Server Host, Transport, and Remote Authorization and Contacts. The Server Host panel includes fields for Server Lookup (IP/FQDN), Priority (1), Host FQDN/IP (10.35.180.112), Port (5060), and Protocol (UDP). The Transport panel has a Monitor dropdown set to None. The Remote Authorization and Contacts panel includes Remote Authorization Table (None), Contact Registrant Table (None), and Session URI Validation (Liberal). An 'Apply' button is located at the bottom right of the configuration area.

**Figure 57: Fax SIP Server**

### 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 58: Media System

**Media System Configuration**

Upload Music File

**Port Range**

Start Port: 16384 \* [1024..32767]

Number of Port Pairs: 600 \* [1..4800]

Regular Call Media Port Range: 16384-17584

ICE Call Media Port Range: Not activated

**Music on Hold**

Music on Hold Source: File

Current Music File: Not Installed

Echo Cancellor Type Option: Standard

Echo Cancel NLP Option: Disabled

Send STUN Packets: Disabled

### 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 59: PlusNet Media Profile

**Voice Codec Configuration**

Description: PlusNet G711A

Codec: G.711 A-Law

Payload Size: 20 ms

**Voice Codec Configuration**

Description: PlusNet G.711 u

Codec: G.711 μ-Law

Payload Size: 20 ms

**Voice Codec Configuration**

Description

Codec

Rate 64000 b/s

Payload Size 20 ms

**Voice Codec Configuration**

Description

Codec

Payload Size  ms

**Fax Codec Configuration**

Description

Codec T.38 Fax

Maximum Rate  b/s

Signaling Packet Redundancy  [0..7]

Payload Packet Redundancy  [0..3]

Error Correction Mode

Training Confirmation Procedure

Fallback to Passthrough

Super G3 to G3 Fallback

**Figure 60:** CUCM Media Profile

**Voice Codec Configuration**

Description

Codec

Payload Size  ms

**Figure 61:** Fax Media Profile

### Voice Codec Configuration

Description

Codec

Payload Size  ms

### Voice Codec Configuration

Description

Codec

Payload Size  ms

### Fax Codec Configuration

Description

Codec **T.38 Fax**

Maximum Rate  b/s

Signaling Packet Redundancy  [0..7]

Payload Packet Redundancy  [0..3]

Error Correction Mode

Training Confirmation Procedure

Fallback to Passthrough

Super G3 to G3 Fallback

## 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 62: PlusNet Media List

The screenshot shows the 'PlusNet Media List' configuration window. At the top, the 'Description' field contains 'PlusNet Media List'. Below it is a 'Media Profiles List' containing two entries: 'PlusNet G711A' and 'T.38 PlusNet'. To the right of this list are four buttons: 'Up', 'Down', 'Add/Edit', and 'Remove'. Below the list, there are several configuration fields: 'SDES-SRTP Profile' is set to 'None' with a note 'Associated SIP SG Listen Ports should be TLS only.'; 'DTLS-SRTP Profile' is also set to 'None'; 'Media DSCP' is set to '46' with a range indicator '\* [0..63]'; 'RTCP Mode' is set to 'RTCP'; 'Dead Call Detection' is set to 'Disabled'; and 'Silence Suppression' is set to 'Disabled'.

The screenshot shows the 'CUCM Media List' configuration window, divided into three sections. The 'Gain Control' section has 'Receive Gain' and 'Transmit Gain' both set to '0' with a range of '[-14..+6] dB'. The 'Digit Relay' section has 'Digit (DTMF) Relay Type' set to 'RFC 2833' and 'Digit Relay Payload Type' set to '101' with a range of '[96..127]'. The 'Passthrough/Tone Detection' section has 'Modem Passthrough' set to 'Enabled', 'Fax Passthrough' set to 'Disabled', 'CNG Tone Detection' set to 'Disabled', and 'Fax Tone Detection' set to 'Enabled'. It also has 'DTMF Signal to Noise' set to '0' with a range of '[-3..+6] dB' and 'DTMF Minimum Level' set to '-38' with a range of '[-48..-14] dBm0'.

Figure 63: CUCM Media List

**CUCM Media List**

Description: CUCM Media List

Media Profiles List: CUCM G711A

Up, Down, Add/Edit, Remove

SDES-SRTP Profile: None *Associated SIP SG Listen Ports should be TLS only.*

DTLS-SRTP Profile: None

Media DSCP: 46 \* [0..63]

RTCP Mode: RTCP

Dead Call Detection: Disabled

Silence Suppression: Enabled

**CUCM Media List**

**Gain Control**

Receive Gain: 0 [-14..+6] dB

Transmit Gain: 0 [-14..+6] dB

**Digit Relay**

Digit (DTMF) Relay Type: RFC 2833

Digit Relay Payload Type: 101 [96..127]

**Passthrough/Tone Detection**

Modem Passthrough: Enabled

Fax Passthrough: Enabled

CNG Tone Detection: Disabled

Fax Tone Detection: Enabled

DTMF Signal to Noise: 0 [-3..+6] dB

DTMF Minimum Level: -38 [-48..-14] dBm0

**Figure 64: Fax Media List**



Description

Media Profiles List

- Tenor G729
- Tenor G711A
- Tenor T.38

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. Remote Authorization Tables

Remote Authorization Tables and their entries contain information used to respond to request message challenges by an upstream server. The Remote Authorization Tables on this page appear as options in Creating and Modifying Entries in the SIP Servers (For additional information about Remote Authorization Tables, see the Ribbon online SBC 1000/2000 documentation).

Select **Settings > SIP > Remote Authorization Tables** to access the Remote Authorization Tables configuration screen.

Figure 65: Remote Authorization Table

Realm	<input type="text" value="ipfonie.de"/>
Authentication ID	<input type="text" value="107530375765"/> *
Password Setting	<input type="text" value="Use Current"/> ▼
From URI User Match	<input type="text" value="Regex"/> ▼
Match Regex	<input type="text" value=".*"/>

## 7. Signaling Groups

Signaling Groups allow telephony channels to be grouped for routing and shared configuration. These groups are the entity to which calls are routed, and the location from which Call Routes are selected. These are also the location from which Tone Tables and Action Sets are selected. 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 66: PlusNet Signaling Group

Description	<input type="text" value="To/From PlusNet"/>
Admin State	<input type="text" value="Enabled"/> ▼
Service Status	Up

SIP Channels and Routing	
Action Set Table	<input type="text" value="None"/> ▼
Call Routing Table	<input type="text" value="From PlusNet"/> ▼
No. of Channels	<input type="text" value="60"/> * [1..960]
SIP Profile	<input type="text" value="PlusNet SIP Profile"/> ▼
SIP Mode	<input type="text" value="Basic Call"/> ▼
Agent Type	<input type="text" value="Back-to-Back User Agent"/> ▼
Interop Mode	<input type="text" value="Standard"/> ▼

Media Information	
Supported Audio/Fax Modes	<input type="text" value="DSP"/> ▲ <input type="text" value="Proxy"/> ▼ <input type="text" value="Direct"/> ▼ <input type="button" value="Add/Edit"/> * <input type="button" value="Remove"/>
Supported Video/Application Modes	Disabled
Media List ID	<input type="text" value="PlusNet Media List"/> ▼
Play Ringback	<input type="text" value="Auto on 180"/> ▼

SIP Server Table	Test_PlusNet	▼
Load Balancing	Priority: Register All	▼
Channel Hunting	Most Idle	▼
Notify Lync CAC Profile	Disable	▼
Challenge Request	Disable	▼
Outbound Proxy IP/FQDN		
Outbound Proxy Port	5060	[1..65535]
No Channel Available Override	34: No Circuit/Channel Available	▼
Call Setup Response Timer	255	[180..750] secs
Call Proceeding Timer	180	[24..750] secs
QoE Reporting	Disabled	▼
Use Register as Keep Alive	Enable	▼

Tone Table	Default Tone Table	▼
Play Congestion Tone	Disable	▼
Early 183	Disable	▼
Allow Refresh SDP	Enable	▼
Music on Hold	Disabled	▼
RTCP Multiplexing	Disable	▼

Mapping Tables	
SIP To Q.850 Override Table	Default (RFC4497) ▼
Q.850 To SIP Override Table	Default (RFC4497) ▼
Pass-thru Peer SIP Response Code	Enable ▼

SIP IP Details	
Signaling/Media Source IP	Ethernet 3 IP (216.110.2.220) ▼
Signaling DSCP	40 [0..63]
NAT Traversal	
ICE Support	Disabled ▼
Static NAT - Outbound	
Outbound NAT Traversal	None ▼
Static NAT - Inbound	
Detection	Disabled ▼

Listen Ports	Federated IP/FQDN						
<p>Total 1 SIP Listen Port Row</p> <table border="1"> <thead> <tr> <th>Port</th> <th>Protocol</th> <th>TLS Profile ID</th> </tr> </thead> <tbody> <tr> <td>5060</td> <td>TCP</td> <td>N/A</td> </tr> </tbody> </table>	Port	Protocol	TLS Profile ID	5060	TCP	N/A	<p>Total 0 SIP Federated IP Rows</p> <p>-- Table is empty --</p>
Port	Protocol	TLS Profile ID					
5060	TCP	N/A					
<p>Message Manipulation Disabled ▼</p>							

Figure 67: CUCM Signaling Group

Description	CUCM 12.0.1
Admin State	Enabled ▼
Service Status	Up

SIP Channels and Routing	
Action Set Table	None ▼
Call Routing Table	From_CUCM 14.0.1 ▼
No. of Channels	60 [1..960]
SIP Profile	CUCM 12.0.1 ▼
SIP Mode	Basic Call ▼
Agent Type	Back-to-Back User Agent ▼
Interop Mode	Standard ▼

Media Information		
Supported Audio/Fax Modes	DSP Proxy Direct	Add/Edit Remove
Supported Video/Application Modes	Disabled	
Media List ID	Default Media List ▼	
Play Ringback	Auto on 180 ▼	

SIP Server Table	CUCM 12.0.1		Tone Table	Default Tone Table	
Load Balancing	Round Robin		Play Congestion Tone	Disable	
Channel Hunting	Most Idle		Early 183	Disable	
Notify Lync CAC Profile	Disable		Allow Refresh SDP	Enable	
Challenge Request	Disable		Music on Hold	Disabled	
Outbound Proxy IP/FQDN			RTCP Multiplexing	Disable	
Outbound Proxy Port	5060	[1..65535]			
No Channel Available Override	34: No Circuit/Channel Available				
Call Setup Response Timer	255	[180..750] secs			
Call Proceeding Timer	180	[24..750] secs			
QoE Reporting	Disabled				
Use Register as Keep Alive	Enable				

Mapping Tables	
SIP To Q.850 Override Table	Default (RFC4497)
Q.850 To SIP Override Table	Default (RFC4497)
Pass-thru Peer SIP Response Code	Enable

SIP IP Details	
Signaling/Media Source IP	Ethernet 1 IP (10.35.179.136)
Signaling DSCP	40 [0..63]
NAT Traversal	
ICE Support	Disabled
Static NAT - Outbound	
Outbound NAT Traversal	None
Static NAT - Inbound	
Detection	Disabled

Listen Ports			
Total 2 SIP Listen Port Rows			
Port	Protocol	TLS Profile ID	
5060	UDP	N/A	
5060	TCP	N/A	

Federated IP/FQDN		
Total 1 SIP Federated IP Row		
IP/FQDN	Netmask/Prefix	
10.35.180.111	255.255.255.255	

Figure 68: Fax Signaling Group

Description	To/From Tenor-Fax
Admin State	Enabled
Service Status	Up

SIP Channels and Routing	
Action Set Table	None
Call Routing Table	From Tenor-Fax
No. of Channels	60 [1..960]
SIP Profile	Tenor-Fax
SIP Mode	Basic Call
Agent Type	Back-to-Back User Agent
Interop Mode	Standard

Media Information	
Supported Audio/Fax Modes	DSP Proxy Direct
Supported Video/Application Modes	Disabled
Media List ID	Tenor-Fax Media List
Play Ringback	Auto on 180

SIP Server Table	Tenor_Fax		Tone Table	Default Tone Table	
Load Balancing	Round Robin		Play Congestion Tone	Disable	
Channel Hunting	Most Idle		Early 183	Disable	
Notify Lync CAC Profile	Disable		Allow Refresh SDP	Enable	
Challenge Request	Disable		Music on Hold	Disabled	
Outbound Proxy IP/FQDN			RTCP Multiplexing	Disable	
Outbound Proxy Port	5060	[1..65535]	<b>Mapping Tables</b>		
No Channel Available Override	34: No Circuit/Channel Available		SIP To Q.850 Override Table	Default (RFC4497)	
Call Setup Response Timer	255	[180..750] secs	Q.850 To SIP Override Table	Default (RFC4497)	
Call Proceeding Timer	180	[24..750] secs	Pass-thru Peer SIP Response Code	Enable	
QoE Reporting	Disabled				
Use Register as Keep Alive	Enable				
Forked Call Answered Too Soon	Disable				

<b>SIP IP Details</b>	
Signaling/Media Source IP	Ethernet 1 IP (10.35.179.136)
Signaling DSCP	40 * [0..63]
<b>NAT Traversal</b>	
ICE Support	Disabled
<b>Static NAT - Outbound</b>	
Outbound NAT Traversal	None
<b>Static NAT - Inbound</b>	
Detection	Disabled

Listen Ports			
Total 2 SIP Listen Port Rows			
Port	Protocol	TLS Profile ID	
5060	UDP	N/A	
5060	TCP	N/A	

Federated IP/FQDN		
Total 1 SIP Federated IP Row		
IP/FQDN	Netmask/Prefix	
10.35.137.106	255.255.255.255	

## 8. 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 a SIP address (URI). Every entry in a Call Routing Table requires a Transformation Table.

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

**Figure 69: PlusNet Transformation**

From PlusNet								January 02, 2020 11:26:53
Total 6 Transformation Entry Rows								Filter...
Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key	
▶	Called Address/Number	\+(4932211057459)	Called Address/Number	\1	Optional (Match One)	Entry ID 4	4	
▶	Called Address/Number	\+(496029995243)	Called Address/Number	\1	Optional (Match One)	Entry ID 6	6	
▶	Calling Address/Number	(*)	Calling Address/Number	\1	Optional (Match One)	From_PlusNet	2	
▶	Calling Address/Number	\+(4991147726289)	Calling Address/Number	\1	Optional (Match One)	Entry ID 5	5	
▶	Called Address/Number	\+(*)	Called Address/Number	\1	Optional (Match One)	From_PlusNet	1	
▶	Called Address/Number	\+(4991147726289)	Called Address/Number	\1	Optional (Match One)	4991147726289	3	

**Figure 70: CUCM Transformation**

From_CUCM 14.0.1								January 02, 2020 11:28:04
Total 2 Transformation Entry Rows								
Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key	
▶	Called Address/Number	(*)	Called Address/Number	+1	Optional (Match One)	From_CUCM	1	
▶	Calling Address/Number	(*)	Calling Address/Number	+1	Optional (Match One)	From_CUCM	2	

**Figure 71: Fax-Tenor Transformation**

From Tenor-Fax								January 02, 2020 11:28:59
Total 5 Transformation Entry Rows								
Admin State	Input Field Type	Input Field Value	Output Field Type	Output Field Value	Match Type	Description	Primary Key	
▶	Calling Address/Number	(4932211057459)	Calling Address/Number	+1	Optional (Match One)	Entry ID 3	3	
▶	Called Address/Number	(496029995243)	Called Address/Number	+1	Optional (Match One)	Entry ID 5	5	
▶	Called Address/Number	(499113929210)	Called Address/Number	+1	Optional (Match One)	From_Tenr-Fax	1	
▶	Called Address/Number	(4932166985357)	Called Address/Number	+1	Optional (Match One)	Entry ID 4	4	
▶	Called Address/Number	1(*)	Called Address/Number	+1\1	Optional (Match One)	TO US FAX	2	



## 9. 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 flexible configuration of which calls will 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 72: PlusNet Call Routing

The screenshot shows the configuration interface for a PlusNet Call Routing table. It is organized into four distinct sections:

- Route Details:** Includes fields for Description (To CUCM 14.0.1), Admin State (Enabled), Route Priority (1), Call Priority (Normal), Number/Name Transformation Table (From PlusNet), and Time of Day Restriction (None).
- Destination Information:** Includes Destination Type (Normal), Message Translation Table (None), Cause Code Reroutes (None), Cancel Others upon Forwarding (Disabled), Fork Call (No), Destination Signaling Groups (a list containing (SIP) CUCM 12.5 and (SIP) CUCM 12.0.1 with Up, Down, Add/Edit, and Remove buttons), and Enable Maximum Call Duration (Disabled).
- Media:** Includes Audio/Fax Stream Mode (DSP), Video/Application Stream Mode (Disabled), Media Transcoding (Enabled), and Media List (None).
- Quality of Service:** Includes Quality Metrics Number of Calls (10), Quality Metrics Time Before Retry (10), Min. ASR Threshold (0), Enable Min MOS Threshold (Disable), Enable Max. R/T Delay (Enable), Max. R/T Delay (65535), Enable Max. Jitter (Enable), and Max. Jitter (3000).

Figure 73: CUCM 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**

**Quality of Service**

Audio/Fax Stream Mode  ▼

Video/Application Stream Mode  ▼

Media Transcoding  ▼

Media List  ▼

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 74: Fax 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) To/From PlusNet	▲
-----------------------	---

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]

# Interoperability Test Results

The following table provides test results for interoperability compliance testing between Ribbon SBC 1000/2000 and CUCM

**Table 4:** Interoperability Compliance Test Results

Test Number	Test Scenario	Setup / Test Results	Status	Comment
4.1	Registration and authentication (registration mode)	The PBX is able to execute the correct resolution of the DNS SRV record	Pass	
4.3	Basic call	With the basic call tests, the standard call scenarios and the CLIP/CLIR features are tested. <ul style="list-style-type: none"> <li>Only en-bloc dialing is supported, overlap sending is not possible.</li> </ul>	Pass	
4.3.1	Normal call	Outgoing call from PBX to PSTN <ul style="list-style-type: none"> <li>En-bloc dialing</li> <li>Local area call (without area code); area code must be set by the PBX</li> <li>Setting of the correct calling number with all available telephone number blocks</li> <li>If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the user's location. This is very important in case of emergency calls, because Plusnet uses the number of the PAI header to route the emergency call to the proper emergency call center.</li> </ul>	Pass	
4.3.1.1	Normal call	The geographic number in the PAI header corresponds to the location 11 of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.1.2	Normal call	Display of A-number in B-party CLIP (national PSTN)	Pass	
4.3.1.3	Normal call	Display of A-number in B-party CLIP (international PSTN)	Pass	
4.3.1.4	Normal call	Display of A-number in B-party CLIP (mobile)	Pass	
4.3.1.5	Normal call	Call to mobile Outgoing call to mobile => mobile phone turned off	Pass	
4.3.1.6	Normal call	Suppression of A-number => CUR	Pass	
4.3.1.7	Normal call	Outgoing call from analog extension	Pass	
4.3.1.8	Normal call	Outgoing call (> 5 min.) => PSTN	Pass	Hold connection for 5 minutes => RTP still correct? Yes
4.3.2	Normal call	Incoming call from PSTN (national) => PBX <ul style="list-style-type: none"> <li>Test all available telephone number blocks</li> </ul>	Pass	
4.3.2.1	Normal call	Display of A-number => CLIP	Pass	
4.3.2.2	Normal call	Incoming call from mobile => PBX Display of A-number => CLIP	Pass	
4.3.2.3	Normal call	Suppression of A-number => CLIR	Pass	
4.3.3	Normal call	Two simultaneous outgoing/incoming calls	Pass	
4.3.4	Normal call	Enabled feature DND (do not disturb)	Pass	

4.3.5	Normal call	Test call with codec G.711	Pass	
4.3.6	Normal call	Test call with codec G.722 (only SIP <=> SIP)	Fail	PlusNet didn't support G.722
4.3.7	Normal call	Test call with codec G.729	Pass	
4.3.2.8	Clip No Screening	Outgoing call from PBX to PSTN <ul style="list-style-type: none"> <li>• With feature Clip No Screening</li> <li>• Test with several different A-numbers</li> <li>• If two or more locations with different area codes are assigned to one SIP trunk, the number in the PAI header has to be the geographic number which belongs to the users location. This is very important in case of emergency calls, because PlusNet uses the number</li> </ul>	Pass	
4.3.2.8.1	Clip No Screening	Despite Clip No Screening the geographic number in the PAI header corresponds to the location of the user	Pass	Is it possible to configure several PAI per SIP trunk? Yes
4.3.2.8.2	Clip No Screening	Display of A-number (NoSClip) at B-party (PSTN)	Pass	
4.3.2.8.3	Clip No Screening	Display of A-number (NoSClip) at B-party (international PSTN; depending on the destination carrier, the NoSClip telephone number may not be displayed in this case!)	Pass	
4.3.2.8.3	Clip No Screening	Call made from a PSTN line to an IP-PBX line with call forward to a line within the same IP-PBX, Answer Call. <ul style="list-style-type: none"> <li>• Either party terminates call.</li> </ul>	Pass	Does the IP-PBX has configuration settings to send SIP status 181 messages to the soft switch? Yes
4.3.2.8.4	Clip No Screening	Display of A-number (NoSClip) at B-party (mobile)	Pass	
4.3.3.9	Special call situations	Outgoing call PBX => PSTN <ul style="list-style-type: none"> <li>• Call is rejected by B-party</li> </ul>	Pass	
4.3.3.10	Special call situations	Outgoing call PBX=> PSTN <ul style="list-style-type: none"> <li>• B-party does not answer; clearing after timer</li> </ul>	Pass	
4.3.3.11	Special call situations	Outgoing call PBX => PSTN <ul style="list-style-type: none"> <li>• B-party busy; busy tone</li> </ul>	Pass	
4.3.3.12	Special call situations	Outgoing call PBX=> PSTN <ul style="list-style-type: none"> <li>• A-party hangs up before call is established (cancel)</li> </ul>	Pass	
4.3.3.13	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• Call is rejected by PBX party</li> </ul>	Pass	
4.3.3.14	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• PBX party does not answer; clearing after timer</li> </ul>	Pass	
4.3.3.15	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• PBX party busy; busy tone</li> </ul>	Pass	

4.3.3.16	Special call situations	Incoming call PSTN => PBX <ul style="list-style-type: none"> <li>• A-party hangs up before call is established (cancel)</li> </ul>	Pass	
4.3.4.17	Call clearing	Incoming / outgoing call; clearing after established call. Correct clearing on both sides <ul style="list-style-type: none"> <li>• PBX party hangs up</li> <li>• PSTN party hangs up</li> </ul>	Pass	
4.3.4.18	Call clearing	Interrupting the network connection of the SIP terminal device during a call <ul style="list-style-type: none"> <li>• Call should be cleared correctly</li> </ul>	Pass	
4.4.19	Hold	PBX => PSTN and PSTN => PBX <ul style="list-style-type: none"> <li>• Test call in both directions</li> </ul>	Pass	
4.4.19.1	Hold	Putting an external call on hold in the PBX	Pass	
4.4.19.2	Hold	If applicable, MoH (music on hold) at A-party (PSTN)	Pass	
4.4.19.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.19.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.4.20	Hold	PBX => PSTN and PSTN => PBX <ul style="list-style-type: none"> <li>• Test call in both directions</li> </ul>	Pass	
4.4.20.1	Hold	Putting an external call on hold in the PSTN	Pass	
4.4.20.2	Hold	If applicable, MoH (music on hold) at A-party (PBX)	Pass	
4.4.20.3	Hold	HOLD RETRIEVE: retrieving the external call	Pass	
4.4.20.4	Hold	Clearing the connection of the A-party while it is put on hold	Pass	
4.5.21	Call transfer	Internal call is transferred to external party: internal => PBX => external	Pass	
4.5.21.1	Call transfer	Call transfer from PBX party => PSTN party with announcement (attendant transfer)	Pass	
4.5.21.2	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer)	Pass	
4.5.21.3	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) <ul style="list-style-type: none"> <li>• Call is rejected or not answered</li> </ul>	Pass	
4.5.21.4	Call transfer	Call transfer from PBX party => PSTN party without announcement (blind transfer) <ul style="list-style-type: none"> <li>• PSTN party busy</li> </ul>	Pass	
4.5.22	Call transfer	Transferred call from external party to PBX: PSTN => PBX	Pass	
4.5.22.1	Call transfer	Call transfer from PSTN => PBX party with announcement (attendant transfer)	Pass	
4.5.22.2	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer)	Pass	
4.5.22.3	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) <ul style="list-style-type: none"> <li>• Call is rejected or not answered</li> </ul>	Pass	
4.5.22.4	Call transfer	Call transfer from PSTN => PBX party without announcement (blind transfer) <ul style="list-style-type: none"> <li>• PBX party busy</li> </ul>	Pass	
4.5.23	Call transfer	Call from external party transferred to another external party: external => PBX => external	Pass	
4.5.23.1	Call transfer	PSTN => PBX party => PSTN with announcement (attendant transfer)	Pass	
4.5.23.2	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer)	Pass	

4.5.23.3	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) <ul style="list-style-type: none"> <li>Call is rejected or not answered</li> </ul>	Pass	
4.5.23.4	Call transfer	PSTN => PBX party => PSTN without announcement (blind transfer) <ul style="list-style-type: none"> <li>PSTN party busy</li> </ul>	Pass	
4.6.24	Call diversion	PBX party CFU to external party (PSTN)	Pass	
4.6.24.1	Call diversion	Internal call (CFU) => PSTN	Pass	
4.6.24.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.3	Call diversion	A-party clears in ring phase	Pass	
4.6.24.4	Call diversion	External => PBX party (CFU) => PSTN	Pass	
4.6.24.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.24.6	Call diversion	A-party clears in ring phase	Pass	
4.6.25	Call diversion	PBX party CFNR to external party	Pass	
4.6.25.1	Call diversion	Internal call (CFNR) => PSTN	Pass	
4.6.25.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.3	Call diversion	A-party clears in ring phase	Pass	
4.6.25.4	Call diversion	External => PBX party (CFNR) => PSTN	Pass	
4.6.25.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.25.6	Call diversion	A-party clears in ring phase	Pass	
4.6.26	Call diversion	PBX party CFB to external party	Pass	
4.6.26.1	Call diversion	Internal call (CFB) => PSTN	Pass	
4.6.26.2	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.3	Call diversion	A-party clears in ring phase	Pass	
4.6.26.4	Call diversion	External => PBX party (CFB) => PSTN	Pass	
4.6.26.5	Call diversion	PSTN party busy, rejects call, does not answer	Pass	
4.6.26.6	Call diversion	A-party clears in ring phase	Pass	
4.6.27	Call diversion	External party (PSTN, mobile, etc.) CFU to PBX party:	Pass	
4.6.27.1	Call diversion	External (CFU) => PBX party	Pass	

4.6.27.2	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.3	Call diversion	A-party clears in ring phase	Pass	
4.6.27.4	Call diversion	External (CFNR) => PBX party	Pass	
4.6.27.5	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.27.6	Call diversion	A-party clears in ring phase	Pass	
4.6.28	Call diversion	External party (PSTN, mobile, etc.) CFB to PBX party:	Pass	
4.6.28.1	Call diversion	PBX party busy, rejects call, does not answer	Pass	
4.6.28.2	Call diversion	A-party clears in ring phase	Pass	
4.6.29	Call diversion	Call deflection: diversion during the ring phase	Pass	
4.6.29.1.1	Call diversion	Internal call PBX party CD => PBX party	Pass	
4.6.29.1.2	Call diversion	PBX party busy	Pass	
4.6.29.1.3	Call diversion	PBX party does not answer	Pass	
4.6.29.1.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.2.1	Call diversion	External call to PBX party CD => PBX party	Pass	
4.6.29.2.2	Call diversion	PBX party busy	Pass	
4.6.29.2.3	Call diversion	PBX party does not answer	Pass	
4.6.29.2.4	Call diversion	A-party clears in ring phase	Pass	
4.6.29.3.1	Call diversion	External call to PBX party CD => external PSTN party	Pass	
4.6.29.3.2	Call diversion	PSTN party busy	Pass	
4.6.29.3.3	Call diversion	PSTN party does not answer	Pass	
4.6.29.3.4	Call diversion	A-party clears in ring phase	Pass	
4.7.30	Call waiting	Incoming call during active internal call	Pass	
4.7.30.1	Call waiting	Call waiting tone	Pass	
4.7.30.2	Call waiting	Display of number of waiting party	Pass	
4.7.30.3	Call waiting	Acceptance of waiting call	Pass	
4.7.30.4	Call waiting	Putting the waiting call on hold	Pass	
4.7.30.5	Call waiting	Retrieve of waiting call	Pass	
4.7.30.6	Call waiting	Holding the 2nd call	Pass	
4.7.30.7	Call waiting	Terminating the active call	Pass	
4.7.30.8	Call waiting	Disregarding the waiting call	Pass	

4.7.30.9	Call waiting	Rejecting the waiting call	Pass	
4.8.31	3-party conference	Establishing a conference according to the operating instructions of the PBX Internal - internal - external	Pass	
4.8.31.1	3-party conference	Selecting a party (internal or external); 3rd party is put on hold <ul style="list-style-type: none"> <li>Switching to 3rd party; 2nd party is put on hold</li> <li>Reactivating the conference</li> <li>Clearing one party (internal or external)</li> <li>Terminating the conference</li> </ul>	Pass	
4.8.32	3-party conference	Establishing a conference according to the operating instructions of the PBX internal - external - external	Pass	
4.8.32.1	3-party conference	Selecting a party (external); 3rd party is put on hold <ul style="list-style-type: none"> <li>Switching to 3rd party; 2nd party is put on hold</li> <li>Reactivating the conference</li> <li>Clearing one party (external)</li> <li>Terminating the conference</li> </ul>	Pass	
4.9.33	Pick up	Picking up a call from another extension of the PBX	Pass	
4.10.34	Call list	Entries in the call list <ul style="list-style-type: none"> <li>Incoming from PSTN</li> <li>Incoming from mobile</li> <li>Incoming CLIR</li> <li>Dialing prefix for outside line in the call list</li> </ul>	Pass	
4.10.35	Call list	Call-back from call list <ul style="list-style-type: none"> <li>To PSTN</li> <li>To mobile</li> </ul>	Pass	
4.11.36	DTMF	DTMF support (G.711) <ul style="list-style-type: none"> <li>PSTN =&gt; PBX (SIP terminal device)</li> <li>PSTN =&gt; PBX (analog or system terminal device)</li> <li>PBX (SIP terminal device) =&gt; PSTN</li> <li>PBX (analog or system terminal device) =&gt; PSTN</li> </ul>	Pass	
4.11.37	DTMF	DTMF support (G.729) <ul style="list-style-type: none"> <li>- PSTN =&gt; PBX (SIP terminal device)</li> <li>- PSTN =&gt; PBX (analog or system terminal device)</li> <li>- PBX (SIP terminal device) =&gt; PSTN</li> <li>- PBX (analog or system terminal device) =&gt; PSTN</li> </ul>	Pass	
4.12.38	Fax	Fax reception (G.711 only)	Pass	
4.12.38.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed <ul style="list-style-type: none"> <li>One-page fax</li> <li>Multi-page fax (at least 5 pages)</li> </ul>	Pass	
4.12.39	Fax	Fax sending (G.711 only)	Pass	
4.12.39.1	Fax	Network-side T.38 re-invite rejected by PBX (response 488) or only G.711 codec is confirmed <ul style="list-style-type: none"> <li>One-page fax</li> <li>Multi-page fax (at least 5 pages)</li> </ul>	Pass	
4.12.40	Fax	Fax reception via T.38	Pass	
4.12.40.1	Fax	Re-invite to T.38 by PBX or network <ul style="list-style-type: none"> <li>- One-page fax</li> <li>- Multi-page fax (at least 5 pages)</li> </ul>	Pass	
4.12.41	Fax	Fax sending via T.38 (not possible in conjunction with the encryption option)	Pass	

4.12.41.1	Fax	<p>Re-invite to T.38 by PBX or network</p> <ul style="list-style-type: none"> <li>• T.38-only invites are not supported</li> <li>• One-page fax</li> <li>• Multi-page fax (at least 5 pages)</li> </ul>	Pass	
5.42	Redundancy	<p>Test of redundancy:</p> <ul style="list-style-type: none"> <li>• Only if redundant connection is possible on the PBX side</li> <li>• This requires at least two PBX servers to be online on the SIP trunk in registration mode, or exactly two PBX servers in peering mode.</li> </ul>	Pass	
5.42.1	Redundancy	<ol style="list-style-type: none"> <li>1. Register all available PBX systems the Plusnet SBC.</li> <li>2. Calls from Plusnet =&gt; PBX are routed by round robin procedure</li> <li>3. Deliberately de-register one PBX system =&gt; no more calls are routed to this PBX</li> <li>4. and/or disconnect the PBX system from LAN =&gt; after the register expire period, no more calls are routed to this PBX.</li> <li>5. Calls are only routed to the remaining PBX systems.</li> </ol>	Pass	

## Conclusion

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This Application Notes document describes the steps required to configure the Ribbon SBC 1000/2000 to successfully interoperate with the Cisco CUCM and PlusNet SIP Trunk. All feature and serviceability test cases have been completed. The majority of test cases passed with noted exceptions and observations provided in [Interoperability Test Results](#).