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# Ribbon SBC 1000/2000 V9.0.4 IOT MS Teams Virgin Media SIP Trunk Application Note

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## Interoperable Vendors

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**Microsoft Teams**

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## Document Overview

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This document outlines the configuration best practices for Virgin Media SIP Trunk involving Ribbon SBC 2000 when deployed with Microsoft Teams. This document also provides the configuration snapshot of the interoperability performed between Ribbon's SBC 2000 and MS Teams.

## Scope

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This document provides configuration best practices for deploying Ribbon's SBC 2000 with MS Teams and associated clients. Note that these are configuration best practices and each customer may have unique needs and networks. Ribbon recommends that customers work with network design and deployment engineers to establish the network design which best meets their requirements.

 The SweLite platform is also supported using version 9.0.4 with MS teams.

## Non-Goals

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

## Audience

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This technical document is intended for telecommunications engineers with the purpose of configuring both the Ribbon SBC 2000 and the MS Teams and associated clients.

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

## Pre-Requisites

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The following aspects are required before proceeding with Ribbon SBC 2000 and MS Teams:

- MS Teams needs to create users with the correct tenant, add SBC IP in MS Teams, and provide certificates to Ribbon Team.
- Remote Desktop access to a Windows host is available for installing MS Teams client.
- Install certificates in Ribbon SBC 2000.

## Product and Device Details

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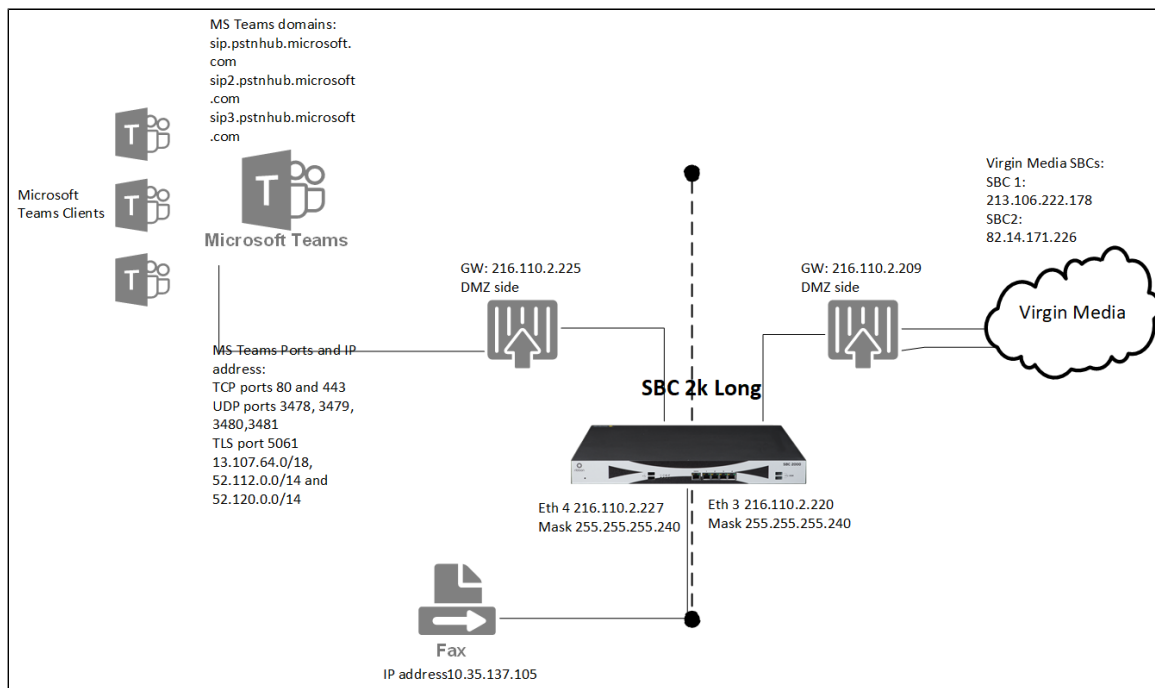
	Equipment /Product	Software Version
<b>Ribbon Communications</b>	SBC 2000	Version 9.0.4
<b>Third-Party Products</b>	MS Teams DR	V.2021.8.9.1 i.USWE2.1
	MS Teams Client	V1.0
	NGT Lite	V1.51
	VentaFax	V7.3.233.582 I

## Network Topology Diagram

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### Interoperability Test Lab Topology (or Call Flow Diagram)

IOT high-level architecture covering call flows and overall topology is depicted below.



## Section-A: SBC 2000 Configuration

### Configuring SBC 2000

The following configuration steps provide an example of how to configure the Ribbon SBC 1000/2000 to interoperate with MS Teams 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. [Message Manipulation](#)
8. [Signaling Groups](#)
9. [Transformation](#)
10. [Call Routing Table](#)
11. [SBC Primary Certificate](#)
12. [Trusted CA Certificates](#)

### 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.

**Figure 1:** Virgin Media SIP Profile

Description: Virgin Media SIP Profile

<b>Session Timer</b> Session Timer: Enable Minimum Acceptable Timer: 600 * secs (90..7200) Offered Session Timer: 3600 * secs (90..7200) Terminate On Refresh Failure: False	<b>MIME Payloads</b> ELIN Identifier: LOC PIDF-LO Passthrough: Enable Unknown Subtype Passthrough: Disable
<b>Header Customization</b> FQDN in From Header: Disable FQDN in Contact Header: Disable Send Assert Header: Always 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	<b>Options Tags</b> 100rel: Supported Path: Not Present Timer: Supported Update: Supported
<b>Timers</b> Transport Timeout Timer: 5000 ms (5000..32000) Maximum Retransmissions: RFC Standard Redundancy Retry Timer: 180000 ms (5000..180000) <hr/> <b>RFC Timers</b> 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)	<b>SDP Customization</b> 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 2: MS Teams 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="Static"/>	100rel: <input type="text" value="Supported"/>
Static Host FQDN/IP[:port]: <input type="text" value="emvirgin.customers.interopdom"/>	Path: <input type="text" value="Not Present"/>
FQDN in Contact Header: <input type="text" value="Static"/>	Timer: <input type="text" value="Supported"/>
Send Assert Header: <input type="text" value="Always"/>	Update: <input type="text" value="Supported"/>
SBC Edge Diagnostics Header: <input type="text" value="Enable"/>	
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 Standard"/>	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"/> default: SBC
<b>RFC Timers</b>	
Timer T1: <input type="text" value="500"/> ms [100..10000]	Session Name: <input type="text" value="VoipCall"/> default: VoipCall
Timer T2: <input type="text" value="4000"/> ms [1000..80000](>= T1)	Digit Transmission Preference: <input type="text" value="RFC 2833/Voice"/>
Timer T4: <input type="text" value="5000"/> ms [1000..100000]	SDP Handling Preference: <input type="text" value="Legacy Audio/F"/>
Timer D: <input type="text" value="32000"/> ms [5000..640000]	
Timer B: 32000 ms	
Timer F: 32000 ms	
Timer H: 32000 ms (64*TimerT1)	
Timer J: <input type="text" value="4000"/> ms [4000..640000]	

Figure 3: Fax SIP Profile

Description:

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

## 2. SIP Server

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

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

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

**Figure 4:** Virgin Media SIP Servers

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	1	Keep Alive Frequency	30 * secs [30..300]
Host FQDN/IP	x.x.x.x *	Recover Frequency	5 * secs [5..300]
Host IP Version	IPv4	Local Username	Anonymous * Local Username of SBC Edge
Port	5060 * [1..65535]	Peer Username	Anonymous * Peer Username of sip server
Protocol	UDP *		

Remote Authorization and Contacts	
Remote Authorization Table	Virgin Media +
Contact Registrant Table	None +
Retry Non-State Nonce	True
Authorization on Refresh	True
Session URI Validation	Liberal

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	2	Keep Alive Frequency	30 * secs [30..300]
Host FQDN/IP	y.y.y.y *	Recover Frequency	5 * secs [5..300]
Host IP Version	IPv4	Local Username	Anonymous * Local Username of SBC Edge
Port	5060 * [1..65535]	Peer Username	Anonymous * Peer Username of sip server
Protocol	UDP *		

Remote Authorization and Contacts	
Remote Authorization Table	Virgin Media +
Contact Registrant Table	None +
Retry Non-State Nonce	True
Authorization on Refresh	True
Session URI Validation	Liberal

Figure 5: MS Teams SIP Server



Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	1	Keep Alive Frequency	30 * secs [30..300]
Host FQDN/IP	sip.pstnhub.microsoft.com *	Recover Frequency	5 * secs [5..300]
Host IP Version	IPv4	Local Username	Anonymous * Local Username of SBC Edge
Port	5061 * [1..65535]	Peer Username	Anonymous * Peer Username of sip server
Protocol	TLS *		
TLS Profile	MS Teams +		
Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None +	Reuse	True
Contact Registrant Table	None +	Sockets	4
Session URI Validation	Liberal	Reuse Timeout	Forever

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	2	Keep Alive Frequency	30 * secs [30..300]
Host FQDN/IP	sip2.pstnhub.microsoft.com *	Recover Frequency	5 * secs [5..300]
Host IP Version	IPv4	Local Username	Anonymous * Local Username of SBC Edge
Port	5061 * [1..65535]	Peer Username	Anonymous * Peer Username of sip server
Protocol	TLS *		
TLS Profile	MS Teams +		
Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None +	Reuse	True
Contact Registrant Table	None +	Sockets	4
Session URI Validation	Liberal	Reuse Timeout	Forever

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	SIP Options
Priority	3	Keep Alive Frequency	30 * secs [30..300]
Host FQDN/IP	sip3.pstnhub.microsoft.com *	Recover Frequency	5 * secs [5..300]
Host IP Version	IPv4	Local Username	Anonymous * Local Username of SBC Edge
Port	5061 * [1..65535]	Peer Username	Anonymous * Peer Username of sip server
Protocol	TLS *		
TLS Profile	MS Teams +		
Remote Authorization and Contacts		Connection Reuse	
Remote Authorization Table	None +	Reuse	True
Contact Registrant Table	None +	Sockets	4
Session URI Validation	Liberal	Reuse Timeout	Forever

Figure 6: Fax SIP Server

Server Host		Transport	
Server Lookup	IP/FQDN	Monitor	None
Priority	1		
Host FQDN/IP	10.35.137.105 *		
Port	5060 * [1..65535]		
Protocol	UDP *		
Remote Authorization and Contacts			
Remote Authorization Table	None		+
Contact Registrant Table	None		+
Session URI Validation	Liberal		

### 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 7: 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	16384-17583		
ICE Call Media Port Range	17585-21184		
Echo Canceller Type Option	Standard		
Echo Cancel NLP Option	Mild		
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 8: Virgin Media Media Profile

**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 9: MS Teams Media Profile

**Voice Codec Configuration**

Description

Codec

Payload Size  ms

**Voice Codec Configuration**

Description

Codec

Payload Size  ms

Figure 10: Fax Media Profile

**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 11:** Virgin Media Media List

Description:

Media Profiles List:

Up

Down \*

Add/Edit

Remove

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

DTLS-SRTP Profile:  +

Media DSCP:  \* [0..63]

RTCP Mode:

Dead Call Detection:

Silence Suppression:

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

Passthrough/Tone Detection	
Modem Passthrough	<input type="text" value="Enabled"/> ▼
Fax Passthrough	<input type="text" value="Enabled"/> ▼
CNG Tone Detection	<input type="text" value="Disabled"/> ▼
Fax Tone Detection	<input type="text" value="Enabled"/> ▼
DTMF Signal to Noise	<input type="text" value="0"/> [-3..+6] dB
DTMF Minimum Level	<input type="text" value="-38"/> [-48..-14] dBm0

Figure 12: MS Teams Media List

Description	<input type="text" value="MS Teams Media List"/>		
Media Profiles List	<input type="text" value="MS Teams G711A"/> <input type="text" value="MS Teams G711U"/>	<input type="button" value="Up"/> <input type="button" value="Down"/> * <input type="button" value="Add/Edit"/> <input type="button" value="Remove"/>	
SDES-SRTP Profile	<input type="text" value="MS Teams"/> ▼	Associated SIP SG Listen Ports should be TLS only. +	
DTLS-SRTP Profile	<input type="text" value="None"/> ▼	+	
Media DSCP	<input type="text" value="46"/> *	* [0..63]	
RTCP Mode	<input type="text" value="RTCP"/> ▼		
Dead Call Detection	<input type="text" value="Disabled"/> ▼		
Silence Suppression	<input type="text" value="Enabled"/> ▼		

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

Passthrough/Tone Detection	
Modem Passthrough	<input type="text" value="Enabled"/> ▼
Fax Passthrough	<input type="text" value="Enabled"/> ▼
CNG Tone Detection	<input type="text" value="Disabled"/> ▼
Fax Tone Detection	<input type="text" value="Enabled"/> ▼
DTMF Signal to Noise	<input type="text" value="0"/> [-3..+6] dB
DTMF Minimum Level	<input type="text" value="-38"/> [-48..-14] dBm0

Figure 13: Fax Media List

Description	<input type="text" value="Tenor FAX Media List"/>		
Media Profiles List	<input type="text" value="VentaFAX G711A"/> <input type="text"/> <input type="text"/>	<input type="button" value="Up"/> <input type="button" value="Down"/> <input type="button" value="Add/Edit"/> <input type="button" value="Remove"/>	*
SDES-SRTP Profile	<input type="text" value="None"/> ▼	Associated SIP SG Listen Ports should be TLS only. +	
DTLS-SRTP Profile	<input type="text" value="None"/> ▼	+	
Media DSCP	<input type="text" value="46"/>	* [0..63]	
RTCP Mode	<input type="text" value="RTCP"/> ▼		
Dead Call Detection	<input type="text" value="Disabled"/> ▼		
Silence Suppression	<input type="text" value="Disabled"/> ▼		

Gain Control		Digit Relay	
Receive Gain	<input type="text" value="0"/> [-14..+6] dB	Digit (DTMF) Relay Type	<input type="text" value="RFC 2833"/> ▼
Transmit Gain	<input type="text" value="0"/> [-14..+6] dB	Digit Relay Payload Type	<input type="text" value="101"/> [96..127]
Passthrough/Tone Detection			
Modem Passthrough	<input type="text" value="Enabled"/> ▼		
Fax Passthrough	<input type="text" value="Enabled"/> ▼		
CNG Tone Detection	<input type="text" value="Disabled"/> ▼		
Fax Tone Detection	<input type="text" value="Enabled"/> ▼		
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 14:** Remote Authorization Table

Realm	<input type="text" value="Realm"/>
Authentication ID	<input type="text" value="virginpbx01_0118337413"/> *
Password Setting	<input type="text" value="Use Current"/> ▼
From URI User Match	<input type="text" value="Regex"/> ▼
Match Regex	<input type="text" value=".*"/>

## 7. Message Manipulation

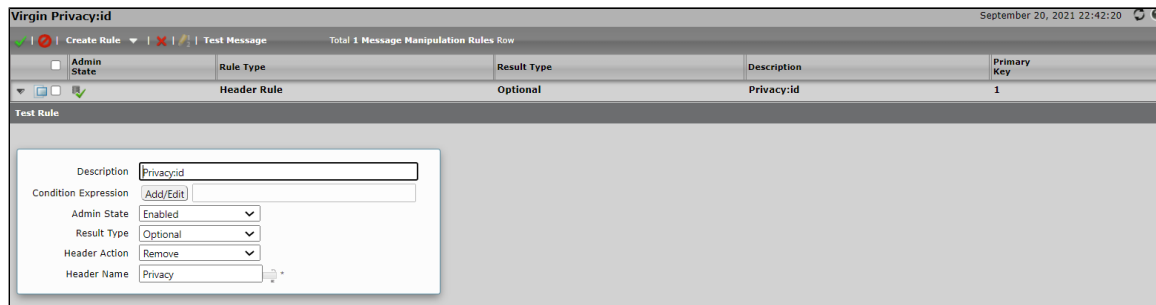
SIP Message Manipulation feature is used by a SIP Signaling Group to manipulate the incoming or outgoing messages. This feature is intended to enhance interoperability with different vendor equipment and applications, and for correcting any fixable protocol errors in SIP messages on a fly without any changes to firmware/software.

The feature's Message Manipulation Engine (MME) manipulates SIP messages to compensate for interoperability issues between different vendor equipment and for dynamically correcting repairable protocol errors in SIP messages. The flows described in this article are message flows and should not be confused with call flows.

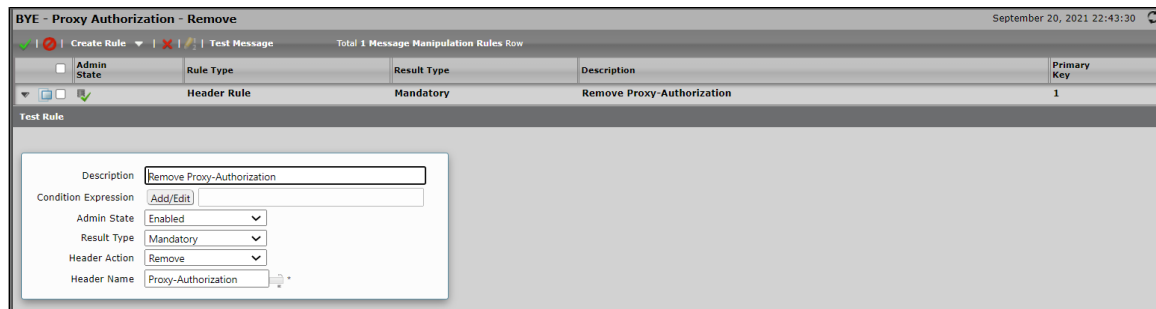
Select **Settings > Message Manipulation > Message Rule Tables** to access the Message Manipulation configuration screen.

1. MS teams by default adds a Privacy :id. You must create this message manipulation to remove it, and the called can see the calling party number.

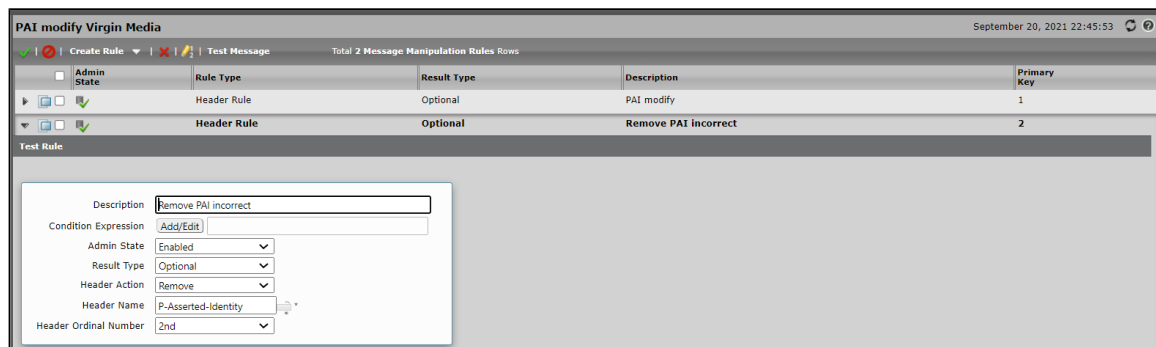
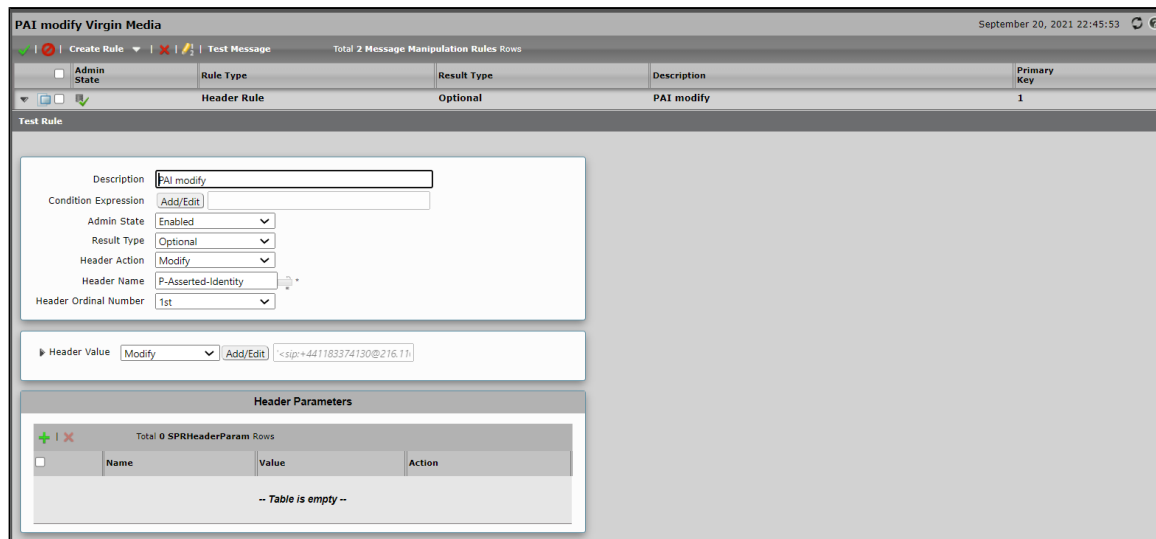
Figure 15: Message Manipulation



2. The SBC 2K sent in the BYE message Proxy-Authentication. Virgin Media asked me to remove it. We created the next Message Manipulation.



3. SBC2K needs to send the next PAI to VM: P-Asserted-Identity: <sip:+441183374130@216.110.2.220:5060;user=phone>. Additional we need to remove the next second PAI: P-Asserted-Identity: <sip:prg1teams@interopdomain.com>



## 8. Signaling Groups



Signaling Groups allow telephony channels for routing and sharing configuration. They are used for

- routing calls and selecting [Call Routes](#)
- selecting Tone Tables and Action Sets
- specifying protocol settings and links to server, media, and mapping tables for SIP

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

**Figure 16:** Virgin Media Signaling Group

Description:

Admin State:

Service Status: Up

---

**SIP Channels and Routing**

Action Set Table:  +

Call Routing Table:  +

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

SIP Profile:  +

SIP Mode:

Agent Type:

Interop Mode:

**Media Information**

Supported Audio/Fax Modes:

Supported Video/Application Modes:

Media List ID:  +

---

SIP Server Table:  +

Load Balancing:

Channel Hunting:

Notify Lync CAC Profile:

Challenge Request:

Outbound Proxy IP/FQDN:

Outbound Proxy Port:  [1..65535]

No Channel Available Override:

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

Call Proceeding Timer:  [24..750] secs

QoE Reporting:

Use Register as Keep Alive:

Forked Call Answered Too Soon:

Play Ringback:

Tone Table:  +

Play Congestion Tone:

Early 183:

Allow Refresh SDP:

Music on Hold:

RTCP Multiplexing:

---

Pass-thru Peer SIP Response Code:

**SIP IP Details**

Teams Local Media Optimization:

Signaling/Media Source IP:

Signaling DSCP:  \* [0..63]

**NAT Traversal**

ICE Support:

**Static NAT - Outbound**

Outbound NAT Traversal:

---

**Static NAT - Inbound**

Detection:

---

**Listen Ports**

Total 2 SIP Listen Port Rows

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

**Federated IP/FQDN**

Total 2 SIP Federated IP Rows

IP/FQDN	Netmask/Prefix
<input type="checkbox"/> 213.106.222.178	255.255.255.255
<input type="checkbox"/> 82.14.171.226	255.255.255.255

Message Manipulation Enabled

**Inbound Message Manipulation**

Message Table List

--

Up  
Down  
Add/Edit  
Remove

**Outbound Message Manipulation**

Message Table List

Virgin Privacyrid
BYE - Proxy Authorization - Remove
PAI modify Virgin Media

Up  
Down  
Add/Edit  
Remove

**Figure 17: MS Teams Signaling Group**

Description (SIP) From/To MS Teams

Admin State Enabled

Service Status Up

**SIP Channels and Routing**

Action Set Table None +

Call Routing Table From MS Teams +

No. of Channels 60 ~ [1..960]

SIP Profile MS Teams SIP Profile +

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 Proxy Direct Add/Edit Remove

Media List ID MS Teams Media List +

SIP Server Table MS Teams +

Load Balancing First

Channel Hunting Most Idle

Notify Lync CAC Profile Disable

Challenge Request Disable

Outbound Proxy IP/FQDN

Outbound Proxy Port 5061 [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 Disable

Play Ringback Auto on 180

Tone Table Default Tone Table +

Play Congestion Tone Disable

Early 183 Enable

Allow Refresh SDP Enable

Music on Hold Disabled

RTCP Multiplexing Enable

Use Register as Keep Alive Disable

Forked Call Answered Too Soon 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**

Teams Local Media Optimization Disable

Signaling/Media Source IP Ethernet 4 IP (216.110.2.227)

Signaling DSCP 40 \* [0..63]

**NAT Traversal**

ICE Support Enabled

ICE Mode Lite

**Static NAT - Outbound**

Outbound NAT Traversal

---

**Static NAT - Inbound**

Detection

**Listen Ports**

Total 1 SIP Listen Port Row

	Port	Protocol	TLS Profile ID
<input type="checkbox"/>	5061	TLS	MS Teams

**Federated IP/QDN**

Total 1 SIP Federated IP Row

	IP/QDN	Netmask/Prefix
<input type="checkbox"/>	sip-all.pstnhub.microsoft.com	255.255.255.255

Message Manipulation

**Figure 18: Fax Signaling Group**

Description

Admin State

Service Status Up

**SIP Channels and Routing**

Action Set Table

Call Routing Table

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

SIP Profile

SIP Mode

Agent Type

Interop Mode

SIP Server Table

**Media Information**

Supported Audio/Fax Modes

Supported Video/Application Modes

Media List ID

Play Ringback

Load Balancing

Channel Hunting

Notify Lync CAC Profile

Challenge Request

Outbound Proxy IP/QDN

Outbound Proxy Port  \* [1..65535]

No Channel Available Override

Call Setup Response Timer  [180..750] secs

Call Proceeding Timer  [24..750] secs

QoE Reporting

Use Register as Keep Alive

Forked Call Answered Too Soon

Tone Table

Play Congestion Tone

Early 183

Allow Refresh SDP

Music on Hold

RTCP Multiplexing

**Mapping Tables**

SIP To Q.850 Override Table

Q.850 To SIP Override Table

Pass-thru Peer SIP Response Code

**SIP IP Details**

Teams Local Media Optimization

Signaling/Media Source IP

Signaling DSCP  \* [0..63]

**NAT Traversal**

ICE Support

**Static NAT - Outbound**

Outbound NAT Traversal

**Static NAT - Inbound**  
 Detection 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/QDN**

Total 1 SIP Federated IP Row		
<input type="checkbox"/>	IP/QDN	Netmask/Prefix
<input type="checkbox"/>	10.35.137.105	255.255.255.255

Message Manipulation Disabled

## 9. 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. In addition, the Action Set Table can configure Transformation tables as a reusable pool.

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

**Figure 19:** Virgin Media Transformation

Description

Admin State Enabled

Match Type Optional (Match One)

**Input Field**

 Type Called Address/Number  
 Value

**Output Field**

 Type Called Address/Number  
 Value

Description

Admin State Enabled

Match Type Optional (Match One)

**Input Field**

 Type Calling Address/Number  
 Value

**Output Field**

 Type Calling Address/Number  
 Value

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="+ (441183374130)"/>	Value <input type="text" value="\1"/>

Figure 20: MS Teams Transformation

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="(.*?)"/>	Value <input type="text" value="\1"/>

Description

Admin State

Match Type

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

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="\+999"/>	Value <input type="text" value="999\1"/>

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="\+112"/>	Value <input type="text" value="112\1"/>

Description

Admin State

Match Type

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/>	Type <input type="text" value="Called Address/Number"/>
Value <input type="text" value="\+18000"/>	Value <input type="text" value="18000\1"/>

Description	<input type="text"/>
Admin State	Enabled <input type="button" value="v"/>
Match Type	Optional (Match One) <input type="button" value="v"/>

Input Field	Output Field
Type <input type="text" value="Calling Address/Number"/> <input type="button" value="v"/>	Type <input type="text" value="Calling Address/Number"/> <input type="button" value="v"/>
Value <input type="text" value="(.*)"/>	Value <input type="text" value="\1"/>

**Figure 21: Fax-Tenor Transformation**

Description	<input type="text" value="International fax"/>
Admin State	Enabled <input type="button" value="v"/>
Match Type	Optional (Match One) <input type="button" value="v"/>

Input Field	Output Field
Type <input type="text" value="Called Address/Number"/> <input type="button" value="v"/>	Type <input type="text" value="Called Address/Number"/> <input type="button" value="v"/>
Value <input type="text" value="(.*)"/>	Value <input type="text" value="+\1"/>

Description	<input type="text"/>
Admin State	Enabled <input type="button" value="v"/>
Match Type	Optional (Match One) <input type="button" value="v"/>

Input Field	Output Field
Type <input type="text" value="Calling Address/Number"/> <input type="button" value="v"/>	Type <input type="text" value="Calling Address/Number"/> <input type="button" value="v"/>
Value <input type="text" value="(.*)"/>	Value <input type="text" value="+\1"/>

## 10. Call Routing Table

Call Routing tables allow you to configure flexible routes for transferring calls between Signaling Groups and for translating the calls. They allow call transfers between ports and protocols, such as ISDN to SIP. These tables serve as one of the central connection points of the system, linking Transformation Tables, Message Translations, Cause Code Reroute Tables, 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 22:** Virgin Media Call Routing

**Route Details**

Description

Admin State

Route Priority

Call Priority

Number/Name Transformation Table  +

Time of Day Restriction  +

**Destination Information**

Destination Type

Message Translation Table  +

Cause Code Reroutes  +

Cancel Others upon Forwarding

Fork Call

Destination Signaling Groups

(SIP) (SIP) From/To MS Teams

(SIP) From/To Tenor FAX

Up

Down

Add/Edit

Remove

Enable Maximum Call Duration

**Media**

Audio/Fax Stream Mode

Video/Application Stream Mode

Media Transcoding

Media List  +

**Quality of Service**

Quality Metrics Number of Calls  [1..100]

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

Min. ASR Threshold  % [0..100]

Enable Min MOS Threshold

Enable Max. R/T Delay

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

Enable Max. Jitter

Max. Jitter  ms [1..3000]

**Figure 23:** MS Teams Routing



Route Details

Description

Admin State

Route Priority

Call Priority

Number/Name Transformation Table  +

Time of Day Restriction  +

---

Destination Information

Destination Type

Message Translation Table  +

Cause Code Reroutes  +

Cancel Others upon Forwarding

Fork Call

Destination Signaling Groups

Enable Maximum Call Duration

---

Media

Audio/Fax Stream Mode

Video/Application Stream Mode

Media Transcoding

Media List  +

Quality of Service

Quality Metrics Number of Calls  [1..100]

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

Min. ASR Threshold  % [0..100]

Enable Min MOS Threshold

Enable Max. R/T Delay

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

Enable Max. Jitter

Max. Jitter  ms [1..3000]

**Figure 24:** 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

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  ▼

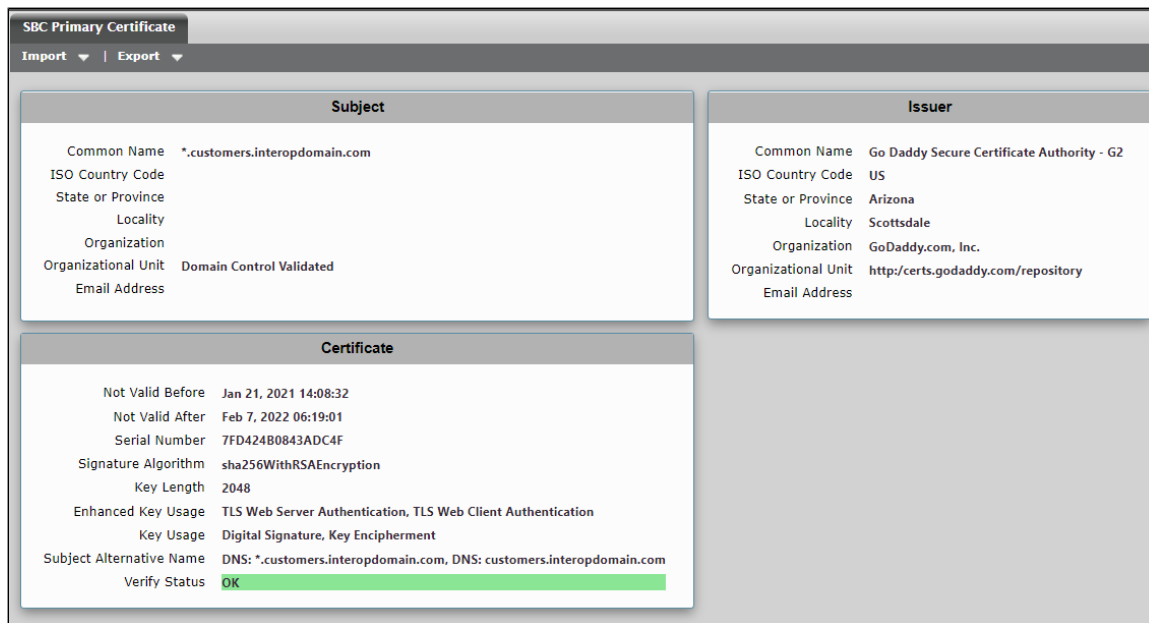
Max. Jitter  ms [1..3000]

## 11. SBC Primary Certificate

You can change the certificate installed on the SBC Edge system by obtaining the signed certificate from a Trusted CA or from a local Stand-Alone Windows Certificate Authority.

Select **Settings > Security > SBC Certificates > SBC Primary Certificate**.

**Figure 25:** SBC Primary Certificate



## 12. Trusted CA Certificates

A trusted certificate authority issues a Trusted CA Certificate. These certificates are imported to the SBC Edge to establish its authenticity on the network.

Select Settings > Security > SBC Certificates > Trusted CA Certificates.

Figure 26: Trusted CA Certificates

## Section-B: MS Teams Configuration

### Configuring MS Teams

The following Microsoft Teams configurations are included in this section:

1. [Configuring Microsoft Teams](#)
2. [Obtain IP address and FQDN](#)
3. [Domain Name](#)
4. [Obtain a Certificate](#)
5. [Public Certificate](#)
6. [Configure and Generate Certificates on the SBC](#)
7. [Configure Office 365 Tenant Voice Routing](#)

### Configuring Microsoft Teams

#### Microsoft Teams Direct Routing Configuration


Consult the Microsoft [documentation](#) for detailed information on Direct Routing interface configuration guidelines, including the RFC standards and the syntax of SIP messages.

#### Obtain IP Address and FQDN

The following table provides the requirements for configuring the SBC to support Teams Direct Routing.

Requirement	How it is used
<b>Public IP address of NAT device (must be Static)*</b>	Required for SBC Behind the NAT deployment.
<b>Private IP address of the SBC</b>	

<b>Public IP address of SBC</b>	Required for SBC with Public IP deployment.
<b>Public FQDN</b>	The Public FQDN must point to the Public IP Address.



 \*NAT translates a public IP address to a Private IP address.

## Domain Name

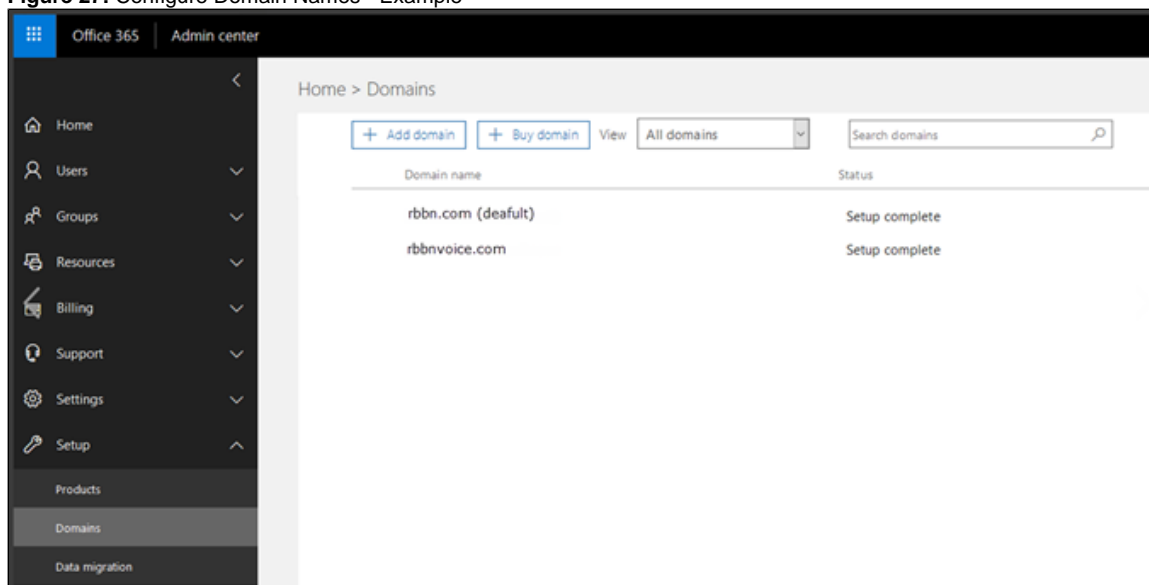
For the SBC to pair with Microsoft Teams, the SBC FQDN domain name must match a name registered in both the **Domains** and **DomainUriMap** fields of the Tenant. Follow the steps to verify that the correct domain name is configured for the Tenant:

1. On the Microsoft Teams Tenant side, execute **Get-CsTenant**.
2. Review the output.
3. Verify that the Domain Name configured is listed in the **Domains** and **DomainUriMap** attributes for the Tenant. If the Domain Name is incorrect or missing, the SBC will not pair with Microsoft Teams.

You can configure users from any SIP domain registered for the tenant. For example, you can configure user [user@example.com](#) with the SBC FQDN name [sbc2.examplevoice.com](#), as long as both names are registered for the tenant.

Domain Name	Use for SBC FQDN	FQDN names - Examples	IPv4 Address
<a href="#">rbbn.com</a>		Valid names: <a href="#">sbc1.rbbn.com</a>	203.0.113.100
<a href="#">rbbnvoice.com</a>		Valid names: <ul style="list-style-type: none"> <li>· <a href="#">sbc2.rbbnvoice.com</a></li> <li>· <a href="#">emea.rbbnvoice.com</a></li> <li>· <a href="#">apac.rbbnvoice.com</a></li> </ul> Non-Valid name; <a href="#">sbc2.emea.rbbnvoice.com</a> (This requires registering domain name <a href="#">emea.rbbnvoice.com</a> in "Domains" first.)	

**Figure 27:** Configure Domain Names - Example



The screenshot shows the Microsoft Teams Admin Center interface. The left sidebar contains navigation options: Home, Users, Groups, Resources, Billing, Support, Settings, Setup, Products, Domains, and Data migration. The main content area is titled 'Home > Domains' and features a '+ Add domain' button, a '+ Buy domain' button, a 'View' dropdown menu set to 'All domains', and a 'Search domains' search box. Below these controls is a table with two columns: 'Domain name' and 'Status'. The table lists two domains: 'rbbn.com (default)' and 'rbbnvoice.com', both with a status of 'Setup complete'.

## Obtain a Certificate

## Public Certificate

Make sure the certificate is issued by one of the supported certification authorities (CAs). Note that wildcard certificates are supported.

- Refer to [Microsoft documentation](#) for the supported CAs.
- Refer to [Domain Name](#) for certificate Common name formats.

## Configure and Generate Certificates on the SBC

Microsoft Teams Direct Routing allows only TLS connections from the SBC for SIP traffic with a certificate signed by one of the trusted certification authorities.

Follow the steps to request a certificate for the SBC External interface and configure it based on the example using the GlobalSign:

- Generate a Certificate Signing Request (CSR) and obtain the certificate from a supported Certification Authority.
- Import the Public CA Root/Intermediate Certificate on the SBC.
- Import the Microsoft CA Certificate on the SBC.
- Import the SBC Certificate.

You can obtain the certificate through the Certificate Signing Request (see the following instructions). You can obtain the Trusted Root and Intermediary Signing Certificates from your certification authority.

## Configure Office 365 Tenant Voice Routing

A Tenant is used within the Microsoft environment as a single independent enterprise that has subscribed to Office 365 services. Through this tenant, administrators can manage projects, users, and roles. Perform the following steps to configure the Tenant. For details on accessing the Tenant, refer to [Microsoft Teams Documentation](#).

1. Create Online PSTN Gateway that points to the SBC:
  - a. Enter the **SBC FQDN** (Example below: [sbc1.rbbn.com](#)). Be sure to configure the FQDN for the Tenant in both the **Domains** and the **DomainUrlMap** fields.
  - b. Enter the **SBC SIP Port** (Example below - SipPort5061).

```
New-CsOnlinePSTNGateway -Fqdn sbc1.rbbn.com -SipSignallingPort SipPort5061 -MaxConcurrentSessions  
<Max Concurrent Session which SBC capable handling> -Enabled $true
```

2. Configure Teams usage for the user:
  - a. Enter the User Identity (Example below: [-user1@domain.com](#))

```
Get-CsOnlineUser -Identity user1@domain.com Set-CsUser -Identity user1@domain.com -EnterpriseVoiceEnabled $true -  
HostedVoiceMail $true -OnPremLineURI tel:+10001001008  
  
Grant-CsOnlineVoiceRoutingPolicy -PolicyName "GeneralVRP" -Identity user1@domain.com  
  
Grant-CsTeamsCallingPolicy -PolicyName AllowCalling -Identity user1@domain.com  
  
Grant-CsTeamsUpgradePolicy -PolicyName UpgradeToTeams -Identity user1@domain.com
```







## Supplementary Services and Features Coverage

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

**Table 1:** Interoperability Compliance Test Results

Test Number	Test Scenario	Setup / Test Results	Coverage
-------------	---------------	----------------------	----------

IOP1	Vendor's eSBC response to SIP OPTIONS messages from SBCthe	<p>No calls are required for this test. SIP trace to be captured for approx 60 seconds and checked for correct signaling.</p> <p>For each Vendors eSBC, the Vendors SBC periodically sends an OPTIONS request to the vendor's eSBC to check if its SIP stack is reachable. If a SIP response 200 OK is received from the IP-PBX, the SIP trunk is placed (or remains) in an In-Service state.</p> <p>Example: OPTIONS sip:ping@&lt;ip-pbx_IP_Addr&gt;:5060 SIP/2.0</p>	✓
IOP2	SBC response to SIP OPTIONS messages from Vendors eSBC	<p>No calls are required for this test. SIP trace to be captured for approx 60 seconds (depending on the agreement) and checked for correct signaling.</p> <p>Vendor's eSBC setup for Solution IP.Addr Mode eSBC is configured to send the OPTIONS messages to the SBC periodically. The SBC responds with SIP response 200OK - Example: "OPTIONS sip:ping@192.168.1.10:5060 SIP/2.0"</p> <p>Check that the eSBC can simultaneously send SIP OPTIONS messages to both the solution SBC addresses.</p>	✓
IOP4	Basic test call from IP-PBX to PSTN line through SBC-A (using SBC-A IPV4 ip address).	<p>The IP-PBX line initiates a call. The call is answered. The IP-PBX line terminates the call.</p> <p>Vendor's eSBC setup for Solution IP.Addr Mode The IP-PBX initiates a call; an Invite is seen from the eSBC to the SBC-A; proxy authentication challenge is returned to the eSBC; a re-invite with correct credentials is received from the eSBC; and the call progresses as expected. Example: Request-Line: INVITE sip:&lt;B-party&gt;@&lt;SBC-A ip.addr TBD&gt;:5060 SIP/2.0 To: sip:&lt;B-Party&gt;@&lt;SBC-A ip.addr TBD&gt;</p> <p>Check the Wireshark trace and confirm that G.711 A law codec with 10 or 20ms packetization is being used. Verify that the INVITE contains the Session-Expires header, and its syntax is correct. Check that the Supported Header supports the 'timer'. Ensure the response in 200 OK is compatible with the INVITE. Verify that the Required Header contains the 'timer'. (See IOP9).</p>	✓
IOP5	Basic test call from IP-PBX to PSTN line through SBC-B (using SBC-B IPV4 ip address)	<p>The IP-PBX line initiates call, Call is answered, the IP-PBX line terminates call.</p> <p>Vendor's eSBC setup for Solution IP.Addr Mode Call from the IP-PBX. Invite seen from eSBC to SBC-B, proxy authentication challenge returned to eSBC, re-invite with correct credentials from eSBC and call progresses as expected. Example: Request-Line: INVITE sip:&lt;B-party&gt;@&lt;SBC-B ip.addr TBD&gt;:5060 SIP/2.0 To: sip:&lt;B-Party&gt;@&lt;SBC-B ip.addr TBD&gt;</p> <p>Check the Wireshark trace and confirm that G.711 A law codec with 10 or 20ms packetization is being used. Verify that the INVITE contains the Session-Expires header, and its syntax is correct. Check that the Supported Header supports the 'timer'. Ensure the response in 200 OK is compatible with the INVITE. Verify that the Required Header contains the 'timer'. (See IOP9).</p>	✓
IOP7b	<p>Called Number format - Vendors eSBC to soft switch number normalisation - Global Dial Plan</p> <p>Test eSBC capability to send the called number in one of the following Global number formats (user part of Request &amp; To URIs).</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyy (where y refers to any number, calling party = national) +yyyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	<p>Configure the Vendor's SBC for Global calling plan.</p> <p>The IP-PBX line initiates call to PSTN line, Call is answered. The IP-PBX line terminates call.</p> <p>Configure the Vendor's eSBC to present the called number in the user part of the Request &amp; To URIs and sent in one of the following formats:</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national) +44yyyyyyyyyy (where y refers to any number, calling party = national) +yyyyyyyyyy (where y refers to any number, calling party = international) yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	✓

IOP8b	<p>Calling Number format - Vendors eSBC to soft switch number normalisation - Global Dial Plan</p> <p>Test eSBC capability to send calling number in one of the following Global number formats (user part of FROM &amp; PAI URIs).</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national)  +44yyyyyyyyy (where y refers to any number, calling party = national)  00yyyyyyyyy (where y refers to any number, calling party = international)  yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	<p>Configure the Vendor's SBC for Global calling plan.</p> <p>The IP-PBX line initiates call to PSTN line, Call is answered.  The IP-PBX terminates call.</p> <p>Configure the Vendor's eSBC to present the calling number in the user part of the From &amp; PAI URIs to be sent in the one of the following formats:</p> <p>0yyyyyyyyy (where y refers to any number, calling party = national)  +44yyyyyyyyy (where y refers to any number, calling party = national)  00yyyyyyyyy (where y refers to any number, calling party = international)  yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	
IOP9b	<p>Called Number format - soft switch to eSBC number normalisation - Global Dial Plan</p> <p>Test eSBC capability of accepting the called number in one of the following Global number formats (user part of Request &amp; To URIs).</p> <p>+44yyyyyyyyy (where y refers to any number, calling party = national)  +yyyyyyyyyy (where y refers to any number, calling party = international)  yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	<p>Configure the Vendor's SBC for Global calling plan.</p> <p>The PSTN line initiates call to the IP-PBX line, Call is answered.  The PSTN line terminates call.</p> <p>Configure the Vendor's eSBC to accept the called number in the user part of the Request &amp; To URIs in one of the following formats:</p> <p>+44yyyyyyyyy (where y refers to any number, calling party = national)  +yyyyyyyyyy (where y refers to any number, calling party = international)  yyyyyyyyyy (where y refers to any number, calling party = unknown)</p> <p>Also check to see that the INVITE contains Session-Expires header and that it is syntactically correct. Check for Supported Header and ensure the 'timer' is supported. Ensure the response in 200 OK is compatible with the INVITE and check for Required Header and if it contains 'timer'.</p>	
IOP10b	<p>Calling Number format - soft switch to eSBC number normalisation - Global Dial Plan</p> <p>Test eSBC capability of accepting the calling number in one of the following Global number formats (user part of FROM &amp; PAI URIs).</p> <p>+44yyyyyyyyy (where y refers to any number, calling party = national)  +yyyyyyyyyy (where y refers to any number, calling party = international)  yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	<p>Configure the Vendor's SBC for Global calling plan.</p> <p>The PSTN line initiates call to the IP-PBX line, Call is answered.  The PSTN line terminates call.</p> <p>Configure the Vendor's eSBC to accept the calling number in the user part of the Request &amp; To URIs in one of the following formats:</p> <p>+44yyyyyyyyy (where y refers to any number, calling party = national)  +yyyyyyyyyy (where y refers to any number, calling party = international)  yyyyyyyyyy (where y refers to any number, calling party = unknown)</p>	
IOP11	<p>Emergency Call Handling -IP-PBX Line to PSTN - UK Emergency call 999</p>	<p>The IP-PBX initiates a call to the Emergency services using 999. The call is answered.  Either party terminates the call.</p> <p>Example:  Request-Line: INVITE sip:999@&lt;SBC-A ip.addr TBD&gt;:5060 SIP/2.0  To: &lt;sip:999@&lt;SBC-A ip.addr TBD&gt;&gt;  From: &lt;sip:&lt;A-party&gt;@&lt;IP-PBX IP.Addr&gt;</p>	
IOP12	<p>Emergency Call Handling -IP-PBX Line to PSTN - UK Emergency call 112</p>	<p>Call made from the IP-PBX line to the Emergency services using 112. Call answered,  Either party terminates call.</p> <p>Example:  Request-Line: INVITE sip:112@&lt;SBC-A ip.addr TBD&gt;:5060 SIP/2.0  To: &lt;sip:112@&lt;SBC-A ip.addr TBD&gt;&gt;  From: &lt;sip:&lt;A-party&gt;@&lt;IP-PBX IP.Addr&gt;</p>	
IOP13	<p>Emergency Call Handling -IP-PBX Line to PSTN - UK Emergency call 18000 - Text Direct</p>	<p>Call made from the IP-PBX line using a text direct set to the Emergency services using 18000. Call answered.  Either party terminates call.</p> <p>Example:  Request-Line: INVITE sip:18000@&lt;SBC-A ip.addr TBD&gt;:5060 SIP/2.0  To: &lt;sip:18000@&lt;SBC-A ip.addr TBD&gt;&gt;  From: &lt;sip:&lt;A-party&gt;@&lt;IP-PBX IP.Addr&gt;</p>	

IOP14	IP-PBX Line to PSTN - call answer - Originator disconnect	Call made from the IP-PBX line to PSTN line, Answer Call. The IP-PBX line terminates call.	✓
IOP15	IP-PBX Line to PSTN - call answer - Terminator disconnect	Call made from the IP-PBX line to PSTN line, Answer Call. The PSTN line terminates call	✓
IOP16	IP-PBX Line to PSTN - Busy subscriber	Call made from the IP-PBX line to a busy PSTN line (without divert on busy) Wait for the soft switch to return the busy response. Ensure that the Vendor's eSBC does not recurse. Set up the call via secondary SIP trunk.	✓
IOP17	IP-PBX Line to PSTN - No answer timeout test	Call made from the IP-PBX line to a PSTN line (without divert on no answer) Do not answer call. Wait for soft switch to return no answer timeout response. Ensure that Vendor's eSBC does not recurse and setup call via secondary SIP trunk.	✓ with Caveat
IOP18	IP-PBX Line to PSTN - Subscriber not reachable	Call made from the IP-PBX line to an invalid number. Wait for soft switch to return response. Ensure that Vendor's eSBC does not recurse and setup call via secondary SIP trunk.	✓
IOP19	PSTN Line to IP-PBX - call answer - Originator disconnect.	Call made from a PSTN line to an IP-PBX line, Answer Call. The Originator disconnects the call.	✓
IOP20	PSTN Line to IP-PBX - call answer - Terminator disconnect	Call made from a PSTN line to an IP-PBX line, Answer Call. The IP-PBX line terminates call.	✓
IOP23	PSTN Line to IP-PBX - subscriber not reachable	Call made from a PSTN line to an invalid number/unprogrammed DDI on the IP-PBX. Wait for the IP-PBX to return response.	✓
IOP24	Verify CLIP service on IP-PBX line (incoming call from PSTN)	Call made from a PSTN line to IP-PBX line. The PSTN line is set to allow the CLI presentation. Check that the CLI is delivered as expected. Either party terminates call.	✓
IOP25	Verify CLIR service on IP-PBX line (incoming call from PSTN)	Call made from a PSTN line to IP-PBX line. The PSTN line is set to restrict the CLI presentation. Check that the CLI is not delivered as expected. Either party terminates call.	✓
IOP26	Verify CLIP service on PSTN line (outgoing call from IP-PBX, From)	Ensure the number used in the From header is agreed with the Virgin Media and entered into the soft switch database for screening.  Call made from an IP-PBX line to a PSTN line. Ensure that the Vendor's eSBC is configured such that the IP-PBX line sends the From header containing the Calling Line ID (CLI) in the INVITE.  Ensure that the Vendor's eSBC allows presentation of its the CLI using the privacy-header (Privacy: none or privacy-header not present).  Ensure that the expected CLI is presented to the PSTN line. Either party terminates call.	✓
IOP27	Verify CLIP service on PSTN line (outgoing call from IP-PBX, PAI/PPI)	Ensure the number used in PAI/PPI header is agreed with the Virgin Media and entered into the soft switch database for screening.  Call made from an IP-PBX line to a PSTN line. Ensure that the Vendor's eSBC is configured such that the IP-PBX line sends PAI/PPI header containing Calling Line ID (CLI) in the INVITE. If the PAI header is populated, use it in preference to the From header. Ensure that the Vendor's eSBC allows the presentation of its the CLI using the privacy-header (Privacy: none or privacy-header not present).  Ensure that the expected CLI is presented to the PSTN line. Either party terminates call.	✓



IOP28	Verify CLIR service on PSTN line (outgoing call from IP-PBX)	<p>Ensure the number used in From/PAI header is agreed with the Virgin Media and entered into the soft switch database for screening.</p> <p>Call made from an IP-PBX line to a PSTN line. Ensure that the Vendor's eSBC is configured such that the IP-PBX line sends the From and/or PAI header, containing either the Calling Line ID or obscured information in the INVITE.</p> <p>Example: From: "user751000" &lt;sip:+441256751000@192.168.1.10&gt;;tag=12345 From: "Anonymous" &lt;sip:anonymous@anonymous.invalid&gt;;tag=12345</p> <p>Ensure that the Vendor's eSBC restricts the presentation of its the CLI using the privacy-header (Privacy: id or Privacy: user or Privacy: user;id).</p> <p>Ensure that the CLI is NOT presented to the PSTN line. Either party terminates call.</p>	✓
IOP29	Verify Call Forward Immediate (unconditional) on a IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	<p>Initiate a call from a PSTN line to an IP-PBX line, forwarding the call to a line within the same IP-PBX, Answer Call. Either party terminates call.</p> <p>Does the IP-PBX has configuration settings to send the SIP status 181 messages to the soft switch?</p>	✓
IOP30	Verify Call Forward Immediate (unconditional) on a IP-PBX line (Incoming call from PSTN, call forward terminates PSTN)	<p>Initiate a call from a PSTN line to an IP-PBX line, forwarding the call to a line in the PSTN, Answer Call. Either party terminates call.</p>	✓
IOP31	Verify Call Forward Busy on IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	<p>Initiate a call from a PSTN line to an IP-PBX line, with Call Forward Busy (or equivalent) to a line within the IP-PBX, Answer Call. Either party terminates call.</p>	✓
IOP32	Verify Call Forward No-answer on IP-PBX line (Incoming call from PSTN, call forward terminates within IP-PBX)	<p>Initiate a call from a PSTN line to an IP-PBX line with Call Forward No-answer (or equivalent) to a line within the IP-PBX, Answer Call. Either party terminates call.</p>	✓
IOP33	Verify Call Hold Service on IP-PBX (Incoming call from PSTN)	<p>Initiate a call from a PSTN line to an IP-PBX line with Call on Hold, Answer call. The IP-PBX line puts the call on hold. Leave the call on hold for 30 seconds and then retrieve the call. Ensure the speech path is re-established in both directions. Either party terminates call.</p>	✓
IOP34	Verify 3-party conference service on IP-PBX (Incoming call from PSTN, 3rd party within IP-PBX)	<p>Initiate a call from a PSTN line to an IP-PBX line with a 3-party conference, Answer call. The IP-PBX line uses the 3-party conference facility to put the PSTN line on hold while dialling 3rd party (another IP-PBX line). Once the 3rd party has answered the call, place the 3 parties in a conference. Ensure that all parties have a two-way speech path. Keep the speech path open for at least 20 seconds. Either party terminates call.</p>	✓
IOP35	Verify 3-party conference service on IP-PBX (Incoming call from PSTN, 3rd party PSTN)	<p>Initiate a call from a PSTN line to an IP-PBX line with a 3-party conference, Answer call. The IP-PBX line uses the 3-party conference facility to put the PSTN line on hold whilst dialling 3rd party (another PSTN line). Once the 3rd party has answered the call, place the 3 parties in a conference. Ensure that all parties have a two-way speech path. Keep the speech path open for at least 20 seconds. Either party terminates call.</p>	✓
IOP36	Verify do-not-disturb service on IP-PBX line (Incoming call from PSTN)	<p>Initiate a call from a PSTN line to an IP-PBX line with the do-not-disturb feature active. Ensure the IP-PBX line does not ring. The PSTN line receives an appropriate announcement or tone.</p> <p>Record the SIP status received from the IP-PBX.</p>	✓
IOP37	Verify Call park service on IP-PBX line (Incoming call from PSTN)	<p>Initiate a call from a PSTN line to IP-PBX line A with the Call Park (or equivalent) feature active. Answer the call. Place the call in the Park condition. After 10 seconds, retrieve the call from the IP-PBX line B, using the Call Park pick-up code. Ensure the speech path is re-established in both directions. Either party terminates the call.</p>	✓

IOP38	Verify Call Waiting on an IP-PBX line, involving a PSTN line	<p>Initiate a call from a PSTN line A to an IP-PBX line with Call Waiting active, Answer call.</p> <p>Initiate a call from a PSTN line B to the same IP-PBX line, which receives an indication that a second call is waiting.</p> <p>The PSTN line B receives the ringback tone.</p> <p>The IP-PBX line answers the call from PSTN line B.</p> <p>The PSTN line A should receive an appropriate indication that they are now on hold.</p> <p>The IP-PBX line toggles the call back to the PSTN line A.</p> <p>Ensure the speech path is re-established in both directions, and that PSTN line B received an indication that they are now on hold.</p> <p>Either party terminates call.</p>	✓
IOP39	Verify DTMF transmission from/to IP-PBX - Inband	<p>Configure the IP-PBX/eSBC to send DTMF transmission in-band.</p> <p>Call made from the IP-PBX line to a PSTN line, Answer call.</p> <p>The PSTN line presses each of the keys on the number pad in turn. Note the far-end experience.</p> <p>The IP-PBX line presses each of the keys on the number pad in turn. Note the far-end experience.</p> <p>Did the received DTMF tone reflect the length of time the key was pressed?</p>	✓
IOP40	Verify DTMF transmission from/to IP-PBX - RFC 2833 - telephone-event	<p>Configure the IP-PBX/eSBC to send DTMF transmission using RFC 2833 - telephone-event.</p> <p>Call made from IP-PBX line to a PSTN line, Answer call.</p> <p>The PSTN line presses each of the keys on the number pad in turn. Note the far-end experience.</p> <p>The IP-PBX line presses each of the keys on the number pad in turn. Note the far-end experience.</p> <p>Did the received DTMF tone reflective the length of time the key was pressed?</p>	✓
IOP41	T.38 Fax transmission mode - PSTN to IP-PBX origination	<p>Configure the ATA/IP-PBX/eSBC such that the Fax transmission is sent using the T.38 Version 0 Fax transmission mode.</p> <p>Call made from PSTN line to an IP-PBX line, Answer call.</p> <p>Fax transmission is completed, and the call is terminated by either of the end terminal devices.</p> <p>Ensure the Wireshark trace shows that the T.38 Fax Transmission is used.</p> <p>Check that the fax is transmitted and received as expected.</p>	✓
IOP42	T.38 Fax transmission mode - IP-PBX to PSTN origination	<p>Configure the ATA/IP-PBX/eSBC such that the Fax transmission is sent using the T.38 Version 0 Fax transmission mode.</p> <p>Call made from the IP-PBX line to a PSTN line, Answer call.</p> <p>Fax transmission is completed, and call is terminated by either of the end terminal devices.</p> <p>Ensure the Wireshark trace shows that the T.38 Fax Transmission is used.</p> <p>Check that the fax is transmitted and received as expected.</p>	✓
IOP43	In-band G.711 Fax transmission mode - PSTN to IP-PBX origination	<p>Configure the ATA/IP-PBX/eSBC such that Fax transmission is sent using in-band G.711 Fax transmission mode.</p> <p>Call made from the PSTN line to an IP-PBX line, Answer call.</p> <p>Fax transmission is completed and call is terminated by either of the end terminal devices</p> <p>Ensure Wireshark trace shows that in-band G.711 Fax Transmission is used.</p> <p>Check that the fax is transmitted and received as expected.</p>	✓
IOP44	In-band G.711 Fax transmission mode - IP-PBX to PSTN origination	<p>Configure the ATA/IP-PBX/eSBC such that Fax transmission is sent using in-band G.711 Fax transmission mode.</p> <p>Call made from the IP-PBX line to a PSTN line, Answer call.</p> <p>Fax transmission is completed and call is terminated by either of the end terminal devices</p> <p>Ensure Wireshark trace shows that in-band G.711 Fax Transmission is used.</p> <p>Check that the fax is transmitted and received as expected.</p>	✓

IOP45	Test of Call in progress audit function - response to in-call OPTIONS from soft switch to eSBC.	<p>Initiate a call from an IP-PBX line to a PSTN line. Answer the call. Leave the two parties in conversation for 10 minutes. Ensure both parties have two-way speech. Either party terminates the call.</p> <p>Check wireshark trace to ensure that in-call OPTIONS are sent by the soft switch and that the eSBC responds with status 200OK. Check to see if the eSBC sends any in-call audit SIP messages.</p>	✓
IOP46	Test of 4 simultaneous calls, 2 inbound, 2 outbound calls	<p>Configure the Vendor's eSBC such that successive calls route to alternate Vendor's SBCs (round robin, cyclic, and so on). Make 4 simultaneous calls: 2 inbound and 2 outbound calls. Answer the calls and ensure two-way speech path is established for each call.</p>	✓
IOP47	Test of eSBC endpoint restart-recovery	<p>Restart the Vendor's eSBC and ensure that, after recovery, inbound and outbound calls are successful.</p>	✓
IOP48	Test of eSBC loss of Ethernet link and reconnection	<p>Remove the Ethernet link between the Vendor's eSBC and Vendor's CE router. Leave in this condition for at least 3 minutes. Reconnect the Ethernet link and ensure that after approximately 2 minutes inbound and outbound calls are successful.</p>	✓
IOP49	Test of Primary SBC loss	<p><b>** Contact MSL engineer to carry out the following **</b> On the Primary Vendors SBC, carry out the ALLSTOP command to disable the Vendor's SBC.</p> <p>Call made from the IP-PBX line to a PSTN Line. Call should attempt to route to the Primary SBC. On a non-response to an INVITE, the eSBC re-routes the call to the Secondary SBC. Wait for call answer. Either party terminates call.</p> <p><b>** Contact MSL engineer to carry out the following **</b> Restart the Primary SBC.</p>	✓
IOP51	Test of verify call forward Internal Busy	<p>Additional test to cover when vendors are using Microsoft Skype for Business 2015.</p> <p>The PBX Subscriber 1 makes a call to another PBX Subscriber 2 so that PSTN to call PBX subscriber 1 is Busy.</p> <p>The PSTN calls PBX user 1. The call should automatically go to voicemail after 10 seconds when forwarding is off.</p> <p>VM is on another PBX Internal Line call should go to Voice Mail.</p> <p>If voicemail PSTN to listen VM announcement if another PBX user check speech is clear in both directions.</p> <p>If forwarded to voicemail PSTN terminated call after hearing VM announcement.</p> <p>If forwarded to another user another either party terminate the call after checking speech is clear in both directions.</p>	✓
IOP52	Test of Call forward internal on No Answer	<p>Additional test to cover when vendors is using Microsoft Skype for Business 2015</p> <p>PSTN call PBX user 1. PBX User 1 not to answer the call</p> <p>The call should automatically go to voicemail (VM) which is in another internal PBX line if call forwarding is turned off.</p> <p>Call automatically goes to voicemail after 10 seconds</p> <p>PSTN terminated call after hearing VM announcement.</p> <p>If forwarded is ON call is forwarded to another PBX user internal</p> <p>Check speech quality, terminate the call after checking speech is clear in both directions</p>	Not executed

IOP53	Test Call from PBX to PSTN	<ol style="list-style-type: none"> <li>1. Vendors eSBC to be configured to offer T.38 in addition to G711A-law and G711-U law</li> <li>2. Call made from the PBX to the PSTN</li> <li>3. Call to be established and two dialog for 10 minutes.</li> <li>4. Check Wireshark output. You should not see T.38 being reflected in the protocol column after call having been established for 7 minutes.</li> <li>5. If T.38 is reflected in the protocol column make a note of this.</li> </ol>	✓
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#### Legend

Supported	✓
Not Supported	✗

## Caveats

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

- IOP 17 - IP-PBX Line to PSTN - No answer timeout test. Ribbon SBC sent the call to second Virgin Media SBC, after the first Virgin Media SBC didn't answer, after that, the SBC sent "user not available" to MS Teams.

## Support

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

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

## References

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

<https://ribboncommunications.com/products>

## Conclusion

This Interoperability document describes a successful configuration and interop involving Ribbon SBC 2000 and MS Teams.

 The SweLite platform is also supported using version 9.0.4 with MS teams.

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

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