Ribbon SBC Edge 8.0.0 IOT Skype for Business 2015 Colt Application Notes

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Document Overview

This document provides a configuration guide for Ribbon SBC Edge Series (Session Border Controller) when connecting to Skype for Business 2015 and Colt SIP Trunking.

This configuration guide supports features described on the Microsoft Technet https://technet.microsoft.com/ web site.

- For additional information on Skype for Business 2015, please visit http://microsoft.com
- For additional information on Ribbon SBC Edge Series (Session Border Controller), please visit https://ribboncommunications.com/

Introduction

The interoperability compliance testing focuses on verifying inbound and outbound calls flows between Ribbon SBC Edge and Skype for Business 2015.

Audience

This is a technical document intended for telecommunications engineers with the purpose of configuring both the Ribbon SBC and the third-party product. There will be steps that require navigating third-party as well as the Ribbon SBC Command Line Interface (CLI). Understanding the basic concepts of TCP/UDP, IP/Routing, and SIP/RTP are also necessary to complete the configuration and for troubleshooting, if necessary.

This configuration guide is offered as a convenience to Ribbon customers. The specifications and information regarding the product in this guide are subject to change without notice. All statements, information, and recommendations in this guide 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 in this guide.

Requirements

The following equipment and software were used for the sample configuration provided:

Table 1: Requirements

	Equipment	Software Version
Ribbon	SBC 2000	V8.0.0build502
	Tenor AFM200	P108-09-26
Third-party Equipment	Microsoft Skype for Business 2015 Mediation Server	6.0.9319.0
	Polycom CX600 SIP Phone	4.0.7577.44455
	VentaFax	7.6.243.597 I

Reference Configuration

The following reference configuration shows connectivity between third-party and Ribbon SBC Edge.



Support

For any questions regarding this document or the content herein, please contact your maintenance and support provider.

Third-Party Product Features

The testing was executed with the Colt test plan. The following features were tested:

- Basic Calls
- Enhanced Calls
- Codec Support
- DTMF Support
- CLI Services
- Encryption

Verify License

SIP Calls

Skype for Business 2015 Configuration

The following new configurations are included in this section:

- 1. PSTN Gateway
- 2. Voice Policy
- 3. PSTN Usage
- 4. Route
- 5. Trunk Configuration

1. PSTN Gateway

Topology Builder > Shared Components > PSTN Gateways

-

Figure 2: Define a new IP/PSTN Gateway

s Skype for Business Server 2015, Topology Builder 📃 🗖 🗙									
File Action Help									
 Skype for Business Server S4B Lab Lync Server 2010 Lync Server 2013 Skype for Business Server 2015 Skype for Business Server 2015 Shared Components SQL Server stores File stores File stores PS New IP/PSTN Gateway Off Urd Help SIP Video trunks Branch sites 	The properties for this item are not available for editing.								

Figure 3: Define FQDN

Skype for Business Server 2015, Topology Builder	
File Action Help	
 Skype for Business Server S4B Lab Lync Server 2010 Skype for Business Server 2013 Skype for Business Server 2015 Shared Components 	g.
Define New IP/PSTN Gateway	x
Define the PSTN Gateway FQDN	
Define the fully qualified domain name (FQDN) for the PSTN gateway. FQDN: *	
10.35.177.244	8
Help Back Next Cancel	

Figure 4: Define IP Address

Skype for Business Server 2015, Topology Builder 🛛 🗖 🗙								
File Action Help								
 Skype for Business Server S4B Lab Lync Server 2010 Lync Server 2013 Skype for Business Server 2015 Shared Components 	The properties for this item are not available for editing.							
📳 Define N	ew IP/PSTN Gateway							
Define the IP address								
Enable IPv4								
 Use all configured IP addresses. 								
 Limit service usage to selected IP addr 	esses.							
PSTN IP address:								
○ Enable IPv6								
 Use all configured IP addresses. 								
 Limit service usage to selected IP addr 	esses.							
PSTN IP address:								
Help	Back Next Cancel							

Figure 5: Define Root Trunk

9	Skype for Business Server 2015, Topology Builder 📃 🗖 🗙								
File	e Action Help								
⊿	Image: Skype for Business Server The Image: S4B Lab Image: Server 2010 Image: Image: Dync Server 2013 Image: Skype for Business Server 2015	properties for this item are not available for editing.							
	A Shared Components Define New IP	/PSTN Gateway							
	Define the root trunk								
	Trunk name: *								
	10.35.177.244								
	Listening port for IP/PSTN gateway: *								
	5060								
	SIP Transport Protocol:								
	Associated Mediation Server:								
	fe.skype2015.sonusnet.com S4B Lab								
	Associated Mediation Server port: *								
	5068								
	Help	Back Finish Cancel							

2. Voice Policy

Control Panel > Voice Routing > Voice Policy

Figure 6: Edit Voice Policy

DIAL PLAN	VOICE POLICY	ROUTE	PSTN USAGE	TRUNK	CONFIGURATION	TEST VOICE ROUTING
Create vo	pice routing test	case inforn	nation			
Edit Voice F	olicy - RewaSBC	Policy				
🗸 ок	X Cancel					
Scope: Us Name: *	er				-	
RewaSBC F	Policy]	
Description	1:				_	
Policy for F	Rewa SBC2000]	
^ Calling	Features					
🖌 Ena	able call forwardin	g				✓ Enable team call
🖌 Ena	able delegation					✓ Enable PSTN reroute
🖌 Ena	able call transfer					Enable bandwidth policy override
📃 Ena	able call park					Enable malicious call tracing
🖌 Ena	able simultaneous	ringing of p	hones			
Associated	PSTN Usages					
🖶 New	Select 💋	🥕 Show deta	ils Remove	1	ŧ	
PSTN usa	ge record	Asso	ociated routes			
RewaSB	5	Rev	waSBC			
Call forwar	ding and simultan	eous ringing	g PSTN usages:			
Route usir	ig the call PSTN usa	iges		•	?	
Translated	number to test:					
					Go	

Associated PST	N Usages			
Select	Remove	1	÷	
PSTN usage re	ecord		Associated voice policies	
RewaSBC			RewaSBC Policy	
Translated num	ber to test:			Go

3. PSTN Usage

Control Panel > Voice Routing > PSTN Usage

DIAL PLAN	VOICE POLICY	ROUTE	PSTN USAGE	TRUNK CONFIGURATION	TEST VOICE ROUTING
Create vo	pice routing test of	ase inforr	nation		
View PSTN	Usage Record - F	RewaSBC			
X Close					
Name:					
RewaSBC					
Associated	Routes				
Route		Patt	ern to match		
RewaSBO	:	,*			
Associated	Voice Policies				
Voice pol	icy	Des	cription		
3. Rewa	SBC Policy	Pol	icv for Rewa SBC2	000	

4. Route

Control Panel > Voice Routing > Route

DIAL PLAN	VOICE POL	ICY ROUT	e ps	TN USAGE	TRUNK	CONFIGURATION	TEST VOICE ROUTING
Create v	oice routing	test case inf	ormatio	n			
Edit Voice	Route - Rewa	SBC					
🧹 ок	🗙 Cancel						
Scope:							
Name: *							
RewaSBC							
Descriptio	n:						
Route to F	Rewa SBC 2000						
Build	a Pattern to	Match					
Add the	starting digits	s that you wa	nt this re	oute to hand	le, or creat	e	
the expr	ression manual	lly by clicking	Edit.				
Starting	digits for nun	bers that yo	u want t	o allow:			
Type a v	alid number an	d then click A	dd.			Add	
						Exceptions	
						Permove	
						Kellove	
Match	this nattern: *						
*	uns pattern.						
•							
Ec	dit Res	et 🥐					
Suppre	ess caller ID						
Altern	ate caller ID:						
Associated	d trunks:						
Pstn	Gateway:10.35.	179.136				Add	
						Remove	

Associated PST	N Usages			
🛀 Select	Remove	1	₽	
PSTN usage re	cord		Associated voice policies	
RewaSBC			RewaSBC Policy	
Translated num	her to test			
				G

5. Trunk Configuration

Control Panel > Voice Routing > Trunk Configuration

Figure 9: Edit Trunk Configuration

DIAL PLAN	VOICE POLICY	ROUTE	PSTN USAGE	TRUNK CONFIGURATION	TEST VOICE ROUTING				
Create vo	Create voice routing test case information								
Edit Trunk (Configuration - P	stnGatewa	y:10.35.179.136						
🗸 ок	X Cancel								
Scope: Po Name: *	ol								
PstnGatew	ay:10.35.179.136								
Description	n:								
Rewa SBC	configuration								
Maximum	early dialogs suppo	orted:							
20									
Encryption	support level:								
Required				•					
Refer supp	ort:								
None				•					
Enable	media bypass								
🖌 Central	lized media process	sing							
Enable	RTP latching								
Enable	forward call histor	у							
Enable	forward P-Asserte	d-Identity (data						
Enable	outbound routing	failover tin	ner						
~ Associa	ated PSTN Usages	•							
<u></u>	Select Remove	1	₽ ₽						
PSTN	l usage record		Associated routes						

Ribbon SBC 1000/2000 Configuration

The following steps provide an example of how to configure Ribbon SBC 1000/2000:

- 1. SIP Profile
- 2. SIP Server

- 3. Media Profile
- 4. Media List
- 5. Transformation Table
- 6. Call Routing Table
- 7. Message Rule Tables
- 8. Signaling Groups

1. SIP Profile

Select Settings > SIP > SIP Profiles

SIP Profiles control how the Ribbon SBC 1000/2000 communicates with SIP devices. These control important characteristics such as session timers, SIP Header customization, SIP timers, MIME payloads, and option tags. The following figures shows the default SIP profile used for the SBC 1000 /2000 for this testing effort:

Figure 10: SIP Profiles

Description Colt SIP Profile	
Session Timer	MIME Payloads
Session Timer Enable Minimum Acceptable Timer 600 Offered Session Timer 3600 Terminate On Refresh Failure False	ELIN Identifier LOC PIDF-LO Passthrough Enable Unknown Subtype Passthrough Disable
Header Customization	Options Tags
FQDN in From HeaderDisableFQDN in Contact HeaderDisableSend Assert HeaderAlwaysSBC Edge Diagnostics HeaderDisableTrusted InterfaceEnableUA HeaderRibbon SBC EdgeCalling Info SourceRFC StandardDiversion Header SelectionLastRecord Route HeaderRFC 3261 Standard	100rel Supported Path Not Present Timer Supported Update Supported
Timers	SDP Customization
Transport Timeout Timer 5000 Maximum Retransmissions RFC Standard Redundancy Retry Timer 180000 —————————————————————————————————	Send Number of Audio Channels Connection Info in Media Section Origin Field Username SBC
Timer T1 500	Session Name VoipCall
Timer T2 4000	Digit Transmission Preference RFC 2833/Voice
Timer T4 5000 Timer D 32000 Timer B 32000 ms Timer F 32000 ms Timer H 32000 ms (64*TimerT1) Timer J 4000	SDP Handling Preference Legacy Audio/Fax

Description Default SIP Profile	
Session Timer	MIME Payloads
Session Timer Disable	ELIN Identifier LOC PIDF-LO Passthrough Enable Unknown Subtype Passthrough Disable
Header Customization	Options Tags
FQDN in From HeaderDisableFQDN in Contact HeaderDisableSend Assert HeaderTrusted OnlySBC Edge Diagnostics HeaderEnableTrusted InterfaceDisableCalling Info SourceRFC StandardDiversion Header SelectionLastRecord Route HeaderRFC 3261 Standard	100rel Supported Path Not Present Update Supported
Timers	SDP Customization
Transport Timeout Timer 5000 Maximum Retransmissions RFC Standard Redundancy Retry Timer 180000 —————————————————————————————————	Send Number of Audio ChannelsTrueConnection Info in Media SectionTrueOrigin Field UsernameSBCSession NameVoipCallDigit Transmission PreferenceRFC 2833/VoiceSDP Handling PreferenceLegacy Audio/Fax

2. SIP Server

Select Settings > SIP > SIP Server Tables

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

Figure 11: Skype

Server Host	Transport
Server Lookup IP/FQDN Priority 1 Host FQDN/IP fe.skype2015.sonusnet.com Host IP Version IPv4 Port 5068 Protocol TCP	MonitorSIP OptionsKeep Alive Frequency30Recover Frequency5Local UsernameAnonymousPeer UsernameAnonymous
Remote Authorization and Contacts	Connection Reuse
Remote Authorization Table None Contact Registrant Table None Session URI Validation Liberal	Reuse True Sockets 4 Reuse Timeout Forever

Figure 12: Fax

Server Host	Transport
Server Lookup IP/FQDN Priority 1 Host FQDN/IP 10.35.137.43 Port 5084 Protocol UDP	MonitorSIP OptionsKeep Alive Frequency30Recover Frequency5Local UsernameAnonymousPeer UsernameAnonymous
Remote Authorization and Contacts	
Remote Authorization Table None Contact Registrant Table None Session URI Validation Liberal	

Figure 13: Colt



3. Media Profile

Select Settings > Media > 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. The following figures are the media profiles of the voice codecs used for the SBC 1000/2000 in this testing effort and are shown for reference only:

Description	Default G711A
Codec	G.711 A-Law
Payload Size	20

Figure 15: Voice Codec G711 U-Law

Voice Code	ec Configuration
Description	Default G711u
Codec	G.711 μ-Law
Payload Size	20

Figure 16: Voice Codec G729

Voice Cod	ec Configuration
Description	G.729
Codec	G.729
Payload Size	20

Figure 17: T.38

Fax Codec Configu	uration
Description	T.38
Maximum Rate	1.38 Fax 14400
Payload Packet Redundancy	3 0
Training Confirmation Procedure	Enabled Locally Generate
Super G3 to G3 Fallback	Disabled Disabled

4. Media List

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

Figure 18: Media List

Description Media Profiles List	Skype Media List Default G711A
SDES-SRTP Profile	None
DTLS-SRTP Profile	None
Media DSCP	46
RTCP Mode	RTCP
Dead Call Detection	Disabled
Silence Suppression	Enabled
Gain Control	Digit Relay
Gain Control	Digit Relay
Gain Control Receive Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Inssthrough/Tone Detection
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Ussthrough/Tone Detection Enabled Enabled Disabled
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection Fax Tone Detection	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 assthrough/Tone Detection Enabled Enabled Disabled Enabled
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection Fax Tone Detection DTMF Signal to Noise	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Issthrough/Tone Detection Enabled Enabled Disabled Enabled 0

Description	Fax Media List
Media Profiles List	Default G711A Default G711u G.729 T.38 *
SDES-SRTP Profile	None
DTLS-SRTP Profile	None
Media DSCP	46
RTCP Mode	RTCP
Dead Call Detection	Disabled
Silence Suppression	Disabled
Gain Control	Digit Relay
Gain Control Receive Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 assthrough/Tone Detection Enabled Enabled
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 assthrough/Tone Detection Enabled Enabled Enabled
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection Fax Tone Detection	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 assthrough/Tone Detection Enabled Enabled Enabled Enabled
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection Fax Tone Detection DTMF Signal to Noise	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 assthrough/Tone Detection Enabled Enabled Enabled Enabled 0

Description	Colt Media List
Media Profiles List	Default G711A Default G711u T.38 *
SDES-SRTP Profile	None
DTLS-SRTP Profile	None
Media DSCP	46
RTCP Mode	RTCP
Dead Call Detection	Disabled
Silence Suppression	Disabled
Gain Control	Digit Relay
Gain Control Receive Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Insthrough/Tone Detection
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Insthrough/Tone Detection
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection Fax Tone Detection	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Assthrough/Tone Detection Enabled Enabled Enabled Enabled
Gain Control Receive Gain 0 Transmit Gain 0 Pa Modem Passthrough Fax Passthrough CNG Tone Detection Fax Tone Detection DTMF Signal to Noise	Digit Relay Digit (DTMF) Relay Type RFC 2833 Digit Relay Payload Type 101 Insthrough/Tone Detection Enabled Enabled Enabled Enabled Enabled 0

Figure 19: From Skype

Description Admin State	Remove 00 add + Enabled		
Match Type	Optional (Match One)		
	Input Field		Output Field
Type C a	Input Field	Туре	Output Field Called Address/Number



6. Call Routing Table

Select Settings > 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). Routes are defined by Call Routing Tables, which allow 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).

Figure 20: From Colt

	Route	Details	
Descript Admin St Route Prio Call Prio Number/Name Transformation Ta Time of Day Restrict	tion From Colt to tate Enabled rity 1 rity Normal able From Colt to tion None	Fax	
	Destination	Information	
Destination Type Message Translation Table Cause Code Reroutes Cancel Others upon Forwarding Fork Call Destination Signaling Groups Enable Maximum Call Duration	Normal None Disabled Not Licensed (SIP) To/From Fa	X *	
Media		Quality of S	Service
Audio/Fax Stream Mode Video/Application Stream Mode Media Transcoding Media List	DSP Disabled Enabled None	Quality Metrics Number Quality Metrics Time Befor Min. ASR Th Enable Min MOS Th Enable Max. R/ Max. R/ Enable Max Max	of Calls 10 re Retry 10 meshold 0 meshold Disabled /T Delay Enabled /T Delay 65535 ax. Jitter Enabled ax. Jitter 3000

	Route	Details	
Descript Admin St Route Prio Call Prio Number/Name Transformation Ta Time of Day Restrict	tion tate Enabled rity 1 rity Normal able From Colt tion None		
	Destination	n Information	
Destination Type Message Translation Table Cause Code Reroutes Cancel Others upon Forwarding Fork Call Destination Signaling Groups Enable Maximum Call Duration	Normal None Disabled Not Licensed (SIP) To/From Sk	cype ★ *	
Media		Quality of Service	
Audio/Fax Stream Mode Video/Application Stream Mode Media Transcoding Media List	DSP Disabled Enabled None	Quality Metrics Number of Calls Quality Metrics Time Before Retry Min. ASR Threshold Enable Min MOS Threshold Enable Max. R/T Delay Max. R/T Delay Enable Max. Jitter Max. Jitter	10 10 0 Disabled Enabled 65535 Enabled 3000

Figure 21: From Skype to Colt

	Route Details					
Descript Admin St Route Prio Call Prio Number/Name Transformation Ta Time of Day Restrict	Description Admin State Enabled Route Priority 1 Call Priority Normal Number/Name Transformation Table From Skype to Colt Time of Day Restriction None					
	Destination	Information				
Destination Type Message Translation Table Cause Code Reroutes Cancel Others upon Forwarding Fork Call Destination Signaling Groups Enable Maximum Call Duration	Normal None Disabled Not Licensed (SIP) To/From Co Disabled	olt 🔹	*			
Media		Quality	y of Service			
Audio/Fax Stream Mode Video/Application Stream Mode Media Transcoding Media List	DSP Disabled Enabled None	Quality Metrics Nur Quality Metrics Time Min. As Enable Min MC Enable Ma Ma Enabl	mber of Calls Before Retry SR Threshold DS Threshold ax. R/T Delay ax. R/T Delay le Max. Jitter Max. Jitter	10 10 0 Disabled Enabled 65535 Enabled 3000		

7. Message Rule Tables

Select Settings > Message Manipulation > Message Rule Tables

Message Rule Tables are sets of Condition Rules and are applied in SIP Signaling Groups when Message Manipulation is enabled.

Figure 22: Colt Outbound

Description Condition Expression Admin State Result Type	Replace NEWPORT.lync2013.sonusnet.com with 216.110.2.229 Enabled Optional
Match Regex NE	WPORT.lync2013.sonusnet.com *
Replace Regex 210	5.110.2.229 *

8. Signaling Groups

Select Settings > Signaling Groups

Signaling Groups allow telephony channels to be grouped together for the purposes of routing and shared configuration. These groups are the entity to which calls are routed, as well as 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, this will specify protocol settings and link to server, media, and mapping tables.

Figure 23: Internal Side

Description To/From Skype Admin State Enabled Service Status Up		
SIP	Channels and Routing	
Action Set Table Call Routing Table No. of Channels SIP Profile Agent Type Interop Mode SIP Server Table Load Balancing Channel Hunting Notify Lync CAC Profile Challenge Request Outbound Proxy IP/FQDN Outbound Proxy IP/FQDN Outbound Proxy IP/FQDN No Channel Available Override Call Setup Response Timer Call Setup Response Timer Call Proceeding Timer QoE Reporting	None From Skype to Colt 60 Default SIP Profile Basic Call Back-to-Back User Agent Standard Skype Round Robin Most Idle Disable 5060 34: No Circuit/Channel Available 255 180 Disabled	Supported Audio/Fax Modes DSP Proxy Direct * Supported Video/Application Modes * * Media List ID Skype Media List * Media List ID Skype Media List * Play Ringback Auto on 180 * Tone Table Default Tone Table * Play Congestion Tone Disable * Allow Refresh SDP Enable * Music on Hold Disabled * Multiplexing Disable *
Forked Call Answered Too Soon	Disable	Mapping lables 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 (10.35.177.232) Signaling DSCP 40 NAT Traversal
	Listen Ports	Federated IP/FQDN
Port Protocol 5060 UDP 5060 TCP	TLS Profile ID N/A N/A	IP/FQDN Netmask/Prefix fe.skype2015.sonusnet.com 255.255.255
Message Manipulation Disabled		

Description To/From Fax		
Service Status Up		
SIP	Channels and Routing	Media Information
Action Set Table	None	Media information
Call Routing Table	From Skype to Colt	DSP
No. of Channels	60	Supported Audio/Fax Modes
SIP Profile	Default SIP Profile	Direct
SIP Mode	Basic Call	Supported
Agent Type	Back-to-Back User Agent	Video/Application *
Interop Mode	Standard	Modes
SIP Server Table	Fax	Media List ID Fax Media List
Load Balancing	Round Robin	Play Ringback Auto on 180
Channel Hunting	Most Idle	Tone Table Default Tone Table
Notify Lync CAC Profile	Disable	Play Congestion Tone Disable
Challenge Request	Disable	Farly 183 Disable
Outbound Proxy IP/FQDN		Allow Refresh
Outbound Proxy Port	5060	SDP Enable
No Channel Available Override	34: No Circuit/Channel Available	Music on Hold Disabled
Call Setup Response Timer	255	RTCP Disable
Call Proceeding Timer	180	Multiplexing
QUE Reporting	Disabled	
Earling Call Approvated Tap Soon	Enable	Mapping Tables
Forked Call Answered 100 Soon	Disable	
		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 ID Esternat 1 ID (10 35 177 333)
		Signaling DSCP 40
		NAT Traversal
		ICE Support Disabled
		Static NAT - Outbound
		Outbound NAT Traversal None
		Static NAT - Inbound
		Detection Disabled
	Listen Ports	Federated IP/FQDN
Total 2 SIP Listen Port Rows		Total 2 SIP Federated IP Rows
Port Protocol	TLS Profile ID	IP/FQDN Netmask/Prefix
5060 UDP	N/A	10.35.137.43 255.255.255
5060 TCP	N/A	10.35.137.106 255.255.255
Message Manipulation Disabled		

Figure 24: External Side

r						
Description To/From Colt						
Admin State Enabled						
Service Status Up						
SI	P Channels and Routing			Madia Informa	tion	
Action Set Table	None			Media informa	luon	
Call Routing Table	From Colt			DSP		
No. of Channel	60		Support Audio/Eax Mod	ed Proxy		*
SIP Profile	Colt SIP Profile			Direct		•
SIP Mode	Basic Call		Support	ed Proxy		
Agent Type	Back-to-Back User Agent		Video/Applicati	ion Direct		*
Interop Mode	Standard		Moc			·
SIP Server Table	Colt		Media List	ID Colt Media List		
Load Balancing	Round Robin		Play Ringba	Ack Auto on 180		
Notify Lync CAC Profile	Dischle		Ione Ial	DIE Default Tone Tab	le	
Challenge Reques	Disable		To	ne Disable		
Outbound Proxy IP/FOD			Early 1	83 Disable		
Outbound Proxy Por	5060		Allow Refre	sh Enable		
No Channel Available Override	34: No Circuit/Channel Ava	ilable	S	DP =		
Call Setup Response Time	255		Music on Ho	Old Disabled		
Call Proceeding Time	r 180		Multiplexi	ng Disable		
QoE Reporting	Disabled					
Use Register as Keep Alive	Enable			Mapping Tabl	es	
Forked Call Answered Too Soor	Disable			mapping inst		
			SIP To Q	.850 Override Table	Default (RFC4497	Ŋ
			Q.850 To	SIP Override Table	Default (RFC4497	0
			Pass-thru Peer	SIP Response Code	Enable	
				SIP IP Details	;	
			Signaling/Media	Source IP Ethernet	3 IP (216.110.2.22	9)
			Signa	aling DSCP 40		
				NAT Traversal		_
			IC	CE Support Disabled		
				Static NAT - Outbo	und	_
			Outbound NA	T Traversal None		
				-Static NAT - Inbou	und	_
				Detection Disabled		
	Listen Darte			Fodorated ID/FC		
	Listen Forts			redelated in/ro		
Total 1 SIP Listen Port Row			Total 1 SIP Federate	ed IP Row		
Port Protocol	TLS Profile ID		IP/FODN	Netr	nask/Prefix	
5060 UDD	NI/A		217 110 220 11	1 255	255 255 255	
5060 0DF	N/A		217.110.230.11	1 255.	.255.255.255	
Message Manipulation Enabled						
Inhou	nd Message Manipulation			bound Message Ma	ninulation	
uodni	na message manipulation		Uut	assand message Ma	mpulation	
				Replace NEWPORT.lyn	c2013.sonusnet.	•
	A			Replace NEWPORT.lyn	ic2013.sonusnet.	
Message Table List		*	Message Table List	Replace NEWPORT.lyn	ac2013.sonusnet.cz	*

Test Results

Table 2: Test Results

S. No	Procedure	Observation	Result	Comment
----------	-----------	-------------	--------	---------

Basic	Calls		
1-1	IP Phone to PSTN Phone, IP Phone disconnect after answer	PASS	
1-2	IP Phone to PSTN Phone, PSTN Phone busy	PASS	
1-3	IP Phone to PSTN Phone, PSTN Phone no answer	PASS	
1-4	IP Phone to PSTN Phone, PSTN Phone disconnect after answer	PASS	
1-5	PSTN Phone to IP Phone, PSTN Phone disconnect after answer	PASS	
1-6	PSTN Phone to IP Phone, IP Phone busy	N/A	SfB doesn't send 486 Busy Here
1-7	PSTN Phone to IP Phone, IP Phone no answer	PASS	
1-8	PSTN Phone to IP Phone, IP Phone disconnect after answer	PASS	
1-9	PSTN Phone to IP Phone, network disconnect	PASS	
1-11	IP Phone to International Mobile	PASS	
1-12	IP Phone to International PSTN Phone, remote ringback	PASS	
1-13	IP Phone to PSTN Phone, Long Duration Call	PASS	
1-14	IP Phone to PSTN Phone, Mute both ends of call	PASS	
Enhan	iced Calls	I	
2-1	PSTN 1 to IP Phone A1, A1 blind transfers to PSTN 2	PASS	
2-2	PSTN 1 to IP Phone A1, A1 consultative transfers to PSTN 2	PASS	
2-3	PSTN 1 to IP Phone A1, A1 forwards to PSTN 2, Unconditional	PASS	
2-4	PSTN 1 to IP Phone A1, A1 forwards to busy PSTN 2, Unconditional	PASS	
2-5	PSTN 1 to IP Phone A1, A1 forwards to IP Phone A2, Unconditional	PASS	
2-6	PSTN 1 to IP Phone A1, A1 forwards to Mobile, Unconditional	PASS	
2-7	PSTN 1 to IP Phone A1, A1 forwards to PSTN 2, No Answer	PASS	
2-8	PSTN 1 to IP Phone A1, A1 forwards to IP Phone A2, No Answer	PASS	
2-9	PSTN 1 to IP Phone A1, A1 forwards to Mobile, No Answer	PASS	
2-10	IP Phone A1 to PSTN 1, A1 conference to PSTN 2, after answer	PASS	
2-11	IP Phone A1 to PSTN 1, A1 conference to IP Phone A2, after answer	PASS	
2-12	IP Phone A1 to PSTN 1, A1 conference to IP Phone A2, mixed codecs	PASS	
Codec	Support		

4-1	IP Phone to PSTN Phone, G.729 codec	PASS	
4-2	IP Phone to PSTN Phone, G.711 alaw codec	PASS	
4-3	PSTN Phone to IP Phone, G.729 codec	PASS	
4-4	PSTN Phone to IP Phone, G.711 alaw codec	PASS	
4-5	IP Phone to PSTN Phone, G.726 32K codec	PASS	
4-6	PSTN Phone to IP Phone, G.726 32K codec	PASS	
4-7	IP Phone to PSTN Phone, G.711 Ulaw codec	PASS	
4-8	PSTN Phone to IP Phone, G.711 Ulaw codec	PASS	
4-9	IP Phone to PSTN Phone, iLBC codec	NOT SUPPOR TED	iLBC not supported by both SBC and SfB
4-10	PSTN Phone to IP Phone, iLBC codec	NOT SUPPOR TED	iLBC not supported by both SBC and SfB
4-11	IP Phone to IP Phone, G.722 codec	NOT SUPPOR TED	SfB uses SILK for peer to peer if no bandwidth limitations are applied or detected, otherwise will use RTA for low bandwidth
DTMF	Support		
5-1	IP Phone to PSTN Phone, DTMF using RFC2833	PASS	
5-2	PSTN Phone to IP Phone, DTMF using RFC2833	PASS	
5-3	IP Phone to PSTN Phone, DTMF using H.245 Signal	NOT SUPPOR TED	SfB doesn't support it
5-4	PSTN Phone to IP Phone, DTMF using H.245 Signal	NOT SUPPOR TED	SfB doesn't support it
5-5	IP Phone to PSTN Phone, DTMF using H.245 Alphanumeric	NOT SUPPOR TED	SfB doesn't support it
5-6	PSTN Phone to IP Phone, DTMF using H.245 Alphanumeric	NOT SUPPOR TED	SfB doesn't support it
5-7	IP Phone to PSTN Phone, DTMF Before Answer	PASS	
CLI Se	rvices		
6-1	Caller ID Presentation (CLIP) with No Screening	PASS	
6-2	Caller ID Presentation (CLIP) Screening with Correct CLI	PASS	
6-3	Caller ID Presentation (CLIP) Screening with Incorrect CLI	PASS	
6-4	IP Phone to PSTN Phone, Caller ID Restriction (CLIR)	PASS	
6-5	PSTN Phone to IP Phone, Caller ID Restriction (CLIR)	PASS	
Encry	otion		

7-1	IP Phone to PSTN Phone, TLS + RTP	PASS	
7-2	PSTN Phone to IP Phone, TLS + RTP	PASS	
7-3	IP Phone to PSTN Phone, TLS + SRTP	PASS	
7-4	PSTN Phone to IP Phone, TLS + SRTP	PASS	

Conclusion

These Application Notes describe the configuration steps required for the Ribbon SBC 1000/2000 to successfully interoperate with Skype for Business 2015. All feature and serviceability test cases were completed and passed with the exceptions/observations noted in Test Results.

Appendix A - TLS/SRTP Configuration

Figure 25: TLS Profile

	TLS Parameters	
	Common Attributes	
TLS Protocol	TLS 1.0-1.2	
Mutual Authentication	Disabled	
landshake Inactivity Timeout	10	
	Client Attributes	
Client Cipher List	TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384 TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256 TLS_ECDHE_RSA_WITH_3DES_EDE_CBC_SHA TLS_RSA_WITH_AES_256_CBC_SHA256 TLS_RSA_WITH_AES_128_CBC_SHA256 TLS_RSA_WITH_AES256_CBC_SHA TLS_RSA_WITH_AES128_CBC_SHA	
Verify Peer Server Certificate	Disabled	

Figure 26: SRTP Profile

SRTP Config				
Description Operation Option Crypto Suite	COLT SRTP Profile Required AES_CM_128_HMAC_SHA1_80			
Master Key				
Master Key Lifetime	Never Expires			
Derivation Rate	0			
Key Identifier Length	0			

Figure 27: Colt SIP Server

	Server Host	t		Transp	ort
Server Lookup Priority Host FQDN/IP Port Protocol TLS Profile	IP/FQDN 1 217.110.230 5061 TLS Colt TLS Pro	.111 file		Monitor None	
Remote Authorization and Contacts		٦	Connection Reuse		
Remote Authorization Table None Contact Registrant Table None Session URI Validation Liberal			Reuse Sockets Reuse Timeout	True 4 Forever	

Figure 28: Colt Media List

1edia List Details: C	olt Media List	
Description	Colt Media List	
	Default G711A	
	Default G711u	
Madia Drafilas List	T.38	
Media Profiles List		
	~	
SDES-SRTP Profile	COLT SRTP Profile	
DTLS-SRTP Profile	None	
Media DSCP	46	
RTCP Mode	RTCP	
Dead Call Detection	Disabled	
Silence Suppression	Disabled	
Gain Control	Digit Relay	
Receive Cain 0	Digit (DTME) Relay Type PEC 2022	
Transmit Gain 0	Digit Relay Payload Type 101	
Pa	ssthrough/Tone Detection	
Modem Passtbrough	Fushlad	
Fax Passthrough	Enabled	
CNG Tone Detection	Enabled	
Fax Tone Detection	Enabled	
DTMF Signal to Noise	0	
DTMF Minimum Level	-38	

Figure 29: Colt Signaling Group

r				
Description To/From Colt				
Admin State Enabled				
Service Status Up				
SIF	Channels and Routing			
		Media Information		
Action Set Table	None			
Call Routing Table	From Colt	DSD		
No. of Channels	60	Supported Proxy *		
SIP Profile	Colt SIP Profile	Audio/Fax Modes Direct		
SIP Mode	Basic Call	Draw		
Agent Type	Back-to-Back User Agent	Supported Proxy Video/Application Direct *		
Interop Mode	Standard	Modes		
SIP Server Table	Colt TLS	Media List ID Colt Media List		
Load Balancing	Bound Robin	Play Ringback Auto on 180		
Channel Hunting	Most Idle	Tone Table Default Tone Table		
Notify Lync CAC Profile	Disable	Play Congestion		
Challenge Request	Disable	Tone		
Outbound Proxy IP/FODN		Early 183 Disable		
Outbound Proxy Port	5060	Allow Refresh		
No Channel Available Override	24 No Circuit/Channel Available	SDP Chable		
Call Setup Response Timer	255 S4. No Circuit/Channel Avanable	Music on Hold Disabled		
Call Proceeding Timer	233	RTCP Multiplexing Disable		
	180	Multiplexing		
QOE Reporting	Disabled			
Use Register as Reep Alive	Enable	Mapping Tables		
Forked Call Answered 100 Soon	Disable			
		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		
		SIP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40		
		SIP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP MAT Traversal ICE Support Disabled Outbound NAT Traversal None Static NAT - Inbound Detection Disabled		
	Listen Ports	SIP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP MAT Traversal ICE Support Disabled Static NAT - Outbound Outbound NAT Traversal None Static NAT - Inbound Detection Disabled		
Total 1 SIP Listen Port Row	Listen Ports	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP MAT Traversal ICE Support Disabled Static NAT - Outbound Outbound NAT Traversal None Static NAT - Inbound Detection Disabled		
Total 1 SIP Listen Port Row Port Protocol	Listen Ports	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP MAT Traversal ICE Support Disabled Static NAT - Outbound Outbound NAT Traversal None Static NAT - Inbound Detection Disabled		
Total 1 SIP Listen Port Row Port Protocol 5061 TLS	Listen Ports TLS Profile ID Colt TLS Profile	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40 MAT Traversal ICE Support Disabled Outbound Outbound <td <="" colspan="2" th=""></td>		
Port Protocol 5061 TLS	Listen Ports TLS Profile ID Colt TLS Profile	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40 NAT Traversal ICE Support Disabled Outbound Outbound <td <="" colspan="2" th=""></td>		
Port Protocol 5061 TLS Message Manipulation Enabled	Listen Ports TLS Profile ID Colt TLS Profile	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40 NAT Traversal ICE Support Disabled Outbound Outbound Outbound Outbound MAT Traversal None Detection Disabled Detection Disabled <td col<="" th=""></td>		
Total 1 SIP Listen Port Row Port Protocol 5061 TLS Message Manipulation Enabled	Listen Ports TLS Profile ID Colt TLS Profile d Message Manipulation	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40 MAT Traversal ICE Support ICE Support Disabled Static NAT - Outbound Outbound NAT Traversal Outbound NAT Traversal None Static NAT - Inbound Detection Detection Disabled Total 1 SIP Federated IP Row IP/FQDN Netmask/Prefix 217.110.230.111 255.255.255.255		
Port Protocol 5061 TLS Message Manipulation Enabled	Listen Ports TLS Profile ID Colt TLS Profile d Message Manipulation	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40 NAT Traversal ICE Support Disabled Outbound Outbound IP/FQDN Total 1 SIP Federated IP Row IP/FQDN Netmask/Prefix 217.110.230.111 Outbound Message Manipulation		
Total 1 SIP Listen Port Row Port Protocol 5061 TLS Message Manipulation Enabled Inboun Message Table List	Listen Ports TLS Profile ID Colt TLS Profile d Message Manipulation	SiP IP Details Signaling/Media Source IP Ethernet 3 IP (216.110.2.229) Signaling DSCP 40 MAT Traversal ICE Support Disabled Get Static NAT - Outbound Outbound NAT Traversal None Static NAT - Inbound Detection Disabled Detection Disabled Static NAT - Inbound Static NAT - Inbound Static NAT - Inbound Detection Disabled Static NAT - Inbound Static NAT - Inbound Static NAT - Inbound Static NAT - Inbound Static NAT - Inbound Static NAT - Inbound Static NAT - Inbound Message Table List Replace NEWPORT.lync2013.sonusnets *		