# Guide for BTIP and Business Talk SIP services Microsoft

# Lync 2013

# Skype for Business 2015

# Skype Online - Cloud Connector Edition

28 march 2019

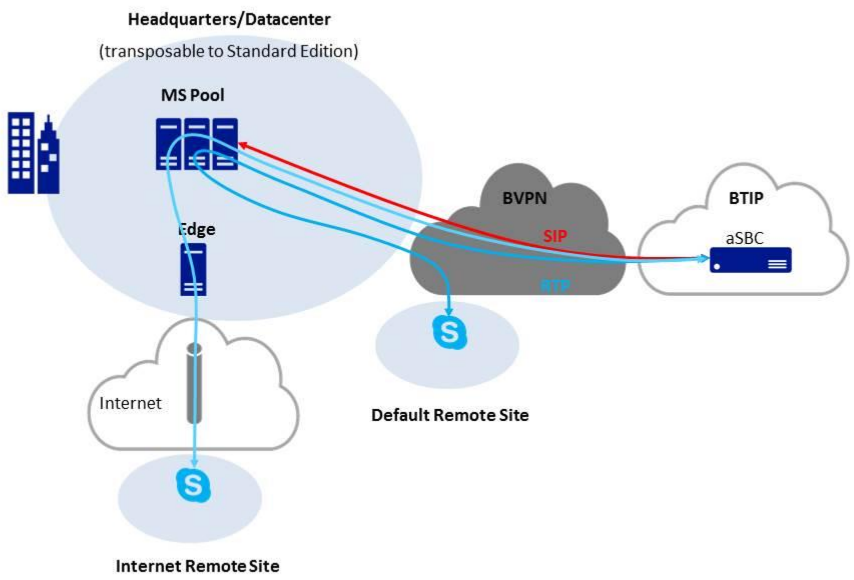
Lync 2013/AudioCodes/Ribbon Checklist 1.6
Skype for Business 2015/AudioCodes/Ribbon Checklist 1.11
Cloud Connector Edition AudioCodes Checklist 2.0

# Contents

# 1 Main certified architectures

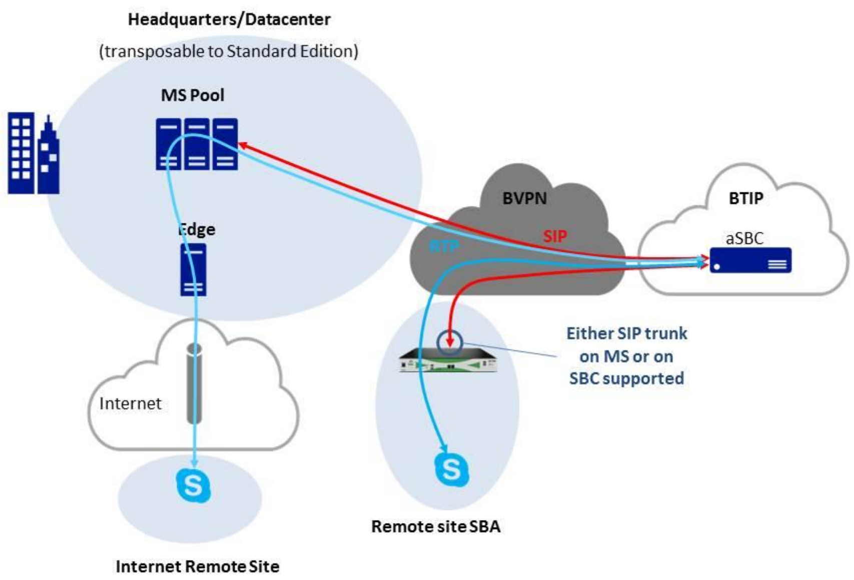
## 1.1 Lync 2013 & Skype for Business 2015 on premises

### 1.1.1 Centralized architecture

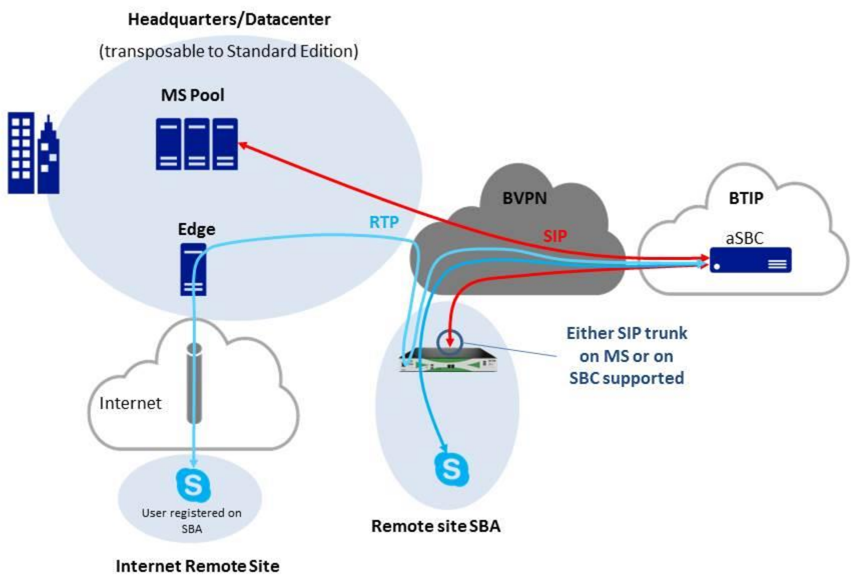


### 1.1.2 Remote site “SBA”

*Example 1*




*Example 2*

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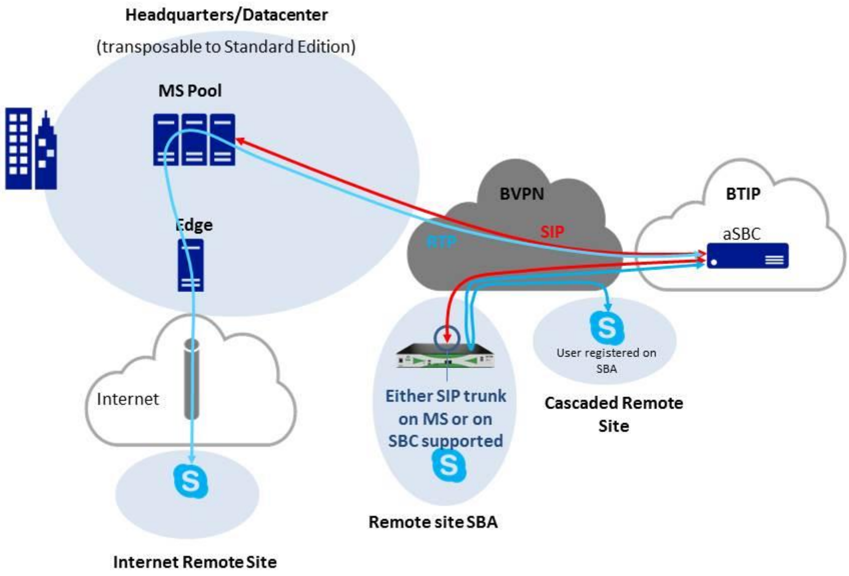
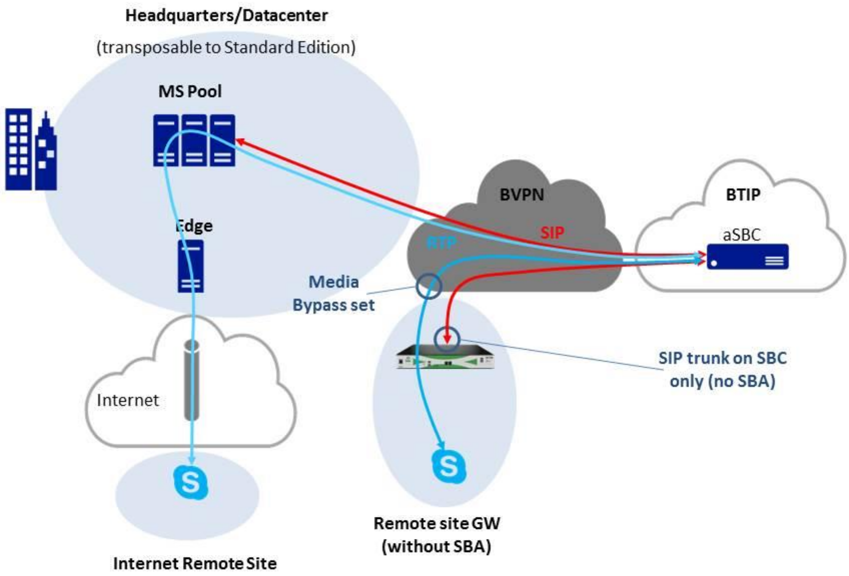
### 1.1.3 “Cascaded” remote site




### 1.1.4 Remote site “GW”

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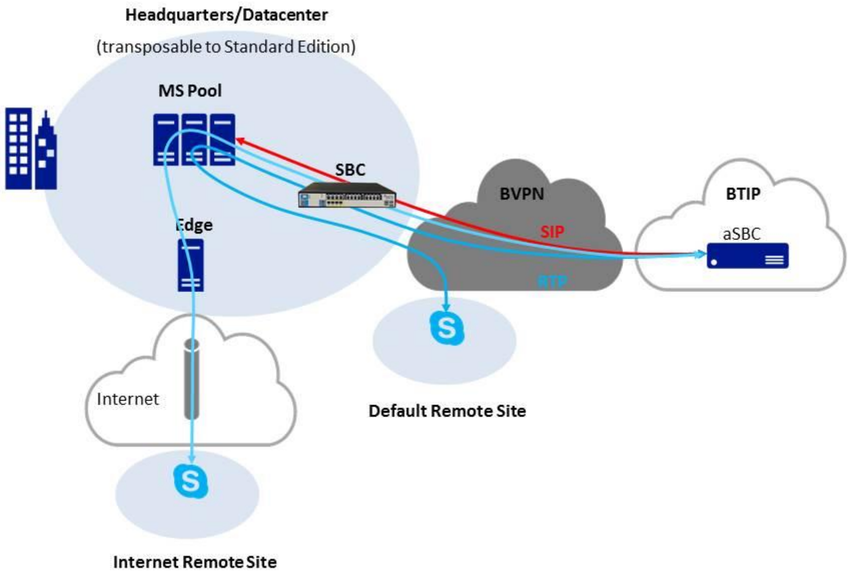
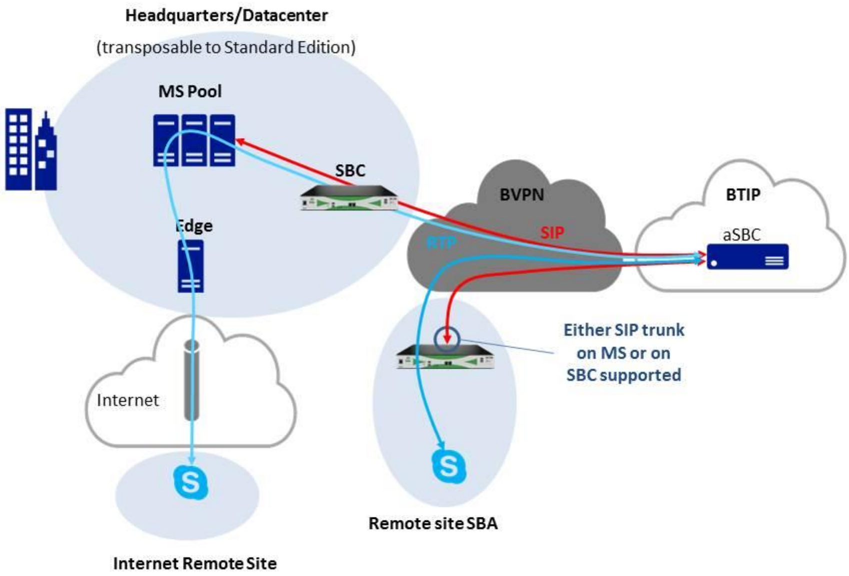
### 1.1.5 Centralized architecture with central SBC




### 1.1.6 Remote site “SBA” and central site with central SBC

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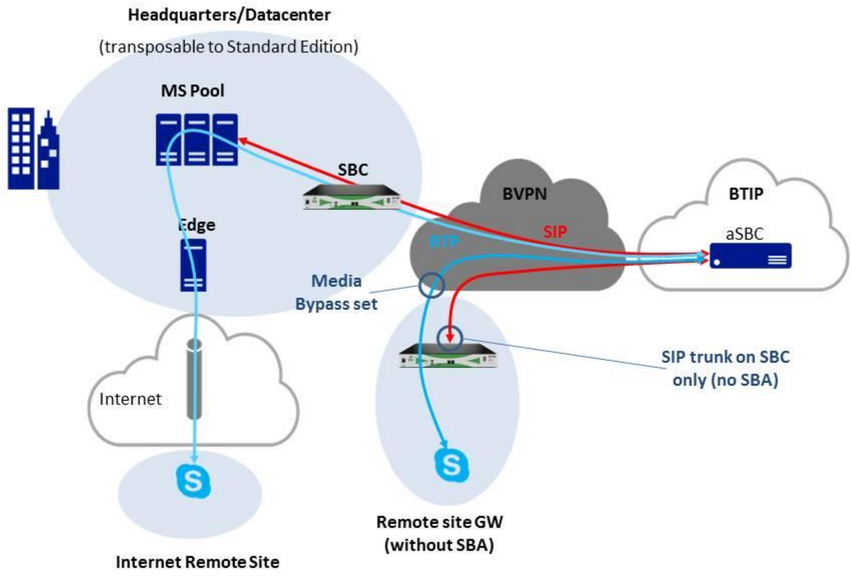
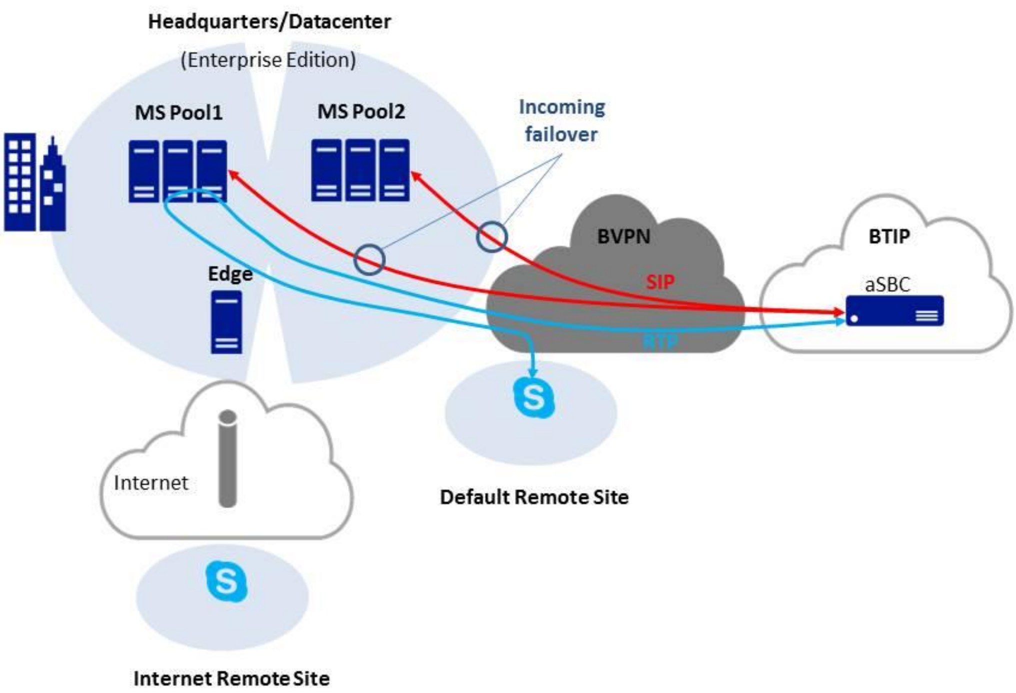
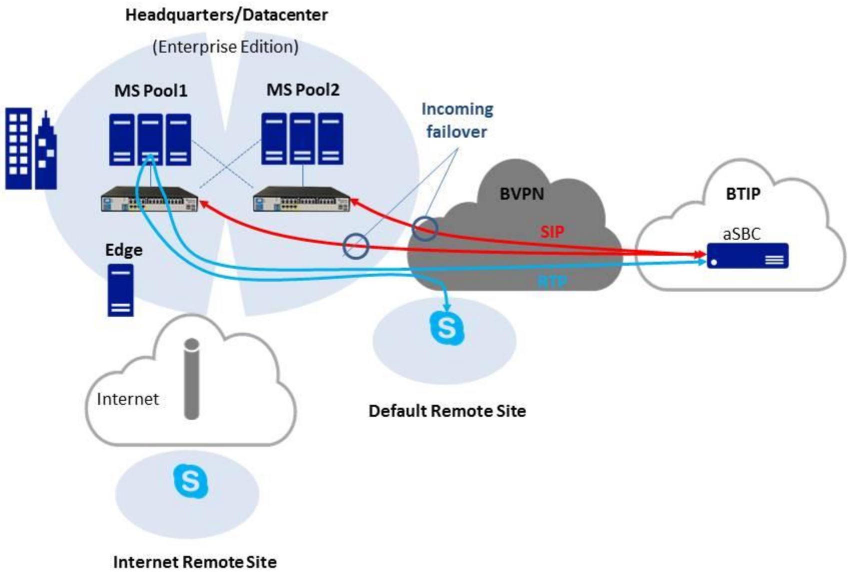
### 1.1.7 Remote site "GW" and central site with central SBC




### 1.1.8 2-pool centralized architecture



### 1.1.9 2-pool architecture with central SBC (Customer specific)




### 1.1.10 FAX

FAX is certified on Business Talk (International Scope) with Skype for Business 2015 and Sonus (Ribbon) GW on SBA remote sites only. Configuration checklist is not available yet but a configuration guide is available (not in this document). French scope (BTIP) is in progress.

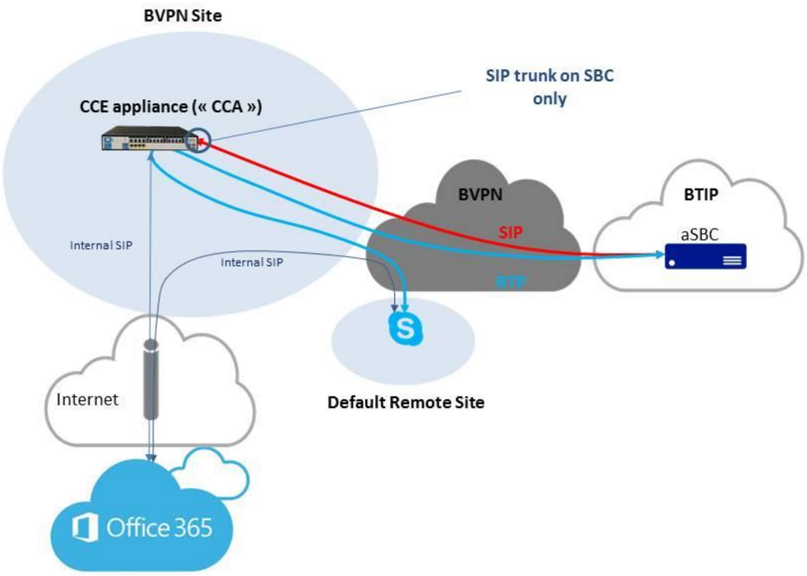
FAX on AudioCodes GW certification is in progress both on French (BTIP) and International (BTalk) scopes.

FAX protocol is T.38.

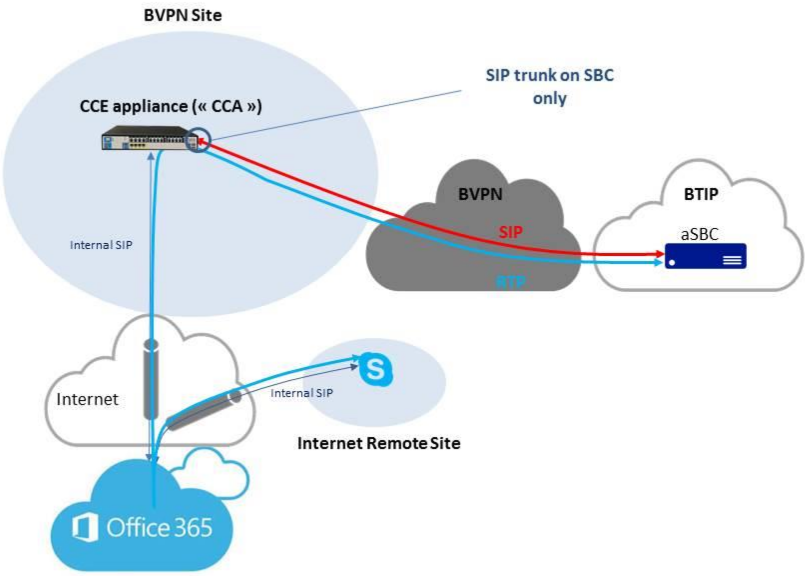
## 1.2 Skype for Business Online

### 1.2.1 Standalone mode

*Example 1 – offnet call from a BVPN remote site*

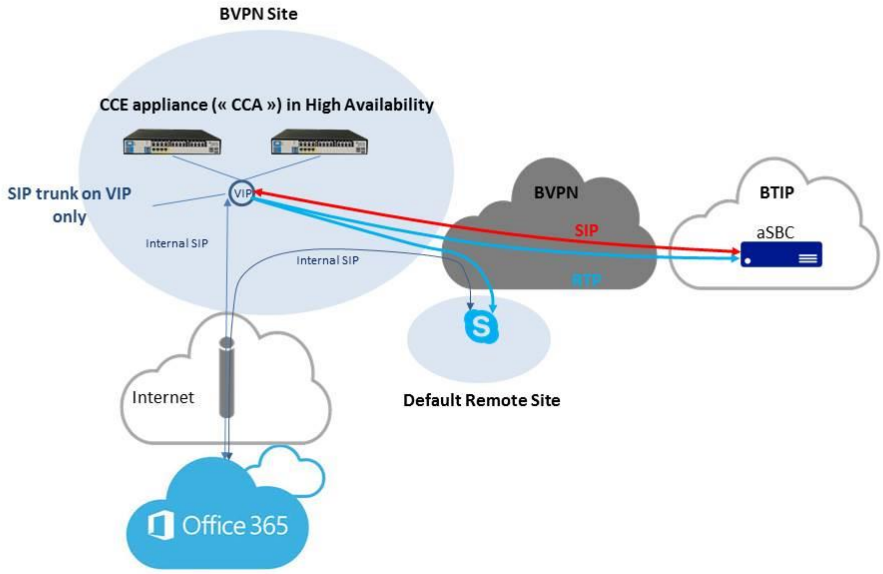



*Example 2 – offnet call from an Internet remote site*



### 1.2.2 Redundant architectures

*Example: high-availability*




*Round-Robin & Nominal/Backup also certified*

# 2 Parameters for connection to BTIP

## 2.1 On-premise architectures

| Head Quarter (HQ) architecture | Level of Service | @IP used by the service | |
|---|---|---|---|
| Standard Edition<br>Enterprise Edition | No redundancy | MS IP@ | |
| Standard Edition pairing<br>100% users on nominal | Local Server redundancy with database replication<br>2 Mediation Servers (MS1, MS2) | MS1 IP@ | MS2 IP@ |
| 2x Standard Edition Pairing<br>50% users registered on nominal of each pair | Offers the same Level Of Service as 1xSE Pairing, but increases the capacity<br>2 Mediation Servers (MS) per pair.<br>Round robin between pairs from incoming calls, even in case of loss of one SE<br>Pair1 : MS1+MS2<br>Pair2 : MS3+MS4 | MS1 IP@<br>MS3 IP@ | MS2 IP@<br>MS4 IP@ |
| Enterprise Edition | Load balancing (one pool)<br>Single pool of Y Mediation Servers (MS) on the same site (Y>1) | MS1 IP@<br>…<br>MSY IP@ | |
| Enterprise Edition | - Local pool redundancy:<br>- 2 Pools of Y and Y' Mediation Servers (MS) on the same site (Y>=1, Y'>=1)<br>OR<br>- Geographical pool redundancy (same region)<br>- 2 Pools of Y and Y' Mediation Servers (MS), each Pool hosted by different sites (Y>=1, Y'>=1) | Pool1_MS1 IP@<br>…<br>Pool1_MSY IP@ | Pool2_MS1 IP@<br>…<br>Pool2_MSY' IP@ |
| Central trunk with central SBC | No redundancy<br>SBC without SBA on HQ acting as a customer SBC for HQ SIP trunk only | SBC IP@ | |

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| Remote Site (RS) architecture | Level of Service | @IP used by the service |
|---|---|---|
| Default remote site | No survivability, no trunk redundancy | N/A |
| Remote site with Mediation Server | No hairpinning through central site<br>Functionning mode:<br>- users remain registered to HQ<br>- SIP trunk is handled by local MS<br>- Nominal ougoing and incoming traffic goes through MS | MS IP@ |
| Remote site with Gateway-SBA (Survivability Branch Appliance) or SBS (Survivability Branch Server) | - Remote survivability for the site hosting the Gateway-SBA or SBS<br>Functionning mode:<br>- SIP trunk is handled by SBA (not SBC part) or SBS<br>- Nominal ougoing and incoming traffic goes through SBA/SBS<br>- In Case of SBA/SBS crash or Local SIP Trunk connectivity loss to a-SBC, remote site phones will re-register on HQ and attempt to use the HQ trunk for incoming and outgoing traffic | SBA MS or SBS MS IP@ |
| Remote site with Gateway-SBA (Survivability Branch Appliance) | - Remote survivability for the site hosting the Gateway-SBA<br>Functioning mode:<br>- SIP trunk is handled by a-SBC part of the appliance (not MS part)<br>- Nominal outgoing and incoming traffic goes through a-SBC<br>- In case of SBA/SBS crash or Local SIP Trunk connectivity loss to a-SBC, remote site phones will re-register on HQ and attempt to use the HQ trunk for incoming and outgoing traffic | SBC IP@ |
| Remote site of "RS-GW" type (Gateway without SBA module) | - Allows local users to use local trunk though they are registered on central HQ (Microsoft "Media-Bypass" feature set locally)<br>- Save bandwidth on central HQ | |
| Remote site cascaded to Remote site with Gateway-SBA or SBS | Allows hairpinning through the closest SBA/SBS instead of through HQ | N/A |

## 2.2 Cloud Connector Edition architectures

| Head Quarter (HQ) architecture | Level of Service | @IP used by the service | |
|---|---|---|---|
| CCE with SBC - Trunk on SBC | No redundancy | SBC IP@ | |
| Dual CCE-SBC - Trunk on SBC - High Availability with single @IP | Redundancy with load balancing behavior | SBCs virtual IP@ | |
| Dual CCE-SBC - Trunk on SBC - Resiliency | Redundancy with nominal/backup behavior | SBC1 IP@ | SBC2 IP@ |

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# 3 BTIP/BTalk certified versions

## 3.1 Lync 2013

Certified Lync Server Cumulative Updates:

- CU January 2017
- CU August 2016
- CU April 2016
- CU January 2016
- CU July 2015
- CU February 2015
- CU November 2014
- CU September 2014
- CU January 2014
- CU October 2013
- CU February 2013
- RTM

Certified SBC:

- Sonus (Ribbon) SBC 1000/2000 6.1
- Sonus (Ribbon) SBC 1000/2000 6.0.1.build 441
- Sonus (Ribbon) SBC 1000/2000 5.0
- Sonus (Ribbon) SBC 1000/2000 4.1

## 3.2 Skype for Business 2015

Certified Skype for Business 2015 Cumulative Updates:

- CU January 2019 (in progress)
- CU December 2017
- CU May 2017
- CU June 2016
- CU March 2016
- CU November 2015
- RTM

Certified SBC:

- Ribbon SBC 1000/2000 8.0 (in progress)
- Ribbon SBC 1000/2000 7.0
- Sonus (Ribbon) SBC 1000/2000 6.1
- Sonus (Ribbon) SBC 1000/2000 6.0.1 build 441
- Sonus (Ribbon) SBC 1000/2000 5.0.1 build 399
- AudioCodes M800/1000 7.20A
- AudioCodes M800/1000 7.00A

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## 3.3 Cloud Connector Edition

Certified devices and software:

- Mediation Server 6.0.9319.410
- CCE AudioCodes appliance (Wizard version) V2.1.0.19
- CCE AudioCodes Mediant software 7.2

Cloud Connector Edition is no longer supported for new deployments. Consider Microsoft Teams instead.

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# 4 Lync 2013 Configuration Checklist

| Menu | Value |
|---|---|
| **DNS requirements** | |
| From the DNS interface:<br>✓ Start > Administrative Tools > DNS | FQDNs of each server (**DNS A** record) |
| From the DNS interface:<br>✓ Start > Administrative Tools > DNS | FQDNs of both nominal and backup aSBC on each site (**DNS A** record) |
| From the DNS interface:<br>✓ Start > Administrative Tools > DNS | **ucupdates-r2**.<*SIP domain*> (**DNS A** record) that maps the FQDN of each server hosting Device Update Service |
| From the DNS interface:<br>✓ Start > Administrative Tools > DNS | **_sipinternaltls._tcp**.<*SIP domain*> (**DNS SRV** record/**Port 5061**) that maps the FQDN of each server offering automatic client sign-in service |
| From the DNS interface:<br>✓ Start > Administrative Tools > DNS | **_ntp._udp**.<*SIP domain*> (**DNS SRV** record/**Port 123**) that maps the FQDN of the Domain Controller |
| **DHCP requirements** | |
| From the customer interface of the router | **Following command** has to **be typed** for each customer interface of the router:<br>✓ ip helper-address *"IP@ of the DHCP Server"* |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | **Following command** has to **be typed**:<br>✓ Set-CsRegistrarConfiguration **–EnableDHCPServer $True** |
| From the DHCP interface:<br>✓ Start > Administrative Tools > DHCP > "select a scope" > Scope Options | DHCP Option **006 DNS Servers** has to **be activated** |
| From the DHCP interface:<br>✓ Start > Administrative Tools > DHCP > "select a scope" > Scope Options | "*DHCPUtil.exe*" and "*DHCPConfigScript.bat*" **files*** have to **be added** on a network share that can be accessed from the DHCP server<br><br>*(*) DHCP Options **120 / 43** have to **be configured** (only if required by the type of endpoints deployed)* |
| From command prompt from the DHCP server:<br>✓ Start > Run… > cmd | **Following command** has to **be typed***:<br>✓ \\<FileShare>\DHCPUtil.exe -SipServer *"SipServer"* -WebServer *"WebServer"* –RunConfigScript<br><br>*(*) DHCP Options **120 / 43** have to **be configured** (only if required by the type of endpoints deployed)* |
| From the DHCP interface:<br>✓ Start > Administrative Tools > DHCP > "select a scope" > Scope Options | DHCP Option **042 NTP Servers** has to **be activated***<br><br>*(*) only if required by the type of endpoints deployed* |
| **AD requirements** | |
| From the AD interface:<br>✓ Start > Administrative Tools > Active Directory Users and Computers | **Each server role** has to **be joined to domain** |
| **Mediation Server Configuration** | |
| From the Microsoft Lync Server Topology Builder interface:<br><br>✓ Start > All Programs > Microsoft Lync | **TCP listening port** has to be set to **5060** |

| Menu | Value |
|---|---|
| Server 2013 > Lync Server Topology Builder<br>✓ Lync Server 2013 > "select a Central Site" > Mediation pools > "select a Mediation Server" | |
| | |
| **Enterprise Edition – Standalone Mediation Servers - Configuration** | |
| From the standalone Mediation Server:<br>✓ Start > Control Panel > Network and Internet > Network Connections > "select the interface of the Mediation Server" > Properties > Internet Protocol Version 4 (TCP/IPv4) | **Default gateway** has to be **filled**<br>**Preferred DNS server** has to be **filled** |
| From the standalone Mediation Server:<br>✓ Start > Control Panel > Network and Internet > Network Connections > "select the interface of the Mediation Server" > Properties > Internet Protocol Version 4 (TCP/IPv4) > Advanced… > DNS tab | **Register this connection's addresses in DNS** has to be **checked** |
| From the Microsoft Lync Server Topology Builder interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder<br>✓ Lync Server 2013 > "select an Enterprise Edition Central Site" > Mediation pools | 2 **Mediation pools** have to **be created** for 2 Standalone Mediation Servers:<br>✓ **Multiple computer pool with the Standalone Mediation Server pool 1 (=FQDN of the Mediation Server pool 1)**<br>✓ **Multiple computer pool with the Standalone Mediation Server pool 2 (=FQDN of the Mediation Server pool 2)**<br>**Enable TCP** port has to be **checked**<br>**Listening port** has to be set to **5060** for each standalone Mediation Server pool |
| From the Microsoft Lync Server Topology Builder interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder<br>✓ Lync Server 2013 > "select an Enterprise Edition Central Site" > Shared Components > PSTN gateways | 2 **PSTN gateways** have to **be created**<br>✓ 1st: **FQDN of Nominal aSBC** (Mediation server pool 1)<br>✓ 2nd: **FQDN of Backup aSBC** (Mediation server pool 1)<br><br>Check that **Use all configured IP addresses** is **selected** for each Mediation Server:<br>**Enable IPv4** has to **checked** and **Enable IPv6** has to be **unchecked** for each Mediation Server<br>Next window contains the **Trunk root** information as followed<br>**Listening port for IP/PSTGN gateway** has to be set to **5060**<br>**SIP Transport Protocol** has to be set to **TCP**<br>**Associated Mediation Server** has to match the **FQDN of Mediation Server pool 1** |
| From the Microsoft Lync Server Topology Builder interface:<br>✓ Start > All Programs > Microsoft Lync | 2 **Additional Trunks** have to **be created**<br>✓ 1st: **Associated PSTN gateway of Nominal aSBC** (Mediation server pool 2) |

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| Menu | Value |
|---|---|
| Server 2013 > Lync Server Topology Builder<br>Lync Server 2013 > "select an Enterprise Edition Central Site" > Shared Components > Trunks | ✓ 2nd: Associated PSTN gateway of Backup aSBC (Mediation server pool 2)<br>Listening port for IP/PSTGN gateway has to be set to 5060<br>SIP Transport Protocol has to be set to TCP<br>Associated Mediation Server has to match the FQDN of Mediation Server pool 2 |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Route | 4 Routes have to be created for 2 Standalone Mediation Servers*:<br>✓ from Standalone Mediation Server 1a to nominal aSBC (=FQDN of the nominal aSBC from the Mediation Server 1a)<br>✓ from Standalone Mediation Server 1b to backup aSBC (=FQDN of the backup aSBC from the Mediation Server 1b)<br>✓ from Standalone Mediation Server 2a to nominal aSBC (=FQDN of the nominal aSBC from the Mediation Server 2a)<br>✓ from Standalone Mediation Server 2b to backup aSBC (=FQDN of the backup aSBC from the Mediation Server 2b)<br>A gateway (=FQDN of the nominal aSBC from the Mediation Server 1a) has to be associated to First Route<br>A gateway (=FQDN of the backup aSBC from the Mediation Server 1b) has to be associated to Second Route<br>A gateway (=FQDN of the nominal aSBC from the Mediation Server 2a) has to be associated to Third Route<br>A gateway (=FQDN of the backup aSBC from the Mediation Server 2b) has to be associated to Fourth Route<br>A PSTN Usage has to be associated to each Route<br><br>*(*) Routes for a site Headquarter includes its Remote Sites without MGW* |
| Enterprise Edition – Standalone Mediation Servers – Specific configuration for Remote Site deployment | |
| From the Microsoft Lync Server Topology Builder interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder<br>✓ Lync Server 2013 > "select a Branch Sites" > Lync Server 2013 > Shared Components > PSTN gateways | 2 PSTN gateways have to be created for the Standalone Mediation Server:<br>✓ to nominal aSBC (=FQDN of the nominal aSBC)<br>✓ to backup aSBC (=FQDN of the backup aSBC)<br>Check that 2 Trunks were created while creating PSTN gateways<br>Listening port has to be set to 5060 for each PSTN gateways<br>SIP transport protocol has to be set to TCP for each PSTN gateways |
| From the Microsoft Lync Server Topology Builder interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder<br>✓ Lync Server 2013 > "select a Branch Sites" > Mediation pools | A Mediation pools has to be configured for the Standalone Mediation Server:<br>✓ One single computer pool (=FQDN of the Mediation Server)<br>2 PSTN Gateways have to be associated to the Standalone Mediation Server:<br>✓ FQDN of the nominal aSBC<br>✓ FQDN of the backup aSBC<br>Use all configured IPv4 IP addresses has to be checked:<br>Listening port has to be set to 5060 |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Dial Plan | A Site dial plan has to be created for each Remote site with a Standalone Mediation Server<br>A New Normalization Rule for extension numbers has to be associated:<br>✓ Pattern to match has to be edited<br>✓ Translation rule has to be edited<br>✓ Internal extension has to be checked<br>Normalization Rule for extension numbers has to be moved up before the |

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| Menu | Value |
|---|---|
| | existent Normalization Rule for Prefix All |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Voice Policy | An User policy has to be created for each Remote site with a Standalone Mediation Server<br>Enable call park has to be checked<br>Enable PSTN reroute has to be unchecked<br>A PSTN Usage has to be associated to each User policy |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Users > "select an user of Remote Site with a Standalone Mediation Server" | The specific voice policy has to be assigned to each RS (with a Standalone Mediation Server) user |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Route | 2 Routes have to be created for each Remote site with a Standalone Mediation Server :<br>✓ to nominal aSBC<br>✓ to backup aSBC<br>A gateway (=FQDN of nominal aSBC) has to be associated to First Route<br>A gateway (=FQDN of backup aSBC) has to be associated to Second Route<br>A PSTN Usage has to be associated to each Route |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Trunk Configuration | A Site trunk has to be created for each Remote site with a Standalone Mediation Server<br>Enable refer support has to be unchecked<br>Encryption support level has to be set to Optional<br>A Translation Rule (to remove digit "+" for outbound calls to BTIP SIP) has to be associated to each Site trunk |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | Following commands have to be typed for each Remote site with a Standalone Mediation Server:<br>✓ Set-CsTrunkConfiguration –Identity *"Site"* –RTCPActiveCalls $False<br>✓ Set-CsTrunkConfiguration –Identity *"Site"* –RTCPCallsOnHold $False |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Route | A PSTN Usage of Branch Sites has to be associated to each Route of Headquarter<br><br>Note that routes must be in the following order:<br>1) Route of Branch Sites to nominal aSBC<br>2) Route of Branch Sites to backup aSBC<br>3) Route of Headquarter to nominal aSBC<br>4) Route of Headquarter to backup aSBC |
| Users Configuration | |
| From the AD interface:<br>✓ Start > Administrative Tools > Active Directory Users and Computers<br>✓ New > User | User information (the user logon name) has to be filled |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control | Each user has to be assigned to a pool<br>Format <SAMAccountName>@<*SIP domain*> has to be selected<br>Telephony has to be set to Enterprise Voice<br>An *E164 telephone number* format followed by an extension number has to |

| Menu | Value |
| --- | --- |
| Panel<br>✓ Users > Enable users > Add… > Find | be entered in the **line URI** |
| **Routing mechanisms for Microsoft Lync Server 2013** | |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Dial Plan | A **Site dial plan** has to **be created for each site**<br>A **New Normalization Rule** for extension numbers has to **be associated**:<br>✓ **Pattern to match** has to **be edited**<br>✓ **Translation rule** has to **be edited**<br>✓ **Internal extension** has to **be checked**<br>**Normalization Rule** for extension numbers has to **be moved up** before the **existent Normalization Rule** for Prefix All<br><br>*(*) Site dial plan for a site Headquarter includes its Remote Sites without MGW* |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Voice Policy | A **Site policy** has to **be created for each site***<br>**Enable call park** has to **be checked**<br>**Enable PSTN reroute** has to **be unchecked**<br>**A PSTN Usage** has to **be associated to each Site policy**<br><br>*(*) Site policy for a site Headquarter includes its Remote Sites without MGW* |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Route | 2 **Routes** have to **be created for each site*** :<br>✓ **to nominal aSBC**<br>✓ **to backup aSBC**<br>A **gateway (=FQDN of nominal aSBC)** has to **be associated to First Route**<br>A **gateway (=FQDN of backup aSBC)** has to **be associated to Second Route**<br>A **PSTN Usage** has to **be associated to each Route**<br><br>*(*) Routes for a site Headquarter includes its Remote Sites without MGW* |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Trunk Configuration | A **Site trunk** has to **be created for each site***<br>**Enable refer support** has to **be unchecked**<br>**Enable forward call history** has to **be checked**<br>**Encryption support level** has to be set to **Optional**<br>A **Translation Rule** (to remove digit "+" for outbound calls to BTIP SIP) has to **be associated to each Site trunk**<br><br>*(*) Site trunk for a site Headquarter includes its Remote Sites without MGW* |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | **Following commands** have to **be typed for each site***:<br>✓ Set-CsTrunkConfiguration –Identity *"Site"* **–RTCPActiveCalls $False**<br>✓ Set-CsTrunkConfiguration –Identity *"Site"* **–RTCPCallsOnHold $False**<br><br>*(*) A Site Headquarter includes its Remote Sites without MGW* |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | **Following command** has to **be typed**:<br>✓ Set-CsMediaConfiguration **–EncryptionLevel SupportEncryption** |
| **Specific Normalization Rule** | |
| Voice Mail Feature : | A **Normalization Rule** has to **be associated to each Site dial plan*** |

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| Menu | Value |
|---|---|
| From the Microsoft Lync Server Control Panel interface:<br>Start > All Programs > Microsoft Lync Server 2013<br>> Lync Server Control Panel<br>✓ Voice Routing > Dial Plan | *(*) to be adapted according the client architecture* |
| Call Park Feature :<br>From the Microsoft Lync Server Control Panel interface:<br>Start > All Programs > Microsoft Lync Server 2013<br>> Lync Server Control Panel<br>Voice Routing > Dial Plan | **A Normalization Rule** has to **be associated to each Site dial plan***<br><br>*(*) to be adapted according the client architecture* |
| **Music On Hold** | |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell<br>**Note:**<br>The customized MoH is played **For Softphone Devices**<br>The embedded firmware MoH is played **For Lync Phone Edition Devices** | **The global clientpolicy is used:**<br>**Following commands** have to **be typed for Softphones**<br>✓ New-CsClientPolicy –Identity **global** –EnableClientOnHold **$True** –MusicOnHoldAudioFile *<FILE PATH>*<br>**Note:**<br>No more need to associate Each user **to a specific Client Policy,** check only while user creation that client policy field is set to **Automatic** |
| **Unified Messaging on Microsoft Exchange Server 2013** | |
| From the Exchange Server Administration Url:<br>https://exchangeserverIPaddress/ecp<br>logon using administrator credential<br>✓ Select **Unified Messaging**<br>✓ Double click on **UM DialPlan** then click on **configure** | On the General tab, **VoIP security** has to be set to **Secured** |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | **Following command** has to **be typed**<br>Set-UMservice –Identity *<ExchangeServer>* **–UMStartUpMode TLS** |
| From the Exchange Server Administration Url:<br>https://exchangeserverIPaddress/ecp<br>logon using administrator credential<br>✓ Select **Unified Messaging**<br>✓ Double click on **UM DialPlan** then click on **configure** | On the Settings tab**, Audio codec** has to be set to **GSM** |
| From the Exchange Server Administration Url:<br>https://exchangeserverIPaddress/ecp<br>logon using administrator credential<br>✓ Select **Unified Messaging**<br>✓ Double click on **UM DialPlan** then click on **configure** | On the Outlook Voice Access, A **Subscriber Access Number** (E164 telephone number format) has to **be added** |
| From the Exchange UM server (Config file):<br>✓ C:\Program Files\Microsoft\Exchange | <add key="MinimumRtpPort" value="**49152**" /> |

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| Menu | Value |
|---|---|
| Server\V15\Bin\MSExchangeUM | <add key="MaximumRtpPort" value="**57500**" /> |
| From the Exchange UM server (Local Group Policy Editor ):<br>✓ Start > Run… > gpedit.msc | **Audio Policy-based QoS** is configured<br>Source port: **49152:57500**<br>Protocol: **TCP and UDP**<br>DSCP: **46** |
| From the Front End Server:<br>✓ C:\Program Files\Common Files\Microosft Lync Server 2013\Support\OcsUmUtil.exe<br>✓ On the OcsUmUtil tool:<br>▪ Click **Load Data**<br>▪ Double click on **contacts** | Select **Use this pilot number from Exchange UM** has to match the subscriber access number (E.164 telephone number format) |
| Analog Devices Configuration | |
| *From the Microsoft Server 2013 Control Panel and Management Shell* | |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Voice Policy | An **User policy** has to **be created for each site with Analog Devices**<br>**Enable call park** has to **be checked**<br>**Enable PSTN reroute** has to **be unchecked**<br>An **Existent PSTN Usage** has to **be associated** by selecting it |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | **Following command** has to **be typed for each Analog Device** :<br>✓ New-CsAnalogDevice *"LineURI"* –DisplayName *"DisplayName"* –RegistrarPool *"RegistrarPool"* **–AnalogFax $False** –Gateway *"Gateway"* –OU *"OU"* |
| From the Microsoft Lync Server Management Shell interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | **Following command** has to **be typed for each Analog Device** :<br>✓ Set-CsAnalogDevice -Identity *"Identity"* –DisplayNumber *"DisplayNumber"*<br>✓ Set-CsAnalogDevice -Identity *"Identity"* –LineURI *"LineURI"*<br>✓ Grant-CsVoicePolicy -Identity *"Identity"* –PolicyName *"PolicyName"* |
| *From the Sonus (NET) (UX 1000/2000 SBA)* | |
| From the UX Web User interface:<br>✓ Settings Tab > Media > Media List | A **Media List** has to **be created:**<br><br>Media List for Analog Devices:<br>**Media Profiles List** has to match the **Voice Codec Profile G711 A-Law**<br>➢ *Digit Relay*<br>**Digit (DTMF) Relay Type** has to be set to **RFC 2833**<br>**Digit Relay Payload Type** has to be set to **101** |
| From the UX Web User interface:<br>✓ Settings Tab > CAS > CAS Signaling Profiles | A **FXS CAS Signaling Profiles** has to **be created** |
| From the UX Web User interface:<br>✓ Settings Tab > Signaling Groups | A **CAS Signaling Group** has to **be created:**<br><br>CAS Signaling Group for Analog Devices connectivity:<br>➢ *CAS Protocol*<br>**CAS Signaling Profile** has to match the **CAS Signaling Profile** for Analog Devices<br>➢ *Channels and Routing*<br>**Channel Hunting** has to be set to **Own Number** |

| Menu | Value |
|---|---|
| | **Tone Table** has to match the **Analog Device Tone Table**<br>**Call Routing Table** has to match the **Analog Device Call Routing Table**** **for routing calls received from Analog Devices**<br>➢ *Assigned Channels*<br>**Channel Phone Number** has to match the **Analog Device phone number**<br><br>*(**) Please note that Call Routing Table must be added later (after specific Call Routing Tables configuration)* |
| From the UX Web User interface:<br>✓ Settings Tab > Transformation | A **Transformation Table** has to **be created**:<br><br>Transformation Table for Lync to Analog Device calls:<br>➢ *Input Field*<br>**Value** has to match **the Analog Device telephone number E.164 format**<br>➢ *Output Field*<br>**Value** has to be set to **\1** |
| From the UX Web User interface:<br>✓ Settings Tab > Call Routing Table | A **Call Routing Table** has to **be created** for calls received from Lync (if it doesn't exist) or additionals **Call Routing Entries** have to **be created** in the Call Routing Table for calls received from Lync (if it exists)<br><br>Call Routing Entry for Lync to Analog Device calls:<br>➢ *Route Details*<br>**Number/Name Transformation Table** has to match the **Transformation Table for Lync to Analog Device calls**<br>➢ *Destination Information*<br>**Destination Signaling Groups** has to match the **Signaling Group for Analog Device connectivity**<br>➢ *Media*<br>**Media List** has to match the **Media List for Analog Device**<br><br>A **Call Routing Table** has to **be created** for calls received from the Analog Devices<br><br>Call Routing Entry Tenor to Lync calls:<br>➢ *Route Details*<br>**Number/Name Transformation Table** has to match the **Transformation Table used to send a phone number without modification**<br>➢ *Destination Information*<br>**Destination Signaling Groups** has to match the **Signaling Group for Lync connectivity**<br>➢ *Media*<br>**Media List** has to match the **Media List for Analog Device**<br><br>*(**) Please note that Call Routing Table must be added to CAS Signaling Groups configuration* |
| *From the AudioCodes (Mediant 800/1000 SBA)* | |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > TDM submenu > Select TDM Bus Settings | **PCM Law Select** has to be set to **A-Law**<br>**TDM Bus Clock Source** has to be set to **Network** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > | **CAS Transport Type** has to be set to **CASRFC2833Relay** |

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| Menu | Value |
|---|---|
| Media submenu > Select Voice Settings<br>From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > Media submenu > Select Analog Settings | Check that **Analog Settings** are **filled** with default value |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > Coders and Profiles submenu > Select Analog Coders | **Coder Name** has to be set to **G711 A-Law**<br>**Packetization Time** has to be set to**20ms**<br>**Payload Type** has to be set to **8** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Trunk Group > Select Trunk Group<br><br>From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Trunk Group > Select Trunk Group Settings | A **Trunk Group** has to be created with the following parameters:<br>**Module** has to be set to **Module 2 FXS**<br>**Channels** has to be set to the **Analog Device** port on the gateway<br>**Phone Number** has to match the **Analog Device** phone number<br>**Trunk Group ID** has to match the **Analog Device** Trunk Group ID<br>**Tel Profile ID** has to match the **Tel Profile ID** if configured else the **default profile 0** has to be associated<br><br>**Trunk Group ID** has to match the **Analog Device** Trunk Group ID<br>**Channel Select Mode** has to be set to **By Dest Phone Number** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Manipulation > Select Dest Number IP -> Tel<br>From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Manipulation > Select Dest Number Tel -> IP | **Destination Prefix** has to match the **Analog Device** Phone Number as declared on the **Trunk Group Table**<br><br>**Source Trunk Group** has to match the **Analog Device Trunk Group** already created<br>**Prefix to add** has to match a rule manipulation in order to has a **E.164 format number** to send to Lync Server |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Routing > Select Tel to IP Routing<br><br>From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Routing > Select IP to Tel Routing | **Tel to IP Routing Mode** has to be set to **Route Calls after manipulation**<br>**Src IP Group ID** has to be set to **-1**<br>**Src Trunk Group ID** has to match the **Analog Device Group ID**<br>**Dest IP Group ID** has to match the **Lync Server Group ID**<br><br>**IP toTel Routing Mode** has to be set to **Route Calls before manipulation**<br>**Dest Phone Prefix** has to match the **Analog Device** phone number<br>**Trunk Group ID** has to match the **Analog Device Trunk Group ID**<br>**IP Profile ID** has to match the **Tel Profile ID** if configured else the **default profile 0** has to be associated |
| E1/T1 Access Configuration | |

| Menu | Value |
|---|---|
| *From the Sonus (NET) (UX 1000/2000 SBA) with FXS ports* | |
| From the UX Web User interface:<br>✓ Settings Tab > Signaling Groups | An **ISDN Signaling Group** has to **be created**:<br><br>ISDN Signaling Group for E1/T1 connectivity:<br>➢ *Port and Protocol*<br>**Port Name** has to be **selected**<br>**Switch Variant** has to be set to **Euro ISDN**<br>➢ *Channels and Routing*<br>**Tone Table** has to match the **Tone Table if** configured else the **Default Tone Table** has to be selected<br>**Call Routing Table** has to match the **E1/T1 Call Routing Table** for routing calls received from E1/T1 access**<br><br>*(**) Please note that Call Routing Table must be added later (after specific Call Routing Tables configuration)* |
| From the UX Web User interface:<br>✓ Settings Tab > Transformation<br><br><br><br><br><br><br><br><br><br><br><br><br><br><br><br><br><br>From the UX Web User interface:<br>✓ Settings Tab > Transformation | Transformation Table for T2 to Lync calls<br>A **Transformation Table** has to **be created**:<br><br>Transformation Entry for T2 to Lync calls (Called):<br>➢ *Input Field*<br>**Type** has to be set to **Called Address/Number**<br>**Value** has to match the **T2 number**<br>➢ *Output Field*<br>**Type** has to be set to **Called Address/Number**<br>**Value** has to match the **E.164 Lync number**<br><br>Transformation Entry for T2 to Lync calls (Calling):<br>➢ *Input Field*<br>**Type** has to be set to **Calling Address/Number**<br>**Value** has to be **filled**<br>➢ *Output Field*<br>**Type** has to be set to **Calling Address/Number**<br>**Value** has to be **filled**<br><br>Transformation Table for Lync to T2 calls<br><br>A **Transformation Table** has to **be created**:<br><br>Transformation Entry for Lync to T2 calls (Called):<br>➢ *Input Field*<br>**Type** has to be set to **Called Address/Number**<br>**Value** has to be **filled**<br>➢ *Output Field*<br>**Type** has to be set to **Called Address/Number**<br>**Value** has to be **filled**<br><br>Transformation Entry for Lync to T2 calls (Calling):<br>➢ *Input Field*<br>**Type** has to be set to **Calling Address/Number**<br>**Value** has to be **filled** |

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| Menu | Value |
|---|---|
| | ➢ *Output Field*<br>**Type** has to be set to **Calling Address/Number**<br>**Value** has to be **filled** |
| From the UX Web User interface:<br>✓ Settings Tab > Call Routing Table | Call Routing Table for Lync to T2 calls<br><br>A **Call Routing Table** has to **be created** for calls received from Lync (if it doesn't exist) or an additional **Call Routing Entry** has to **be created** in the Call Routing Table for calls received from Lync (if it exists)<br><br>Call Routing Entry for Lync to T2 calls:<br>➢ *Route Details*<br>**Number/Name Transformation Table** has to match the **Transformation Table for Lync to T2 calls**<br>➢ *Destination Information*<br>**Destination Signaling Groups** has to match the **Signaling Group for E1/T1 connectivity**<br>➢ *Media*<br>**Media List** has to match the **Media List without crypto**<br><br>Call Routing Table for T2 to Lync calls<br><br>A **Call Routing Table** has to **be created** for calls received from E1/T1 access<br><br>Call Routing Entry for T2 to Lync calls:<br>➢ *Route Details*<br>**Number/Name Transformation Table** has to match the **Transformation Table T2 to Lync calls**<br>➢ *Destination Information*<br>**Destination Signaling Groups** has to match the **Signaling Group for Lync connectivity**<br>➢ *Media*<br>**Media List** has to match the **Media List without crypto**<br><br>*(**) Please note that Call Routing Table must be added to ISDN/SIP Signaling Groups configuration* |
| *From AudioCodes Mediant (800/ 1000 SBA)* | |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > PSTN submenu > Select Trunk Settings | **Protocol Type** has to be set to **E1 Euro ISDN**<br>**Line Code** has to be set to**HDB3**<br>**Framing Method** has to be set to **E1 FRAMING MFF CRC4 EXT** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Trunk Group > Select Trunk Group | A **Trunk Group** has to be created with the following parameters:<br>**Module** has to be set to **Module 1 PRI**<br>**Channels** has to be set to **T2 line number of channels**<br>**Phone Number** has to match the **T2** phone number<br>**Trunk Group ID** has to match the **T2** Trunk Group ID<br>**Tel Profile ID** has to match the **Tel Profile ID** if configured else the **default profile 0** has to be associated |

| Menu | Value |
|---|---|
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Trunk Group > Select Trunk Group Settings | **Trunk Group ID** has to match the **T2 Trunk Group ID**<br>**Channel Select Mode** has to be set to **Cyclic Ascending** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > Control Network submenu > Select Proxy Set Table | A **Proxy Set Table** has to be created with the following parameters:<br>**Proxy Set ID** has to be **filled**<br>**Proxy Address** has to match the **SBA FQDN**<br>**Transport Type** has to be set to **TLS**<br>**Enable Proxy Keep Alive** has to be set to **Using Options** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > Control Network submenu > Select IP Group Table | An **IP Group Table** has to be created with the following parameters:<br>**Index** has to be **filled**<br>**Type** has to be set to **Server**<br>**Proxy Set ID** has to match the **SBA proxy Set ID** already created |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Manipulation > Select Dest Number IP -> Tel<br><br>From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Manipulation > Select Dest Number Tel -> IP | **Destination Prefix** has to be filled with the **prefix** of the received number<br>**Source IP Address** has to match the **SBA IP Address**<br>**Stripped Digits from Left** has to be **filled**<br>**Prefix to Add** has to be **filled**<br><br>**Source Trunk Group** has to match the **T2 Trunk Group** already created<br>**Destination Prefix** has to match the **T2 Line number**<br>**Stripped Digits from Left** has to be **filled**<br>**Prefix to add** has to match the corresponding Lync device on **E.164 format number** |
| From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Routing > Select Tel to IP Routing<br><br>From the AudioCodes Web User interface:<br>✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Routing > Select IP to Tel Routing | **Tel to IP Routing Mode** has to be set to **Route Calls after manipulation**<br>**Src IP Group ID** has to be set to **-1**<br>**Src Trunk Group ID** has to match the **T2 Group ID**<br><br>**IP toTel Routing Mode** has to be set to **Route Calls before manipulation**<br>**Source IP Address** has to match the **Gateway IP Address**<br>**Trunk Group ID** has to match the **T2 Trunk Group ID**<br>**IP Profile ID** has to match the **Tel Profile ID** if configured else the **default profile 0** has to be associated |
| **Dial-in Conferencing feature** | |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Routing > Dial Plan | A **Dial-in conferencing region** has to **be added** (associated to Dial-in Access Number) |
| **Call Back feature** | |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control | A specific **translation Rule** has to **be associated to each Site trunk**<br><br>*(*) to be adapted to the client architecture*<br>*(**) first priority before translation rule removing the « + » digit* |

| Menu | Value |
|---|---|
| Panel<br>✓ Voice Routing > Trunk Configuration | |
| **Call Park feature** | |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>✓ Voice Features | A Number range has to be created **for each Site**<br><br>*(*) to be adapted to the client architecture* |
| **CALL ADMISSION CONTROL** | |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>Network Configuration > Global | Edit Global Setting –Global<br>Check **Enable call admission control** |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>Network Configuration > Bandwidth Policy | Create Bandwidth Policy for **CAC "from site to WAN"**<br>**New** *"name"*<br>Audio limit: **according to site sizing**<br>Audio session limit: **100**<br>Create Bandwidth Policy for **CAC "from Edge to WAN"**<br>**New** *"name"*<br>Audio limit: **according to site sizing**<br>Audio session limit: **9999999999**<br>Create Bandwidth Policy for **CAC "from site to SIP Trunk"**<br>**New** *"name"*<br>Audio limit: **according to site sizing**<br>Audio session limit: **97**<br>Create Bandwidth Policy for **CAC "0"**<br>**New** *"name"*<br>Audio limit: **0**<br>Audio session limit: **40** |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>Network Configuration > Region | Create WAN Region<br>**New** *"name"*<br>**Associate site name**<br>Uncheck **Enable audio alternate path** (recommended)<br>Check or Uncheck **Enable video alternate path** to your convenience |

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| Menu | Value |
|---|---|
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>Network Configuration > Site | Create **Site for users** and associate a **Bandwidth policy** between this **Site** and the **Region**<br>**New** *"name"*<br>Associate **Region**<br>Associate **Bandwidth Policy** for **CAC "from site to WAN"**<br><br>Create **Site for edge** and associate a **Bandwidth policy** between this **Site** and the **Region**<br>**New** *"name"*<br>Associate **Region**<br>Associate **Bandwidth Policy** for **CAC "from Edge to WAN"**<br><br>Create **Site for aSBC** and associate a **Bandwidth policy** between this **Site** and the **Region**<br>**New** *"name"*<br>Associate **Region**<br>Associate **Bandwidth Policy** for **CAC "0"** |
| From the Microsoft Lync Server Management Shell interface:<br>Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | Creation of Bandwidth Policy for intersite links<br>**New-CsNetworkInterSitePolicy –Identity** *"name of the intersitelink"* **-BWPolicyProfileID** *"name of the policy for CAC from site to SIP Trunk"* **-NetworkSiteID1** *"name of the site for user"* **-NetworkSiteID2** *"name of the sitefor the SBC"* |
| From the Microsoft Lync Server Control Panel interface:<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel<br>Network Configuration > Subnet | Create subnet for each site<br>**New**<br>Add **subnet ID**<br>Add **mask**<br>Associate with **Network site ID** |
| Quality of Service | |
| From the Microsoft Lync Management Shell interface::<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | Enable client media port range:<br>**Set-CsConferencingConfiguration –ClientMediaPortRangeEnabled $true –ClientMediaPort 50000 –ClientAudioPort 50060 –ClientVideoPort 57600 –ClientAppSharingPort 32800** |
| From the Microsoft Lync Management Shell interface::<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | Configure ApplicationSharing port range on Lync application servers:<br>**Set-CsApplicationServer ApplicationServer:**<serverFQDN> **-AppSharingPortStart 32768 –AppSharingPortCount 16383** |
| From the Microsoft Lync Management Shell interface::<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | Configure ApplicationSharing port range on Lync Conferencing servers:<br>**Set-CsApplicationServer ConferencingServer:**<serverFQDN> **-AppSharingPortStart 32768 –AppSharingPortCount 16383** |
| Configuration requirements (warnings) | |
| Configuring Clients ports range for LPE and SoftPhone | |
| From the Microsoft Lync Management Shell interface::<br>✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell | Enable client media port range:<br>**Set-CsConferencingConfiguration –ClientMediaPortRangeEnabled $true –ClientAudioPort 50060 –ClientAudioPortRange 48** |

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| Menu | Value |
|---|---|
| **Configuring Clients ports range for VVX** | |
| ✓ Using VVX Web UI | Navigate through the VVX Web Interface: http:<VVX_IP_Address><br><br>Go to Settings tab > Network menu > RTP<br><br>Configure the Port Range Start to: 50060 |
| ✓ Using VVX configuration file (.cfg) | Configure the following line in the VVX configuration file :<br>`tcpIpApp.port.rtp.mediaPortRangeStart="50060"`<br><br>Import the new configuration file to the VVX using the WebUI or through the IIS server |
| **Others Devices** | |
| Check that the audio range port respect the OBS recommendations | The default audio range is: 50060-50107. |

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# 5 Skype for Business 2015 Configuration Checklist

| Menu | Value |
|---|---|
| **Skype for Business Configuration (Topology Builder)** | |
| On the Topology builder interface:<br>✓ Central Site > skype for business 2015 > **Mediation Pools**, right click and Edit properties | **Enable TCP** port has to be **checked**<br>**Listening port** has to be set to **5060** for each Mediation Server in skype for Business topology |
| On the Topology builder interface:<br>✓ Central Site > Skype for Business 2015 > Shared components > Trunks, right click edit properties | **FQDN** of **nominal aSBC** for BT/BTIP traffic<br><br>Specify **nominal aSBC BT/BTIP trunk name**<br>Listening port for IP/PSTN gateway**: 5060**<br>SIP Transport protocol: **TCP**<br>Associated Mediation Server: **Mediation Server FQDN**<br>Associated Mediation Server port: **5060** |
| On the Topology builder interface:<br>✓ Central Site > Skype for Business 2015 > Shared components > Trunks, right click edit properties | **FQDN** of **backup aSBC** for BT/BTIP traffic<br><br>Specify **backup aSBC BT/BTIP trunk name**<br>Listening port for IP/PSTN gateway**: 5060**<br>SIP Transport protocol: **TCP**<br>Associated Mediation Server: **Mediation Server FQDN**<br>Associated Mediation Server port: **5060** |
| **Skype for Business Configuration (Control Panel)** | |
| **Dial Plan**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Dial Plan | Type: **Dial Plan** type<br>Name: **Dial Plan** name |
| **Voice Policy**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | Name: **Voice Policy** name<br>Enable call park: **Checked**<br>Enable PSTN reroute: **Unchecked** |
| **PSTN usage**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | New PSTN Usage record<br>Name: **BT/BTIP PSTN Usage name** |
| **Routes (aSBC nominal route)**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | Edit PSTN Usage record<br>Associated routes → New<br>Name: **aSBC nominal Route name**<br>Associated Trunks → Add<br>**Select** corresponding **aSBC nominal Trunk** from drop down list |
| **Routes (aSBC backup route)**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | Edit PSTN Usage record<br>Associated routes → New<br>Name: **aSBC backup Route name**<br>Associated Trunks → Add<br>**Select** corresponding **aSBC backup Trunk** from drop down list |
| **Trunk configuration** | New |

| Menu | Value |
|---|---|
| On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Trunk configuration | Name: **BT/BTIP Trunk name**<br>Encryption support level : **Optional**<br>Refer support : **None**<br>Enable forward call History : **Checked** |
| **Trunk configuration (SFB PowerShell)**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPActiveCalls $False***<br>✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPCallsOnHold $False*** | **-Site:** The name of the site |

Configuration Checklist in case of Ribbon SBC 1000/2000 Gateway:

This configuration checklist will follow this color convention:

- **Green: in case of RS SBA**
- **Blue: in case of HQ with GW aboard**

| Skype for Business– RS SBA or HQ with GW aboard - Trunk SIP on Ribbon SBC BT/BTIP configuration | |
|---|---|
| **PSTN usage**<br>On the Skype for Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | New Ribbon SBC BT/BTIP PSTN Usage record<br>Name: Ribbon SBC **BT/BTIP PSTN Usage name** |
| **Route (Ribbon SBC BT/BTIP)**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | Edit PSTN Usage record<br>Associated routes → New<br>Name: **Ribbon SBC for BT/BTIP route name**<br>Associated Trunks → Add<br>**Select** corresponding **Ribbon SBC Trunk** from drop down list |
| **Trunk configuration**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Trunk configuration | New<br>Name: **Ribbon SBC for BT/BTIP Trunk name**<br>Encryption support level : **Optional**<br>Refer support : **None**<br>Enable forward call History : **Checked** |
| **Trunk configuration (SFB PowerShell)**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPActiveCalls $False***<br>✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPCallsOnHold $False*** | **-Site:** The name of the remote site |
| **Ribbon SBC BT/BTIP configuration** | |
| **SIP Profile** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Profile > Default SIP Profile | Session Timer:<br>Session Timer: **Disabled**<br>Header Customization:<br>UA Header: **Ribbon SBC** |

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| Menu | Value |
|---|---|
| | Calling Info Source: **RFC Standard**<br>Options Tags:<br>100rel: **Supported**<br>Update: **Supported**<br>SDP Customization:<br>Send Number of Channels: **True**<br>Connection Info In Media Section: **True**<br>Digit Transmission Preference: **RFC 2833/Voice** |
| **Media** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Media > Media System Configuration | Port Range:<br>Start Port: **16384**<br>Number of Port pairs: **600**<br>Echo Canceller Type Option: **Standard**<br>Echo Cancel NLP Option: **Mild**<br>Send STUN Packets: **Enabled**<br>Music On Hold:<br>Music on Hold Source: **File** |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Media > Media Profiles | Default G711a:<br>Codec: **G711 A-law**<br>Payload Size: **20 ms**<br>Default G711µ:<br>Codec: **G711 µ-law**<br>Payload Size: **20 ms** |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Media > Media List | Default Media List:<br>Media Profiles List: **G711a**<br>**G711µ**<br>Crypto Profile ID: **None**<br>Media DSCP: **46**<br>RTCP Mode: **RTCP**<br>Dead Call Detection: **Disabled**<br>Silence Suppression: **Disabled** |
| **Secondary interface (only for RS SBA)** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Node Interfaces > Logical Interfaces > Ethernet 1 IP | Configure Secondary Interface: **Enabled**<br>Secondary Address: **IP address of the secondary interface of the Ribbon gateway (dedicated for BT/BTIP traffic)**<br>Secondary Mask: **Mask corresponding to secondary interface subnet** |
| **From/To SFB <-> Offnet routing BT/BTIP traffic** | |
| **SIP Server Table** | |
| **From/To SBA –BT/BTIP or From/To MS Pool –BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Server Tables > Create SIP Server | Host: **SBA or MS Pool IP address**<br>Port: **5060**<br>Protocol: TCP<br>Monitor: **SIP Options** |
| **From/To BT/BTIP-SBA or From/To MS Pool –BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Server Tables > Create SIP Server | 1st Entry: ACME aSBC nominal<br>Host: **ACME aSBC nominal IP address**<br>Port: **5060**<br>Protocol: TCP |

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| Menu | Value |
|---|---|
| | Monitor: SIP Options<br>2nd Entry: ACME aSBC backup<br>Host: ACME aSBC backup IP address<br>Port: 5060<br>Protocol: TCP<br>Monitor: SIP Options |
| **Transformation Rules** | |
| **SBA to BT/BTIP or MS Pool to BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **BT/BTIP to SBA or BT/BTIP to SBA**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: must normalize received number on Skype for Business E.164 number format<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **Call Routing Tables** | |
| **From SBA or From MS Pool**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | **SBA to BT/TIP** or **MS Pool to BT/TIP** entry:<br>Description: **SBA to BT/BTIP or MS pool to BT/BTIP**<br>Route Priority: **1**<br>Number/Name Transformation Table: SBA to BT/BTIP **or** MS Pool to BT/BTIP<br>Destination Signalling Group: (SIP) From/To BT/TIP-SBA **or** From/To BT/TIP-SBA<br>Media Transcoding: Enabled (If licenced) |
| **From BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | **BT/TIP to SBA** or **BT/TIP to MS Pool** entry:<br>Description: **BT/BTIP to SBA or BT/BTIP to MS Pool**<br>Route Priority: **1**<br>Number/Name Transformation Table: BT/BTIP to SBA **or** BT/BTIP to MS Pool<br>Destination Signalling Group: **(SIP) From/To SBA-BT/BTIP or From/To MS Pool-** |

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| Menu | Value |
|---|---|
| | **BT/BTIP**<br>Media Transcoding: **Enabled** (If licenced) |
| **Signaling Groups** | |
| **(SIP) From/To SBA – BT/BTIP or From/To MS Pool – BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **SIP From/To SBA – BT/BTIP or From/To MS Pool – BT/BTIP**<br>Call Routing Table: **From SBA or From MS Pool**<br>SIP Server Table: From/To SBA –BT/BTIP or MS Pool –BT/BTIP<br>Signalling/Media Source IP :Ribbon BT/BTIP interface IP address<br>Listen Ports:5060 /TCP<br>Federated IP/FQDN: SBA or MS Pool FQDN |
| **(SIP) From/To BT/BTIP-SBA or From/To BT/BTIP-MS Pool**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **SIP From/To BT/BTIP-SBA or From/To BT/BTIP-MS Pool**<br>Call Routing Table: **From BT/BTIP**<br>SIP Server Table: From/To BT/BTIP -SBA or From/To BT/BTIP-MS Pool<br>Signalling/Media Source IP: Ribbon BT/BTIP interface IP address<br>Listen Ports:5060 /TCP<br>Federated IP/FQDN: ACME aSBC nominal IP address<br>ACME aSBC backup IP address |
| | |
| **From/To SFB <-> Offnet routing E1/T1 traffic (only for RS SBA)** | |
| **System Companding Law** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >System > System companding law | Companding law: **A-Law** |
| **SIP Server Table** | |
| **From/To SBA –PSTN**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Server Tables > Create SIP Server | Host: **SBA IP**<br>Port: example **5060 (must be the same as defined on Skype for Business topology builder)**<br>Protocol: TCP<br>Monitor: **SIP Options**<br>Note:<br>If using same protocol and port as BT/BTIP the same SIP Server table can be used |
| **Transformation Rules** | |
| **SBA to PSTN**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Called Address/Number |

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| Menu | Value |
|---|---|
| | Output Field Value: depend on transformation need |
| **PSTN to SBA**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: must normalize received number on Skype for Business E.164 number format<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **Call Routing Tables** | |
| **From SBA**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | SBA to PSTN entry:<br>Description: **SBA to PSTN**<br>Route Priority: **1**<br>Number/Name Transformation Table: SBA to PSTN<br>Destination Signalling Group: (ISDN) From/To PSTN-SBA<br>Media Transcoding: Enabled (If licenced) |
| **From PSTN**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | PSTN to SBA entry:<br>Description: **PSTN to SBA**<br>Route Priority: **1**<br>Number/Name Transformation Table: PSTN to SBA<br>Destination Signalling Group: **(SIP) From/To SBA-PSTN**<br>Media Transcoding: **Enabled** (If licenced) |
| **Signaling Groups** | |
| **(SIP) From/To SBA – PSTN**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **SIP From/To SBA – PSTN**<br>Call Routing Table: **From SBA**<br>SIP Server Table: From/To SBA –PSTN<br>Signalling/Media Source IP :Ribbon E1/analog interface IP address<br>Listen Ports:5060 /TCP<br>Federated IP/FQDN: SBA IP address |
| **(ISDN) PSTN**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > Signaling Group > ISDN Signaling Group | Description: **ISDN PSTN**<br>Switch variant: **Euro ISDN**<br>Call Routing Table: **From PSTN** |
| **From/To SFB <-> Offnet routing Analog Devices traffic** | |

| Menu | Value |
|---|---|
| **SIP Server Table** | |
| **From/To SBA –Analog Device**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Server Tables > Create SIP Server | Host: **SBA FQDN/IP address**<br>Port: example **5060 (must be the same as defined on Skype for Business topology builder)**<br>Protocol: TCP<br>Monitor: **SIP Options**<br><br>If using same protocol and port as BT/BTIP the same SIP Server table can be used ( no need to create a new SIP Server table) |
| **Transformation Rules** | |
| **SBA to Analog**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **Analog Device to SBA**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: must normalize received number on Skype for Business E.164 number format<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **Call Routing Tables** | |
| **From SBA**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | SBA to analog device entry:<br>Description: **SBA to Analog Device**<br>Route Priority: **1**<br>Number/Name Transformation Table: SBA to PSTN<br>Destination Signalling Group: (CAS) Analog Device<br>Media Transcoding: Enabled (If licenced) |
| **From Analog Device**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | Analog Device to SBA entry:<br>Description: **Analog Device to SBA**<br>Route Priority: **1**<br>Number/Name Transformation Table: Analog |

| Menu | Value |
|---|---|
| | Device to SBA<br>Destination Signalling Group: **(SIP) From/To SBA-Analog Device**<br>Media Transcoding: **Enabled** (If licenced) |
| **Signaling Groups** | |
| **(SIP) From/To SBA – Analog Device**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **SIP From/To SBA – Analog Device**<br>Call Routing Table: **From SBA**<br>SIP Server Table: From/To SBA –Analog Device<br>Signalling/Media Source IP :Ribbon E1/analog interface IP address<br>Listen Ports:5060 /TCP<br>Federated IP/FQDN: SBA IP address |
| **(CAS) Analog**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **CAS Analog**<br>CAS Signalling Profile: **CAS Analog**<br>Call Routing Table: **Analog to SBA**<br>Assigned Channels: **Analog Devices information** |
| **Skype for Business– RS GW BT/BTIP configuration** | |
| **PSTN usage**<br>On the Skype for Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | New Ribbon SBC BT/BTIP PSTN Usage record<br>Name: Ribbon Gateway **BT/BTIP PSTN Usage name** |
| **Route (Ribbon SBC BT/BTIP)**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Voice Policy | Edit PSTN Usage record<br>Associated routes → New<br>Name: **BT/BTIP Ribbon GW route name**<br>Associated Trunks → Add<br>**Select** corresponding **Ribbon GW Trunk** from drop down list |
| **Trunk configuration**<br>On the Skype for Business Server Control Panel Interface:<br>✓ Voice Routing > Trunk configuration | New<br>Name: **Ribbon SBC for BT/BTIP Trunk name**<br>Encryption support level : **Optional**<br>Refer support : **None**<br>Enable forward call History : **Checked**<br>Enable media bypass : **Checked** |
| **Trunk configuration (SFB PowerShell)**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPActiveCalls $False***<br>✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPCallsOnHold $False*** | **-Site:** The name of the site |
| **Ribbon GW BT/BTIP configuration** | |
| **SIP Profile** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Profile > Default SIP Profile | Session Timer:<br>Session Timer: **Disabled**<br>Header Customization:<br>UA Header: **Ribbon SBC**<br>Calling Info Source: **RFC Standard** |

| Menu | Value |
|---|---|
| | Options Tags:<br>100rel: **Supported**<br>Update: **Supported**<br>SDP Customization:<br>Send Number of Channels: **True**<br>Connection Info In Media Section: **True**<br>Digit Transmission Preference: **RFC 2833/Voice** |
| **Media** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Media > Media System Configuration | Port Range:<br>Start Port: **16384**<br>Number of Port pairs: **600**<br>Echo Canceller Type Option: **Standard**<br>Echo Cancel NLP Option: **Mild**<br>Send STUN Packets: **Enabled**<br>Music On Hold:<br>Music on Hold Source: **File** |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Media > Media Profiles | Default G711a:<br>Codec: **G711 A-law**<br>Payload Size: **20 ms**<br>Default G711µ:<br>Codec: **G711 µ-law**<br>Payload Size: **20 ms** |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Media > Media List | Default Media List:<br>Media Profiles List: **G711a**<br>**G711µ**<br>Crypto Profile ID: **None**<br>Media DSCP: **46**<br>RTCP Mode: **RTCP**<br>Dead Call Detection: **Disabled**<br>Silence Suppression: **Disabled** |
| **TLS Profile** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Security > TLS Profiles | Create TLS Profile:<br>TLS Protocol: **TLS 1.2 Only**<br>Mutual Authentication: **Enabled**<br>Allow Weak Cipher: **Disable**<br>Handshake Inactivity Timeout: **10**<br>The Client Cipher List is **automatically updated** to display only the ciphers supported for the selected TLS version<br>Validate Server FQDN: **Disabled**<br>Validate Client FQDN: **Disabled** |
| **Secondary interface** | |
| On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Node Interfaces > Logical Interfaces > Ethernet 1 IP | Configure Secondary Interface: **Disabled**<br>Primary address dedicated for BT/BTIP traffic |
| **From/To SFB <-> Offnet routing BT/BTIP traffic** | |
| **SIP Server Table** | |

| Menu | Value |
|---|---|
| **From/To MS Pool –BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Server Tables > Create SIP Server | Host: **MS Pools FQDN/IP address**<br>Port: **5067**<br>Protocol: TLS<br>TLS Profile: Select the TLS Profile created above<br>Monitor: **SIP Options** |
| **From/To BT/BTIP-MS Pool**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >SIP > SIP Server Tables > Create SIP Server | 1st Entry: ACME aSBC nominal<br>Host: **ACME aSBC nominal IP address**<br>Port: **5060**<br>Protocol: TCP<br>Monitor: SIP Options<br>2nd Entry: ACME aSBC backup<br>Host: ACME aSBC backup IP address<br>Port: 5060<br>Protocol: TCP<br>Monitor: SIP Options |
| **Transformation Rules** | |
| **MS Pool to BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **BT/BTIP to MS Pool**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Transformation > New Transformation Table > New Transformation Entry | Calling Entry:<br>Input Field Type: **Calling Address/Number**<br>Input Field Value: depend on transformation need<br>Output Field Type: Calling Address/Number<br>Output Field Value: depend on transformation need<br>Called Entry:<br>Input Field Type: **Called Address/Number**<br>Input Field Value: must normalize received number on Skype for Business E.164 number format<br>Output Field Type: Called Address/Number<br>Output Field Value: depend on transformation need |
| **Call Routing Tables** | |
| **From MS Pool**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | MS Pool to BT/TIP entry:<br>Description: **MS Pool to BT/BTIP**<br>Route Priority: **1**<br>Number/Name Transformation Table: MS Pool to BT/BTIP<br>Destination Signalling Group: (SIP) From/To |

| Menu | Value |
|---|---|
| | BT/TIP-MS Pool<br>Media Transcoding: Enabled (If licenced)<br>Media List: Select the Media List created above |
| **From BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Call Routing Table > Create | BT/TIP to MS Pool entry:<br>Description: **BT/BTIP to MS Pool**<br>Route Priority: **1**<br>Number/Name Transformation Table: BT/BTIP to MS Pool<br>Destination Signalling Group: **(SIP) From/To MS Pool-BT/BTIP**<br>Media Transcoding: Enabled (If licenced)<br>Media List: Select the **Media List created** above |
| **Signaling Groups** | |
| **(SIP) From/To MS Pool – BT/BTIP**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **SIP From/To MS Pool – BT/BTIP**<br>Call Routing Table: **From MS Pool**<br>No. of Channels: **60 (Default)**<br>SIP Server Table: From/To MS Pool –BT/BTIP<br>Signalling/Media Source IP :Ribbon BT/BTIP interface IP address<br>Listen Ports:5067 /TLS<br>TLS Profile: Select the TLS Profile created above<br>Federated IP/FQDN: MS Pools IP/FQDN |
| **(SIP) From/To BT/BTIP-MS Pool**<br>On the Ribbon SBC gateway WebUi Interface:<br>✓ Settings >Signaling Group > SIP Signaling Group | Description: **SIP Froom/To BT/BTIP-MS Pool**<br>Call Routing Table: **From BT/BTIP**<br>No. of Channels: **60 (Default)**<br>SIP Server Table: From/To BT/BTIP –MS Pool<br>Signalling/Media Source IP :Ribbon BT/BTIP interface IP address<br>Listen Ports:5060 /TCP<br>Federated IP/FQDN: ACME aSBC nominal IP address<br>ACME aSBC backup IP address |

Configuration Checklist in case of AudioCodes Mediant 800/1000 E-SBC:

| Skype for Business Configuration in case of RS-GW (Topology Builder) | |
|---|---|
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > **Mediation Pools**, right click and Edit properties | Listening ports **TLS: 5067 – 5067**<br>**Note:**<br>**When both VISIT and B2G offer:**<br>Listening ports TLS must be**: 5069** |
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for BT/BTIP<br>Then click Next to define **root trunk** | **FQDN** of dedicated gateway **for BT/BTIP traffic**<br>Specify **BT trunk name**<br>Listening port for IP/PSTN gateway**: 5067**<br>SIP Transport protocol: **TLS**<br>Associated Mediation Server: **Mediation Pool FQDN** |

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| Menu | Value |
|---|---|
| | Associated Mediation Server port: **5067**<br><br>**Note:**<br>**When both VISIT and B2G offer:**<br>Listening ports TLS must be**: 5069** |
| **Skype for Business Configuration in case of RS-SBA (Topology Builder)** | |
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > **Mediation Pools**, right click and Edit properties | Listening ports **TCP: 5060 – 5060** |
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for BT/BTIP<br>Then click Next to define **root trunk** | **FQDN** of dedicated gateway **for BT/BTIP traffic**<br><br>Specify **BT trunk name**<br>Listening port for IP/PSTN gateway**: 5060**<br>SIP Transport protocol: **TCP**<br>Associated Mediation Server: **SBA FQDN**<br>Associated Mediation Server port: **5060** |
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for E1/analog<br><br>**PSTN & Analog Trunk:**<br>✓ Branch Site > SfB Server > Shared Components > Trunks, right click and **New Trunk** | **FQDN** of dedicated gateway **for E1/Analog traffic**<br><br>Specify **PSTN&Analog trunk name**<br>Listening port for IP/PSTN gateway**: 5060**<br>SIP Transport protocol: **TCP**<br>Associated Mediation Server: **SBA FQDN**<br>Associated Mediation Server port: **5060** |
| **Skype for Business Configuration in case of HQ with GW aboard (Topology Builder)** | |
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > **Mediation Pools**, right click and Edit properties | Listening ports **TCP: 5060 – 5060** |
| On the Topology builder interface:<br>✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for BT/BTIP<br>Then click Next to define **root trunk** | **FQDN** of dedicated gateway **for BT/BTIP traffic**<br><br>Specify **BT trunk name**<br>Listening port for IP/PSTN gateway**: 5060**<br>SIP Transport protocol: **TCP**<br>Associated Mediation Server: **MS Pool FQDN**<br>Associated Mediation Server port: **5060** |
| **AudioCodes Mediant 800/1000 E-SBC configuration** | |
| **TLS Context** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>✓ System > TLS Context | Links Tab<br>TLS Context Certificate<br>TLS Context Trusted Certificates |
| **Media** | |
| **Voice Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>✓ Configuration >VoIP > Media > Voice Settings | Silence Suppression: **Disable**<br>DTMF Transport Type: **RFC 2833 Relay DTMF** |
| **Media Security** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode) | Media security: **Enable** |

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| Menu | Value |
|---|---|
| Configuration >VoIP > Media > Media Security | |
| **RTP / RTCP Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration >VoIP > Media > RTP / RTCP Settings | RTP Base UDP Port: **16400** |
| **Application Enabling** | |
| **Application Enabling** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration >VoIP > Application Enabling > Application Enabling | SBC Application: **Enable** |
| **Coders and Profiles** | |
| **Coders** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration >VoIP > Coders and Profiles > Coders | Coders Table<br>Coder Name : **G711A-law**<br>Packetization time : **20**<br>Rate : **64**<br>Payloed Type : **8**<br>Silence Suppression : **Disabled**<br><br>Coder Name : **G711U-law**<br>Packetization time : **20**<br>Rate : **64**<br>Payload Type : **0**<br>Silence Suppression : **Disabled** |
| **Coders Group Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration >VoIP > Coders and Profiles > Coders Group Settings | Coders Group ID<br>Coder Name : **G711A-law**<br>Packetization time : **20**<br>Rate : **64**<br>Payloed Type : **8**<br>Silence Suppression : **Disabled**<br><br>Coder Name : **G711U-law**<br>Packetization time : **20**<br>Rate : **64**<br>Payload Type : **0**<br>Silence Suppression : **Disabled** |
| **IP Profile Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration >VoIP > Coders and Profiles > IP Profile Settings | SBA or SfB IP Profile ID<br>(GW tab)<br>Early Media : **Enable**<br>Hold : **Enable**<br><br>(SBC Media tab)<br>Extension Coders : **Coders Group**<br>Allowed Audio Coders : **Coders Group** |

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| Menu | Value |
|---|---|
| | Allowed Coders Mode : **Restriction and Preference**<br><br>BTIP IP Profile ID<br>(GW tab)<br>Early Media : **Enable**<br>Hold : **Enable**<br><br>(SBC Media tab)<br>Extension Coders : **Coders Group**<br>Allowed Audio Coders : **Coders Group**<br>Allowed Coders Mode : **Restriction and Preference** |
| **VoIP Network** | |
| **Media Realm Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > Media Realm Table | Skype Media Realm (SBA or SfB)<br>Name : **MRm for Skype**<br>IPv4 Interface Name : **Mediant IPv4 Interface**<br>Port Range Start : **16900**<br>Number of Media Session Legs : **50**<br>Port Range End : **Filled automatically**<br>Default Media Realm : **Yes**<br><br>BTIP Media Realm<br>Name : **MRm for BTIP**<br>IPv4 Interface Name : **Mediant IPv4 Interface**<br>Port Range Start : **16400**<br>Number of Media Session Legs : **50**<br>Port Range End : **Filled automatically**<br>Default Media Realm : **No**<br>**This range is used to accept incoming traffic from SBC in case of BTIP incoming calls, the defined range respects the OBS infra recommandations** |
| **SRD Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > SRD Table | Name : **DefaultSRD** |
| **SIP Interface Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > SIP Interface Table | One SIP Interface Table for RS SBA<br>Name : **SIPInterface_BTIP&SBA**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface**<br>Application Type : **SBC**<br>TCP Port : **5060**<br><br>One SIP Interface Table for HQ with GW aboard<br>Name : **SIPInterface_BTIP&SBA**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface** |

| Menu | Value |
|---|---|
| | Application Type : **SBC**<br>TCP Port : **5060**<br><br>Two SIPs Interfaces Tables for RS GW<br>Name : **SIPInterface_SfB**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface**<br>Application Type : **SBC**<br>TLS Port : **5067**<br>TLS Context Name : **TLS Context**<br><br>Name : **SIPInterface_BTIP**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface**<br>Application Type : **SBC**<br>TCP Port : **5060** |
| **Proxy Set Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > Proxy Set Table | Proxy Set Table for Skype traffic (SBA or SfB)<br>Name : **ProxySet for Skype Traffic**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface**<br>SBC IPv4 SIP Interface : **SIP Interface for Skype Traffic**<br>Proxy Load Balancing Method : **Round Robin**<br>Proxy Keep-Alive Time : **60**<br>Proxy Keep-Alive : **Using OPTIONS**<br><br>(Proxy Address Table)<br>1 Entries : **FQDN or @IP of SBA:5060 TCP** (for SBA)<br>X Entries : **FQDN or @IPs of Mediation Pool:5060 TCP** (for HQ with GW aboard)<br>X Entries : **FQDN or @IPs of Mediation Pool:5067 TLS** (for SfB)<br><br>Proxy Set Table for BTIP Traffic<br>Name : **ProxySet for BTIP Traffic**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface**<br>SBC IPv4 SIP Interface : **SIP Interface for BTIP Traffic**<br>Proxy Load Balancing Method : **Round Robin**<br>Proxy Keep-Alive Time : **60**<br>Proxy Keep-Alive : **Using OPTIONS**<br><br>(Proxy Address Table)<br>2 Entries : **FQDN or @IP of aSBC ACME:5060 TCP** |
| **IP Group Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > IP Group Table | IP Group Table for Skype traffic (SBA or SfB)<br>Name : **IP Profile for Skype Traffic**<br>Type : **Server**<br>Proxy Set : **Proxy Set for Skype Traffic** |

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| Menu | Value |
|---|---|
| | IP Profile : **IP Profile for Skype Traffic**<br>Media Realm : **Media Realm for Skype traffic**<br><br>IP Group Table for BTIP traffic<br>Name : **IP Profile for BTIP Traffic**<br>Type : **Server**<br>Proxy Set : **Proxy Set for BTIP Traffic**<br>IP Profile : **IP Profile for BTIP Traffic**<br>Media Realm : **Media Realm for BTIP traffic** |
| **SIP Definitions** | |
| **General Parameters** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > SIP Definitions > General Parameters | PRACK Mode : **Supported**<br>Channel Select Mode : **Cyclic Ascending**<br>Enable Early Media : **Enable** |
| **SBC** | |
| **Allowed Audio Coders Group** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > SBC > Allowed Audio Coders Group | Allowed Audio Coders Group ID<br>Coder Name 1 : **G711A-Law**<br>Coder Name 2 : **G711U-Law** |
| **IP-to-IP Routing Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > SBC > IP-to-IP Routing Table | SIP Options rule<br>Name : **SIP Options**<br>Alternative Route Options: **Route Row**<br>Source IP Group : **Any**<br>Request Type : **OPTIONS**<br>Destination Type : **Dest Address**<br>Destination IP Group : **None**<br>Destination SIP Interface : **None**<br>Destination Address : **internal**<br><br>Skype to BTIP rule<br>Name : **Skype to BTIP**<br>Alternative Route Options: **Route Row**<br>Source IP Group : **Skype IP Group**<br>Request Type : **All**<br>Destination Type : **IP Group**<br>Destination IP Group : **BTIP IP Group**<br>Destination SIP Interface : **BTIP SIP Interface**<br><br>BTIP to Skype rule<br>Name : **BTIP to Skype**<br>Alternative Route Options: **Route Row**<br>Source IP Group : **BTIP IP Group**<br>Request Type : **All**<br>Destination Type : **IP Group**<br>Destination IP Group : **BTIP IP Group** |

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| Menu | Value |
| --- | --- |
| | Destination SIP Interface : **Skype SIP Interface** |
| **Gateway for PSTN calls (Annex 1) Only for RS SBA and RS GW** | |
| **Trunk Group** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Trunk Group | Configure Group Index<br>Module : **PRI**<br>From/To Trunk : **1**<br>Channels : **1-31**<br>Phone Number : **Phone number used for the Trunk**<br>Trunk Group ID : **Trunk Group ID associated** |
| **Trunk Group Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Trunk Group Settings | Add Trunk Group Settings<br>Name : **E1 PSTN**<br>Trunk Group ID : **Trunk Group ID associated**<br>Channel Selected Mode : **Cyclic Descending**<br>Registration Mode : **Don't Register** |
| **Trunk Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > PSTN > Trunk Settings | Protocol Type : **E1 EURO ISDN**<br>Line Code : **HDB3**<br>Framing Method : **Extend super Frame** |
| **VoIP Network Configuration** | |
| **Media Realm Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > Media Realm Table | Can be the same as Skype Media Realm<br>Name : **MRm for Skype**<br>IPv4 Interface Name : **Mediant IPv4 Interface**<br>Port Range Start : **16900**<br>Number of Media Session Legs : **50**<br>Port Range End : **Filled automatically**<br>Default Media Realm : **Yes** |
| **SRD Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > SRD Table | Same as Skype SRD Table<br>Name : **DefaultSRD** |
| **SIP Interface Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > SIP Interface Table | SIP Interface Table<br>Name : **SIPInterface_PSTN**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface for E1/Analog**<br>Application Type : **GW**<br>TCP Port : **5060** |
| **Proxy Set Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > Proxy Set Table | Proxy Set Table for PSTN traffic<br>Name : **ProxySet for PSTN Traffic**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface for E1/Analog** |

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| Menu | Value |
|---|---|
| | SBC IPv4 SIP Interface : **SIP Interface for PSTN Traffic**<br>Proxy Load Balancing Method : **Round Robin**<br>Proxy Keep-Alive Time : **60**<br>Proxy Keep-Alive : **Using OPTIONS**<br><br>(Proxy Address Table)<br>1 Entry : **FQDN or @IP of SBA:5060 TCP** |
| **IP Group Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > IP Group Table | IP Group Table for Skype traffic<br>Name : **IP Profile for PSTN Traffic**<br>Type : **Server**<br>Proxy Set : **Proxy Set for PSTN Traffic**<br>IP Profile : **IP Profile for Skype Traffic**<br>Media Realm : **Media Realm for Skype Traffic** |
| **Routing** | |
| **General Parameters** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Routing > General Parameters | Enable Alt Routing Tel to IP : **Enable** |
| **IP To Trunk Group Routing** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Routing > IP To Trunk Group Routing | Skype To PSTN rule<br>Name : **Skype To PSTN**<br>Source IP Group : **Skype IP Group**<br>Source SIP Interface : **PSTN SIP Interface**<br>Trunk Group ID : **PSTN Trunk Group ID**<br>Destination Type : **Trunk Group** |
| **TEL To IP** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Routing > TEL To IP | PSTN To Skype rule<br>Name : **PSTN To Skype**<br>Source Trunk Group ID : **PSTN Trunk Group ID**<br>Destination IP Group : **Skype IP Group**<br>SIP Interface : **PSTN SIP Interface**<br>IP Profile : **Skype IP Profile** |
| **Gateway for Analog calls (Annex 2)** | |
| **Trunk Group** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Trunk Group | Configure Group Index<br>Module : **FXS**<br>Channels : **1**<br>Phone Number : **Analog number in e164 format**<br>Trunk Group ID : **Trunk Group ID for Analog** |
| **Trunk Group Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Trunk Group Settings | Add Trunk Group Settings<br>Name : **Analog**<br>Trunk Group ID : **Trunk Group ID for Analog**<br>Channel Selected Mode : **By Dest Phone Number**<br>Registration Mode : **Don't Register** |

| Menu | Value |
| --- | --- |
| **Analog Settings** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Media > Analog Settings | Analog Metering Type : **12 Khz Sinusoidal bursts**<br>FXS Coefficient Type : **Europe** |
| **VoIP Network Configuration** | |
| **Media Realm Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > Media Realm Table | Can be the same as Skype Media Realm<br>Name : **MRm for Skype**<br>IPv4 Interface Name : **Mediant IPv4 Interface**<br>Port Range Start : **16900**<br>Number of Media Session Legs : **50**<br>Port Range End : **Filled automatically**<br>Default Media Realm : **Yes** |
| **SRD Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > SRD Table | Same as Skype SRD Table<br>Name : **DefaultSRD** |
| **SIP Interface Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > SIP Interface Table | SIP Interface Table<br>Name : **SIPInterface_Analog**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface for E1/Analog**<br>Application Type : **GW**<br>TCP Port : **5060** |
| **Proxy Set Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > Proxy Set Table | Proxy Set Table for Analog traffic<br>Name : **ProxySet for Analog Traffic**<br>SRD : **DefaultSRD**<br>Network Interface : **Mediant IPv4 Interface for E1/Analog**<br>SBC IPv4 SIP Interface : **SIP Interface for Analog Traffic**<br>Proxy Load Balancing Method : **Round Robin**<br>Proxy Keep-Alive Time : **60**<br>Proxy Keep-Alive : **Using OPTIONS**<br><br>(Proxy Address Table)<br>1 Entries : **FQDN or @IP of SBA:5060 TCP** |
| **IP Group Table** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > VoIP Network > IP Group Table | IP Group Table for Skype traffic<br>Name : **IP Profile for Analog Traffic**<br>Type : **Server**<br>Proxy Set : **Proxy Set for Analog Traffic**<br>IP Profile : **IP Profile for Skype Traffic**<br>Media Realm : **Media Realm for Skype Traffic** |
| **Manipulations** | |
| **IP To Trunk Group Routing** | |

| Menu | Value |
|---|---|
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Manipulations > IP To Trunk Group Routing | Skype To Analog manipulation rule<br>Name : **Skype To Analog**<br>Source IP Group : **Skype IP Group**<br>Destination Prefix : **Analog phone number** |
| **TEL To IP** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Manipulations > TEL To IP | Analog To Any manipulation rule<br>Name : **Analog To Any**<br>Source Trunk Group ID : **Analog Trunk Group ID**<br>Destination IP Group : **Any**<br>Prefix to Add : **+** |
| **Routing** | |
| **IP To Trunk Group Routing** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Routing > IP To Trunk Group Routing | Skype To Analog routing rule<br>Name : **Skype To Analog**<br>Source IP Group : **Skype IP Group**<br>Source SIP Interface : **Analog SIP Interface**<br>Destination Phone Prefix : **Analog number in e164**<br>Destination Trunk Group : **Trunk Group**<br>Trunk Group ID : **2** |
| **TEL To IP** | |
| On the AudioCodes Mediant WebUi Interface:<br>(Advanced mode)<br>Configuration > VoIP > Gateway > Routing > TEL To IP | Analog To Skype routing rule<br>Name : **Analog To Skype**<br>Source Trunk Group ID : **Analog Trunk Group ID**<br>Destination IP Group : **Skype IP Group**<br>SIP Interface : **Analog SIP Interface**<br>IP Profile : **Skype IP Profile** |

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| Menu | Value |
|---|---|

CAC Configuration Checklist

| **CAC Configuration** | |
|---|---|
| **Enable CAC** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ *Set-CsNetworkConfiguration* ***-EnableBandwidthPolicyCheck***<br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Global | SFB PowerShell<br>**EnableBandwidthPolicyCheck** parameter has to be set to **1**<br><br>SFB Control Panel<br><br>**Enable call admission control** parameter has to be **checked** |
| **Media bypass configuration (**In case of RS SBA and/or RS Default) | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***$a= New-CsNetworkMediaBypassConfiguration -alwaysByPass $false -Enabled $false***<br>✓ *Set-CsNetworkConfiguration* ***–MediaBypassSettings $a***<br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>Network Configuration >Global | SFB PowerShell<br><br>✓ ***AlwaysByPass*** parameter has to be set to **false**<br>✓ ***Enable*** parameter has to be set to **false**<br><br>SFB Control Panel<br>✓ **Enable media bypass** parameter must not be **checked** |
| **Media bypass configuration (**In case of RS GW or a mix of RS GW, RS SBA and RS Default) | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***$a= New-CsNetworkMediaBypassConfiguration -alwaysByPass $ false -Enabled $true***<br>✓ *Set-CsNetworkConfiguration* ***–MediaBypassSettings $a***<br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Global | SFB PowerShell<br><br>✓ ***AlwaysByPass*** parameter has to be set to **false**<br>✓ ***Enable*** parameter has to be set to **true**<br>SFB Control Panel<br>✓ **Enable media bypass** parameter has to be **checked**<br>✓ **Choose "**Use sites and region configuration**"** |
| **Media bypass Trunk Configuration (Only in case of RS-GW)** | |
| **SFB Control Panel**<br><br>On the Skype for Business Control panel interface<br>✓ Voice Routing > Trunk Configuration<br><br>And then select the RS-GW Trunk to edit Trunk configuration | SFB Control Panel<br><br>✓ **Enable media bypass** parameter has to be **checked** |
| **Trunk configuration (SFB PowerShell)**<br><br>On the Skype for Business PowerShell Interface:<br>✓ *Set-CsTrunkConfiguration* ***–Identity <Site> –RTCPActiveCalls $False***<br>✓ *Set-CsTrunkConfiguration* ***–Identity <Site> –RTCPCallsOnHold $False*** | **-Site:** The name of the site |

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| Menu | Value |
|---|---|
| **Network Region** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkRegion* –Identity *<XdsIdentity>* -CentralSite *<Central_Site>* –AudioAlternatePath *$False* -Description *“All Locations”***<br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Global | SFB PowerShell<br>**-Identity:** The name of the network region<br>**-Central site:** The name of the central site as defined on SFB topology builder<br><br>SFB Control Panel<br>**Identity:** The name of the network region<br>**Central site:** The name of the central site as defined on SFB topology builder<br>**Audio alternate path**: Recommended to disable |
| **Bandwidth Policy profiles** | |
| **CAC Onnet – Network sites and Network Region CAC** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –Description “*Descr Name*” -AudioBWLimit *<AudiototalBW>* -AudioBWSessionLimit *<AudiosessionBW>* -VideoBWLimit *<VideototalBW>* -VideoBWSessionLimit *<VideoSessionBW>***<br><br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Bandwidth Policy | SFB PowerShell<br>**-Identity:** The name of the bandwidth region (eg: **CAC_basse**)<br>**-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy<br>**-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **100**<br>**-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br><br>SFB Control Panel<br>**Identity:** The name of the bandwidth region (eg: **CAC_basse**)<br>**AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy<br>**AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to 100<br>**VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>on SFB topology builder |
| **CAC SIP Trunk – Inter site CAC** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –** | SFB PowerShell<br>**-Identity:** The name of the bandwidth region (eg: **CAC_SIPTrunk**)<br>**-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated |

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| Menu | Value |
|---|---|
| ***Description* “*Descr Name*” *-AudioBWLimit* *<AudiototalBW>* *-AudioBWSessionLimit* *<AudiosessionBW>* *-VideoBWLimit* *<VideototalBW>* *-VideoBWSessionLimit* *<VideoSessionBW>***<br><br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Bandwidth Policy | to this BW profile policy<br>**-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **97**<br>**-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br><br>SFB Control Panel<br>**Identity:** The name of the bandwidth region (eg: CAC_SIPTrunk)<br>**AudioBWLimit:** The total bandwidth allowed for BT/BTIP calls on network sites associated to this BW profile policy<br>**AudioBWSession Limit:** The session bandwidth allowed for one BT/BTIP call on network site associated to this BW profile policy → has to be set to **97**<br>**VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>on SFB topology builder |
| **CAC Zero – BT/BTIP network site to Network region CAC** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkBandwidthPolicyProfile* *-Identity* *<BWname>* – *Description* “*Descr Name*” *-AudioBWLimit* *<AudiototalBW>* *-AudioBWSessionLimit* *<AudiosessionBW>* *-VideoBWLimit* *<VideototalBW>* *-VideoBWSessionLimit* *<VideoSessionBW>***<br><br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Bandwidth Policy | SFB PowerShell<br>**-Identity:** The name of the bandwidth region (eg: **CAC_Zero**)<br>**-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy → parameter has to be set to **0**<br>**-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **40**<br>**-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br><br>SFB Control Panel<br>**Identity:** The name of the bandwidth region (eg: **CAC_Zero**)<br>**AudioBWLimit:** The total bandwidth allowed for BT/BTIP calls on network sites associated to this BW profile policy → parameter has to be set to **0**<br>**AudioBWSession Limit:** The session |

| Menu | Value |
|---|---|
| | bandwidth allowed for one BT/BTIP call on network site associated to this BW profile policy → has to be set to **40**<br>**VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>on SFB topology builder |
| **CAC Edge – Edge network site to Network region CAC** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –Description “*Descr Name*” -AudioBWLimit *<AudiototalBW>* -AudioBWSessionLimit *<AudiosessionBW>* -VideoBWLimit *<VideototalBW>* -VideoBWSessionLimit *<VideoSessionBW>***<br><br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration >Bandwidth Policy | SFB PowerShell<br>**-Identity:** The name of the bandwidth region (eg: **CAC_Edge**)<br>**-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy → parameter has to be set to **9999999999**<br>**-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **100**<br>**-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br><br>SFB Control Panel<br>**Identity:** The name of the bandwidth region (eg: **CAC_Edge**)<br>**AudioBWLimit:** The total bandwidth allowed for BT/BTIP calls on network sites associated to this BW profile policy → parameter has to be set to **999999999**<br>**AudioBWSession Limit:** The session bandwidth allowed for one BT/BTIP call on network site associated to this BW profile policy → has to be set to **100**<br>**VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>**VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation)<br>on SFB topology builder |
| **Network Sites** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkSite*-NetworkSIteID *<NSname>* –Description “*Descr Name*” -NetworkRegionID *<NRname>* -BWPolicyProfileID *<BWPname>*** | SFB PowerShell<br>**-NetworkSiteID:** The name of the network site<br>**-Description:** Optional<br>**-NetworkRegionID:** Select the network region to associate to created network site<br>**-BWPolicyProfileID:** Select the bandwidth |

| Menu | Value |
|---|---|
| **SFB Control Panel**<br>On the Skype for Business control panel interface:<br>✓ Network Configuration > Site | profile policy to associate to created network site<br><br>SFB Control Panel<br>**-NetworkSiteID:** The name of the network site<br>**-Description:** Optional<br>**-NetworkRegionID:** Select the network region to associate to created network site<br>**-BWPolicyProfileID:** Select the bandwidth profile policy to associate to created network site |
| **Inter Site Policy** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkInterSitePolicy–Identity <NetworkInterSitename>-BWPolicyProfileID <SIPTRUNK_BWPname> -NetworkSiteID1 <NS1name>-NetworkSiteID2 <BTIP_NS_name>*** | SFB PowerShell<br>**-Identity:** The name of the network inter site policy<br>**-BWPolicyProfileID:** Select the bandwidth profile policy to associate to created network inter site policy<br>**-NetworkSiteID1:** parameter has to correspond to the network site 1 (SFB component) to associate to BTIP using inter site policy<br>**-NetworkSiteID2:** parameter has to correspond to the BT/BTIP network site name<br>**WARNING:** NO Inter site for Remote site Gateway |
| **Subnets** | |
| **SFB PowerShell**<br><br>On the Skype for Business PowerShell Interface:<br>✓ ***New-CsNetworkSubnet-SubnetID <firstsubnetIPaddress>-MaskBits <maskwo/> -NetworkSiteID <associated NS_name>***<br><br>**SFB Control Panel**<br>On the Skype for Business control panel interface:<br>Network Configuration > Subnet | SFB PowerShell<br>**-SubnetID:** The first IP address of the corresponding subnet<br>**-MaskBits:** The subnet mask to associate to subnet to create without / (eg:**32**)<br>**-NetworkSiteID:** Select the network site name from the drop down list to associate to this subnet (eg: **BTIP**)<br><br>SFB Control Panel<br>**-SubnetID:** The first IP address of the corresponding subnet<br>**-MaskBits:** The subnet mask to associate to subnet to create without / (eg:**32**)<br>**-NetworkSiteID:** Select the network site name from the drop down list to associate to this subnet (eg: **BTIP**) |
| **Configuration requirements (warnings)** | |
| **Configuring Clients ports range for LPE and SoftPhone** | |

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| Menu | Value |
| --- | --- |
| **SFB PowerShell**<br>On the Skype for Business PowerShell Interface<br>***Set-CsConferencingConfiguration* –ClientMediaPortRangeEnabled *$true* –ClientAudioPort *50060* –ClientAudioPortRange *48*** | SFB PowerShell<br>-**ClientMediaPortRangeEnable :** must be enabled in order to use the specific range<br>-**ClientAudioPort :** corresponds to the first port used for audio<br>-**ClientAudioPortRange :** corresponds to the audio range |
| **Configuring Clients ports range for VVX** | |
| ✓ Using VVX Web UI :<br>- Navigate through the VVX Web Interface: http:<VVX_IP_Address><br>- Go to Settings tab > Network menu > RTP<br>- Configure the Port Range Start to: 50060 | VVX WebUI |
| ✓ Using VVX configuration file (.cfg)<br>- Configure the following line in the VVX configuration file :<br>tcpIpApp.port.rtp.mediaPortRangeStart="50060"<br>- Import the new configuration file to the VVX using the WebUI or through the IIS server | VVX WebUI<br>or<br>IIS Server |
| **Others Devices** | |
| ✓ Check that the audio range port respect the OBS recommendations<br>The default audio range is: 50060-50107. | |

# 6 Skype for Business Online – AudioCodes Cloud Connector Edition configuration checklist

## 6.1 Generic configuration

| Menu | Value |
|---|---|
| **TCP Mediation Server** | |
| The TCP Mediation Server must be 5068:<br>On the PowerShell interface execute the following command:<br>**Set-CSMediationServer** -Identity *<MediationServer:MS-FQDN>* -SipClientTcpPort *<5068>* | Identity: must match corresponding **mediation server FQDN**<br>SipClientTcpPort: must be set to **5068** |
| **PSTN Gateway** | |
| During Cloud Connector Edition<br>Trunk must be created for SBC | SIP Transport protocol: **TCP**<br>Mediation Server port: **5068** |
| **O365 Cloud Connector Edition** | |
| **Register Check**<br>Open an online session on the PowerShell, then execute:<br>Get-CsTenantFederationConfiguration | SharedSipAddressSpace: must be set to **$true** |
| Open an online session on the PowerShell, then execute:<br>Get-CsTenantHybridConfiguration | UseOnPremiseDialPlan: must be set to **$false** |
| **CCE admin account association**<br>Open an online session on the PowerShell, then execute:<br>Set-CsHybridMediationServer -Id *<UserName>* -FQDN *<MSFQDN>* -AccessProxyExternalFqdn *<EdgeExternationFQDN>* | ID: must be filled with **CCE admin account SIP address**<br>FQDN: must be filled with the associated **Mediation Server FQDN**<br>AccessProxyExternalFqdn: must be filled with the **Edge Server External access FQDN** |
| **User Management** | |
| **User creation in O365 Active Directory**<br>Connect to O365 tenant and create a new user. | DNS: must be the **customer DNS** 'Not the xxx.onmicrosoft.com default domain'<br>User country: must be filled 'important for dial plan usage'<br>Assign appropriate License**:**<br>Plan E3 with CloudPBX add-on option<br>Or<br>Plan E5 'CloudPBX included by default' |
| **Policies assignment and phone number attribution to User**<br>Open an online session on the PowerShell, then execute:<br>Set-CsUser -Identity *<UserName>* -EnterpriseVoiceEnabled *$true* -HostedVoiceMail *$true* -OnPremLineURI *<tel:+PhoneNumber>* | Identity: **User name**<br>EnterpriseVoiceEnabled: **$true**<br>HostedVoiceMail: **$true**<br>OnPremLineUri: **tel:+E164 format number** |
| **User Association to appropriate Cloud Connector Edition**<br>Open an online session on the PowerShell, then execute:<br>Set-CsUserPstnSettings -Id *<UserName>* -HybridPSTNSite *<PSTNSiteName>* | Id: **User name**<br>HybridPSTNSite: **appropriate CCE where the user will be associated** |

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## 6.2 Standalone specific configuration

| Menu | Value |
|---|---|
| **Cloud Connector Edition Wizard (version 2.1.0.22)** | |
| **CCE General Information (step)**<br>During wizard installation ensure that CCE is deployed on standalone mode | Installation Type: **Standalone CCE or First CCE in HA**<br>Site Directory: path to shared directory where CCE files will be stored<br>User: **Skype for Business Online admin user name**<br>Password: Skype for Business Online admin password |
| **CCE Gateway configuration (step)**<br>During the CCE gateway configuration, following mediation server options must be configured | EnableReferSupport: **False**<br>EnableFastFailoverTimer: **False**<br>ForwardPAI: **False**<br>ForwardCallHistory: **True** |
| **Mediation Server “Manual configuration”**<br>Following mediation server parameters must be configured manually through PowerShell Online Interface In addition to above configured parameters<br>To configure the mediation server trunk with VISIT SIP parameters:<br>- Logon the mediation server using the CCE domain<br>- Open PS console and execute the following cmdlet<br>Set-CsTrunkconfiguration -RTCPActiveCalls *$false* -RTCPCallsOnHold *$false* -SRTPMode *Optional* | RTCPActiveCalls: $**False**<br>RTCPCallsOnHold: $**False**<br>SRTPMode: **Optional**<br><br>**WARNING:**<br>The manual configuration will be lost after each CCE update. |
| **AudioCodes SBC Configuration Wizard (wizard version min 2.20)** | |
| **Product (Step 1 of 7)**<br>Choose product type and version: | Product: **Mediant 800, 1000 or software** depending on the Gateway type used for the deployment<br>Version: **7.2**<br>Use defaults from template must be **checked**<br>End Customer: **corresponds to customer name ex: “OBS”**<br>Country: **corresponds to customer country ex: “France”**<br>Integrator: **if needed corresponds to integrator name ex: “OBS”**<br>Installer: **if needed corresponds to installer name ex: “OBS”** |
| **General Setup (Step 2 of 7)**<br>Choose application type, configuration template and network setup | Application: **Cloud Connector (CCE) Appliance**<br>Equipment (interop): **SIP Trunk**<br>SIP Trunk: **Orange BTIP SIP Trunk**<br>Network Setup: **One port:LAN** |
| **System Configuration (Step 3 of 7)**<br>Configure system parameters | Primary NTP Server: **“Optional”** NTP server IP address<br>Secondary NTP Server: **“Optional”** backup NTP server IP address<br>Time Zone: depending on customer local time zone “**default value GMT”**<br>Web Interface: **HTTPS** |

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| Menu | Value |
|---|---|
| | CLI Interface: **SSH**<br>Enable Syslog: **Checked**<br>Syslog IP: **IP address of the syslog server**<br>Local DNS Table**: Unchecked** |
| **User Management** | |
| **LAN Interface Configuration (Step 4 of 7)**<br>Configure LAN network interface | Physical Port: **Group 1(GE_1)**<br>Vlan ID: **Untagged**<br>IP address: **SBC IP address** (ex: 192.168.0.2)<br>Subnet mask: **SBC subnet mask** (ex:255.255.0.0)<br>Default Gateway: **SBC default gateway ip address** (ex:192.168.0.1)<br>Primary DNS: **IP address of the DNS server used by the SBC**<br>Secondary DNS: **"Optional"**<br>OAM Interface: **LAN** |
| **IP-PBX Configuration (Step 5 of 7)**<br>Configure Microsoft Skype CCE address and communication protocol details | Address: **Mediation Server IP address**<br>Backup Address: **Empty**<br>SIP Domain: **CCE FQDN**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5068**<br>Listening Port: **5068**<br>Media Protocol: **RTP**<br>Base Port: **6000**<br>Number of Sessions: **1000** |
| **SIP Trunk Configuration (Step 6 of 7)**<br>Configure Orange BTIP SIP Trunk Address and communication protocol details | Address: **aSBC Nominal Address**<br>Backup Address: **aSBC Backup Address**<br>SIP Domain: **Empty**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5060**<br>Listening Port: **5060**<br>Media Protocol: **RTP**<br>Base Port: **16400**<br>Number of Sessions: **1000**<br>Account Type: **None**<br>Trunk Main Line: **Empty** |
| **Number Manipulation and routing (Step 7 of 7) "Optional"**<br>Configure number manipulation rules and routing policy | Check needed manipulation type and fill:<br>Prefix<br>Remove: corresponds to number of digits to remove<br>Add: corresponds to number of digits to add |

## 6.3 High availability specific configuration

| Menu | Value |
|---|---|
| **Cloud Connector Edition 1 Wizard (version 2.1.0.22)** | |
| **CCE General Information (step)**<br>During wizard installation ensure that CCE is deployed on standalone mode | Installation Type: **Standalone CCE or First CCE in HA**<br>Site Directory: path to shared directory where CCE 1 files will be stored<br>User: **Skype for Business Online admin user name**<br>Password: Skype for Business Online admin password |
| **CCE Gateway configuration (step)**<br>During the CCE gateway configuration, following mediation server options must be configured | EnableReferSupport: **False**<br>EnableFastFailoverTimer: **False**<br>ForwardPAI: **False**<br>ForwardCallHistory: **True** |
| **Mediation Server "Manual configuration"**<br>Following mediation server parameters must be configured manually through PowerShell Online Interface In addition to above configured parameters<br>To configure the mediation server trunk with VISIT SIP parameters:<br>- Logon the mediation server using the CCE domain<br>- Open PS console and execute the following cmdlet<br>Set-CsTrunkconfiguration -RTCPActiveCalls *$false* -RTCPCallsOnHold *$false* -SRTPMode *Optional* | RTCPActiveCalls: $**False**<br>RTCPCallsOnHold: $**False**<br>SRTPMode: **Optional**<br><br>**WARNING:**<br>The manual configuration will be lost after each CCE update. |
| **Cloud Connector Edition 2 Wizard (version 2.1.0.22)** | |
| **CCE General Information (step)**<br>During wizard installation ensure that CCE is deployed on standalone mode | Installation Type: **HA**<br>Site Directory: path to shared directory where CCE 1 installation files were stored<br>User: **Skype for Business Online admin user name**<br>Password: Skype for Business Online admin password |
| **CCE Gateway configuration (step)**<br>During the CCE gateway configuration, following mediation server options must be configured | EnableReferSupport: **False**<br>EnableFastFailoverTimer: **False**<br>ForwardPAI: **False**<br>ForwardCallHistory: **True** |
| **Mediation Server "Manual configuration"**<br>Following mediation server parameters must be configured manually through PowerShell Online Interface In addition to above configured parameters<br>To configure the mediation server trunk with VISIT SIP parameters:<br>- Logon the mediation server using the CCE domain<br>- Open PS console and execute the following cmdlet<br>Set-CsTrunkconfiguration -RTCPActiveCalls *$false* -RTCPCallsOnHold *$false* -SRTPMode *Optional* | RTCPActiveCalls: $**False**<br>RTCPCallsOnHold: $**False**<br>SRTPMode: **Optional**<br><br>**WARNING:**<br>The manual configuration will be lost after each CCE update. |
| **AudioCodes SBC 1 Configuration Wizard (wizard version min 2.20)** | |
| **Product (Step 1 of 7)**<br>Choose product type and version: | Product: **Mediant 800, 1000 or software** depending on the Gateway type used for the deployment<br>Version: **7.2** |

| Menu | Value |
|---|---|
| | Use defaults from template must be **checked**<br>End Customer: **corresponds to customer name ex: "OBS"**<br>Country: **corresponds to customer country ex: "France"**<br>Integrator: **if needed corresponds to integrator name ex: "OBS"**<br>Installer: **if needed corresponds to installer name ex: "OBS"** |
| **General Setup (Step 2 of 7)**<br>Choose application type, configuration template and network setup | Application: **Cloud Connector (CCE) Appliance**<br>Equipment (interop): **SIP Trunk**<br>SIP Trunk: **Orange BTIP SIP Trunk**<br>Network Setup: **One port:LAN** |
| **System Configuration (Step 3 of 7)**<br>Configure system parameters | Primary NTP Server: **"Optional"** NTP server IP address<br>Secondary NTP Server: **"Optional"** backup NTP server IP address<br>Time Zone: depending on customer local time zone "**default value GMT"**<br>Web Interface: **HTTPS**<br>CLI Interface: **SSH**<br>Enable Syslog: **Checked**<br>Syslog IP: **IP address of the syslog server**<br>Local DNS Table**: Unchecked** |
| **User Management** | |
| **LAN Interface Configuration (Step 4 of 7)**<br>Configure LAN network interface | Physical Port: **Group 1(GE_1)**<br>Vlan ID: **Untagged**<br>IP address: **SBC IP address** (ex: 192.168.0.2)<br>Subnet mask: **SBC subnet mask** (ex:255.255.0.0)<br>Default Gateway: **SBC default gateway ip address** (ex:192.168.0.1)<br>Primary DNS: **IP address of the DNS server used by the SBC**<br>Secondary DNS: **"Optional"**<br>OAM Interface: **LAN** |
| **IP-PBX Configuration (Step 5 of 7)**<br>Configure Microsoft Skype CCE address and communication protocol details | Address: **Mediation Server IP address**<br>Backup Address: **Empty**<br>SIP Domain: **CCE FQDN**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5068**<br>Listening Port: **5068**<br>Media Protocol: **RTP**<br>Base Port: **6000**<br>Number of Sessions: **1000** |
| **SIP Trunk Configuration (Step 6 of 7)**<br>Configure Orange BTIP SIP Trunk Address and communication protocol details | Address: **aSBC Nominal Address**<br>Backup Address: **aSBC Backup Address**<br>SIP Domain: **Empty**<br>Keep Alive: **Checked** |

| Menu | Value |
| --- | --- |
| | Transport Type: **TCP**<br>Destination Port: **5060**<br>Listening Port: **5060**<br>Media Protocol: **RTP**<br>Base Port: **16400**<br>Number of Sessions: **1000**<br>Account Type: **None**<br>Trunk Main Line: **Empty** |
| **Number Manipulation and routing (Step 7 of 7) "Optional"**<br>Configure number manipulation rules and routing policy | Check needed manipulation type and fill:<br>Prefix<br>Remove: corresponds to number of digits to remove<br>Add: corresponds to number of digits to add |
| **SBC 1 High Availability IP interface configuration**<br>Configure IP interface for HA mode:<br>On the SBC1 WebUi interface > Setup menu > IP network > IP interface > Add new IP interface for HA | Name: **HA**<br>Application Type: **MAINTENANCE**<br>Ethernet Device: **HA Interface**<br>IP Address: **SBC IP address to use for HA**<br>Prefix Length: **Subnet length prefix (ex:30)** |
| **SBC 1 High Availability Ethernet Device configuration**<br>Configure Ethernet device for HA mode:<br>On the SBC1 WebUi interface > Setup menu > IP network > Ethernet devices > Add new Ethernet device for HA | Name: **HA**<br>VLAN ID: **99**<br>Underlying interface: **HA Group**<br>Tagging: **Untagged**<br>Prefix Length: **1500** |
| **SBC 1 High Availability Ethernet Group configuration**<br>Configure Ethernet group for HA mode:<br>On the SBC1 WebUi interface > Setup menu > IP network > Ethernet groups > Add new Ethernet group for HA | Index: **The number of index (ex:3)**<br>Mode: **Single or REDUN_2RX1_1TX**<br>Member1: **HA Physical port**<br>Member2: **Only in case of redundant mode, HA second port** |
| **SBC 1 High Availability Settings**<br>On the SBC1 WebUi interface > Setup menu > IP network > HA settings | HA Remote Address: **The IP address of the second SBC(ex:192.168.1.1)**<br>HA Device name: **The local SBC device name (ex: SBC2)**<br>Redundant HA device name: **The distant SBC HA device name (ex: SBC1)** |
| **SBC 1 High Availability .INI configuration file export**<br>Export the SBC1 .INI file including HA availability configuration | Check needed manipulation type and fill:<br>Prefix<br>Remove: corresponds to number of digits to remove<br>Add: corresponds to number of digits to add |
| **SBC 1 High Availability .INI configuration file modification**<br>Modify the SBC1 .INI file including HA availability configuration | HA Remote Address: **The IP address of the second SBC(ex:192.168.1.2)**<br>HAUnitIdName: **The local SBC device name (ex: SBC1)** |
| **SBC 2 High Availability settings**<br>Access the SBC2 using **its default IP address** | Import the modified .INI file configuration on the SBC2 |

## 6.4 Nominal/backup mode specific configuration

| Menu | Value |
|---|---|
| **Cloud Connector Edition Wizard (version 2.1.0.22)** | |
| **CCE General Information (step)**<br>During wizard installation ensure that CCE is deployed on standalone mode | Installation Type: **Standalone CCE or First CCE in HA**<br>Site Directory: path to shared directory where CCE files will be stored<br>User: **Skype for Business Online admin user name**<br>Password: Skype for Business Online admin password |
| **CCE Gateway configuration (step)**<br>During the CCE gateway configuration, following mediation server options must be configured | EnableReferSupport: **False**<br>EnableFastFailoverTimer: **False**<br>ForwardPAI: **False**<br>ForwardCallHistory: **True** |
| **Mediation Server "Manual configuration"**<br>Following mediation server parameters must be configured manually through PowerShell Online Interface In addition to above configured parameters<br>To configure the mediation server trunk with VISIT SIP parameters:<br>- Logon the mediation server using the CCE domain<br>- Open PS console and execute the following cmdlet<br>Set-CsTrunkconfiguration -RTCPActiveCalls *$false* -RTCPCallsOnHold *$false* -SRTPMode *Optional* | RTCPActiveCalls: $**False**<br>RTCPCallsOnHold: $**False**<br>SRTPMode: **Optional**<br><br>**WARNING:**<br>The manual configuration will be lost after each CCE update. |
| **Same configuration steps must be performed on All needed CCEs** | |
| **AudioCodes SBC Configuration Wizard (wizard version min 2.20)** | |
| **Product (Step 1 of 7)**<br>Choose product type and version: | Product: **Mediant 800, 1000 or software** depending on the Gateway type used for the deployment<br>Version: **7.2**<br>Use defaults from template must be **checked**<br>End Customer: **corresponds to customer name ex: "OBS"**<br>Country: **corresponds to customer country ex: "France"**<br>Integrator: **if needed corresponds to integrator name ex: "OBS"**<br>Installer: **if needed corresponds to installer name ex: "OBS"** |
| **General Setup (Step 2 of 7)**<br>Choose application type, configuration template and network setup | Application: **Cloud Connector (CCE) Appliance**<br>Equipment (interop): **SIP Trunk**<br>SIP Trunk: **Orange BTIP SIP Trunk**<br>Network Setup: **One port:LAN** |
| **System Configuration (Step 3 of 7)**<br>Configure system parameters | Primary NTP Server: **"Optional"** NTP server IP address<br>Secondary NTP Server: **"Optional"** backup NTP server IP address<br>Time Zone: depending on customer local time zone "**default value GMT"**<br>Web Interface: **HTTPS** |

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| Menu | Value |
|---|---|
| | CLI Interface: **SSH**<br>Enable Syslog: **Checked**<br>Syslog IP: **IP address of the syslog server**<br>Local DNS Table**: Unchecked** |
| **User Management** | |
| **LAN Interface Configuration (Step 4 of 7)**<br>Configure LAN network interface | Physical Port: **Group 1(GE_1)**<br>Vlan ID: **Untagged**<br>IP address: **SBC IP address** (ex: 192.168.0.2)<br>Subnet mask: **SBC subnet mask** (ex:255.255.0.0)<br>Default Gateway: **SBC default gateway ip address** (ex:192.168.0.1)<br>Primary DNS: **IP address of the DNS server used by the SBC**<br>Secondary DNS: **"Optional"**<br>OAM Interface: **LAN** |
| **IP-PBX Configuration (Step 5 of 7)**<br>Configure Microsoft Skype CCE address and communication protocol details | Address: **Mediation Server IP address**<br>Backup Address: **Empty**<br>SIP Domain: **CCE FQDN**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5068**<br>Listening Port: **5068**<br>Media Protocol: **RTP**<br>Base Port: **6000**<br>Number of Sessions: **1000** |
| **SIP Trunk Configuration (Step 6 of 7)**<br>Configure Orange BTIP SIP Trunk Address and communication protocol details | Address: **aSBC Nominal Address**<br>Backup Address: **aSBC Backup Address**<br>SIP Domain: **Empty**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5060**<br>Listening Port: **5060**<br>Media Protocol: **RTP**<br>Base Port: **16400**<br>Number of Sessions: **1000**<br>Account Type: **None**<br>Trunk Main Line: **Empty** |
| **Number Manipulation and routing (Step 7 of 7) "Optional"**<br>Configure number manipulation rules and routing policy | Check needed manipulation type and fill:<br>Prefix<br>Remove: corresponds to number of digits to remove<br>Add: corresponds to number of digits to add |
| **SBC 1 Nominal and Backup configuration**<br>On the SBC1 WebUi interface > Setup menu > Signalling & Media > Proxy > Skype proxy set | Name: **ProxySet_Skype**<br>SBC IPv4 SIP interface: **SIP interface Skype**<br>Proxy Hot Swap: **Enable**<br>Proxy Load Balancing Method: **Random Weights** |
| **Same configuration steps must be performed on both SBCs** | |

## 6.5 Round-Robin mode specific configuration

| Menu | Value |
|---|---|
| **Cloud Connector Edition Wizard (version 2.1.0.22)** | |
| **CCE General Information (step)**<br>During wizard installation ensure that CCE is deployed on standalone mode | Installation Type: **Standalone CCE or First CCE in HA**<br>Site Directory: path to shared directory where CCE files will be stored<br>User: **Skype for Business Online admin user name**<br>Password: Skype for Business Online admin password |
| **CCE Gateway configuration (step)**<br>During the CCE gateway configuration, following mediation server options must be configured | EnableReferSupport: **False**<br>EnableFastFailoverTimer: **False**<br>ForwardPAI: **False**<br>ForwardCallHistory: **True** |
| **Mediation Server "Manual configuration"**<br>Following mediation server parameters must be configured manually through PowerShell Online Interface In addition to above configured parameters<br>To configure the mediation server trunk with VISIT SIP parameters:<br>- Logon the mediation server using the CCE domain<br>- Open PS console and execute the following cmdlet<br>Set-CsTrunkconfiguration -RTCPActiveCalls *$false* -RTCPCallsOnHold *$false* -SRTPMode *Optional* | RTCPActiveCalls: $**False**<br>RTCPCallsOnHold: $**False**<br>SRTPMode: **Optional**<br><br>**WARNING:**<br>The manual configuration will be lost after each CCE update. |
| **Same configuration steps must be performed on both CCEs** | |
| **AudioCodes SBC Configuration Wizard (wizard version min 2.20)** | |
| **Product (Step 1 of 7)**<br>Choose product type and version: | Product: **Mediant 800, 1000 or software** depending on the Gateway type used for the deployment<br>Version: **7.2**<br>Use defaults from template must be **checked**<br>End Customer: **corresponds to customer name ex: "OBS"**<br>Country: **corresponds to customer country ex: "France"**<br>Integrator: **if needed corresponds to integrator name ex: "OBS"**<br>Installer: **if needed corresponds to installer name ex: "OBS"** |
| **General Setup (Step 2 of 7)**<br>Choose application type, configuration template and network setup | Application: **Cloud Connector (CCE) Appliance**<br>Equipment (interop): **SIP Trunk**<br>SIP Trunk: **Orange BTIP SIP Trunk**<br>Network Setup: **One port:LAN** |
| **System Configuration (Step 3 of 7)**<br>Configure system parameters | Primary NTP Server: **"Optional"** NTP server IP address<br>Secondary NTP Server: **"Optional"** backup NTP server IP address<br>Time Zone: depending on customer local time zone "**default value GMT"**<br>Web Interface: **HTTPS** |

| Menu | Value |
|---|---|
| | CLI Interface: **SSH**<br>Enable Syslog: **Checked**<br>Syslog IP: **IP address of the syslog server**<br>Local DNS Table**: Unchecked** |
| **User Management** | |
| **LAN Interface Configuration (Step 4 of 7)**<br>Configure LAN network interface | Physical Port: **Group 1(GE_1)**<br>Vlan ID: **Untagged**<br>IP address: **SBC IP address** (ex: 192.168.0.2)<br>Subnet mask: **SBC subnet mask** (ex:255.255.0.0)<br>Default Gateway: **SBC default gateway ip address** (ex:192.168.0.1)<br>Primary DNS: **IP address of the DNS server used by the SBC**<br>Secondary DNS: **"Optional"**<br>OAM Interface: **LAN** |
| **IP-PBX Configuration (Step 5 of 7)**<br>Configure Microsoft Skype CCE address and communication protocol details | Address: **Mediation Server IP address**<br>Backup Address: **Empty**<br>SIP Domain: **CCE FQDN**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5068**<br>Listening Port: **5068**<br>Media Protocol: **RTP**<br>Base Port: **6000**<br>Number of Sessions: **1000** |
| **SIP Trunk Configuration (Step 6 of 7)**<br>Configure Orange BTIP SIP Trunk Address and communication protocol details | Address: **aSBC Nominal Address**<br>Backup Address: **aSBC Backup Address**<br>SIP Domain: **Empty**<br>Keep Alive: **Checked**<br>Transport Type: **TCP**<br>Destination Port: **5060**<br>Listening Port: **5060**<br>Media Protocol: **RTP**<br>Base Port: **16400**<br>Number of Sessions: **1000**<br>Account Type: **None**<br>Trunk Main Line: **Empty** |
| **Number Manipulation and routing (Step 7 of 7) "Optional"**<br>Configure number manipulation rules and routing policy | Check needed manipulation type and fill:<br>Prefix<br>Remove: corresponds to number of digits to remove<br>Add: corresponds to number of digits to add |
| **SBC 1 Nominal and Backup configuration**<br>On the SBC1 WebUi interface > Setup menu > Signalling & Media > Proxy > Skype proxy set | Name: **ProxySet_Skype**<br>SBC IPv4 SIP interface: **SIP interface Skype**<br>Proxy Hot Swap: **Enable**<br>Proxy Load Balancing Method: **Round Robin** |
| **Same configuration steps must be performed on both SBCs** | |

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