Microsoft Lync 2013 Skype for Business 2015

Configuration Checklists for BTIP and Business Talk SIP services

18 january 2018

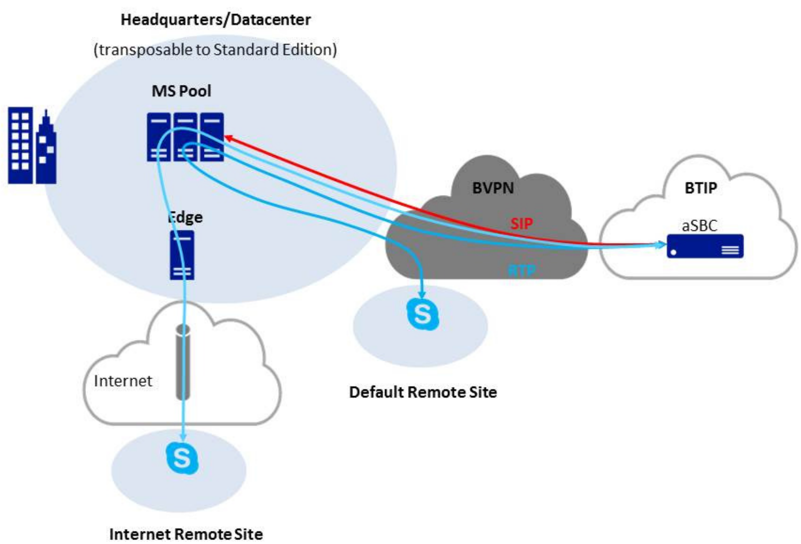
Lync 2013 Checklist version 1.6

Skype for Business 2015 Checklist version 1.10

Contents

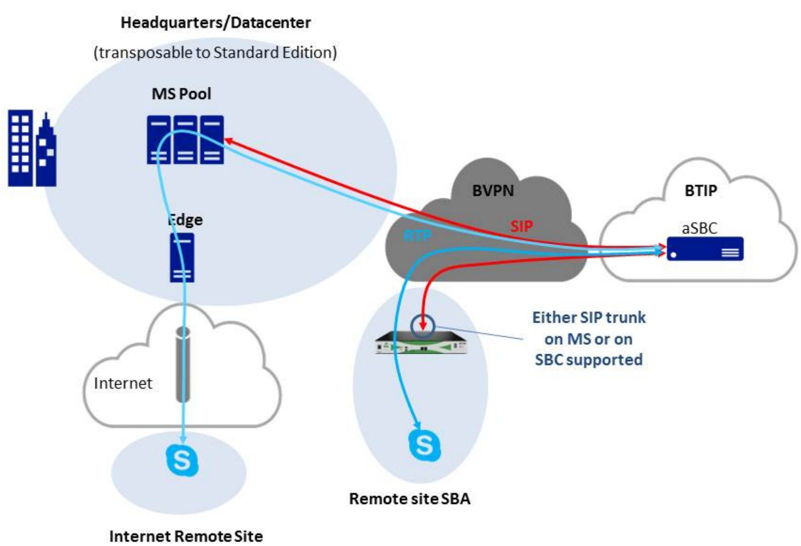
1 Main certified architectures

1.1 Centralized architecture

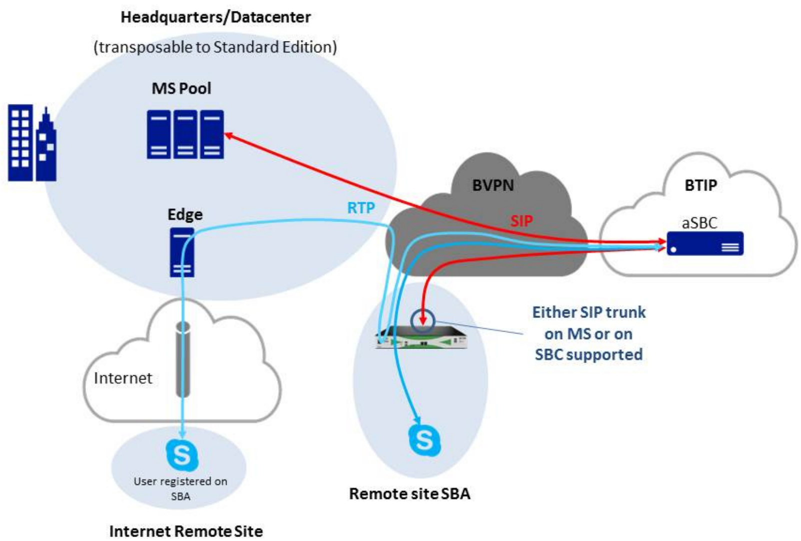


1.2 Remote site "SBA"

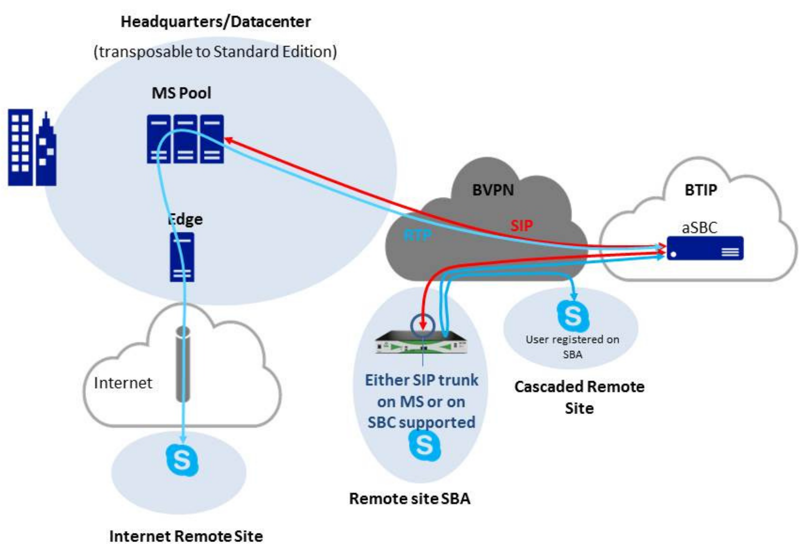
1.2.1 Example 1



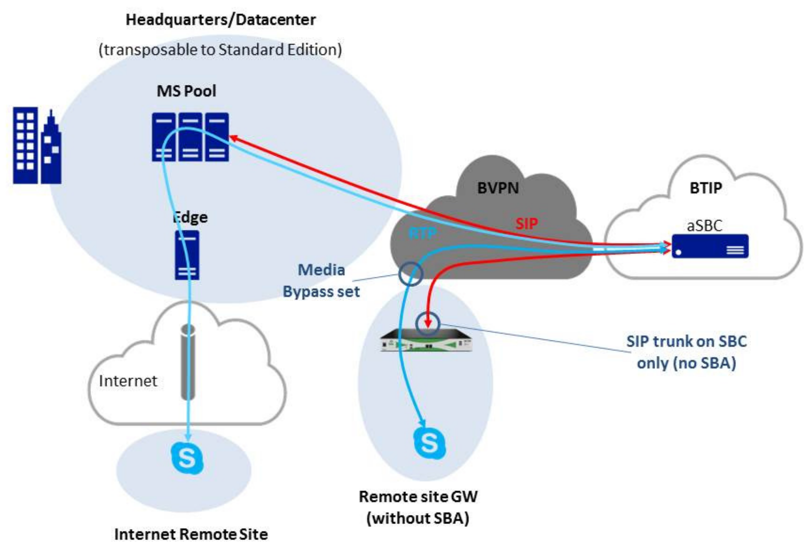
1.2.2 Example 2



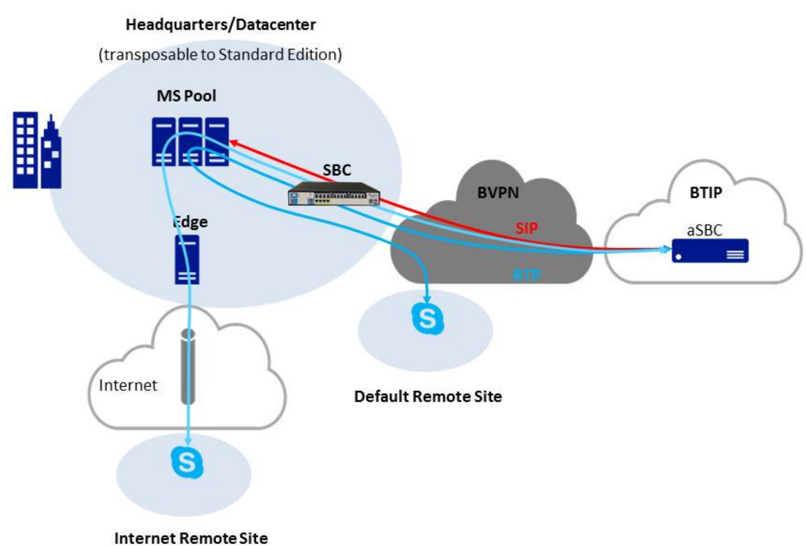
1.1 “Cascaded” remote site



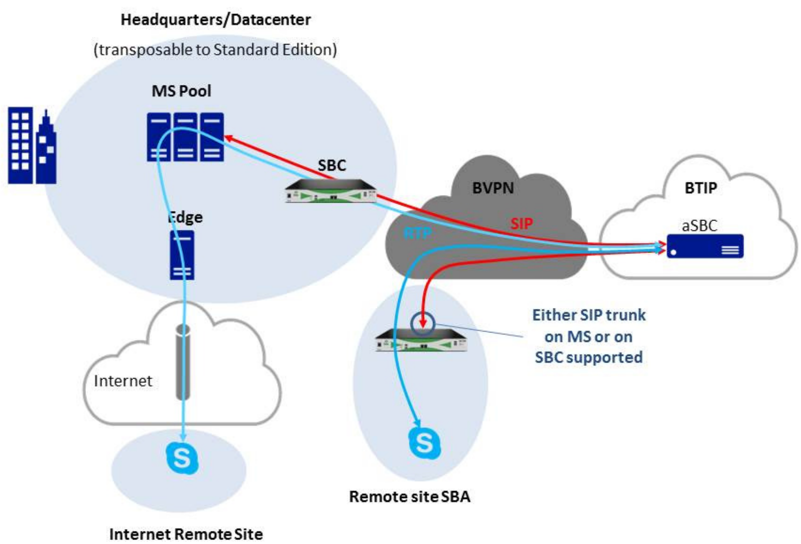
1.2 Remote site "GW"



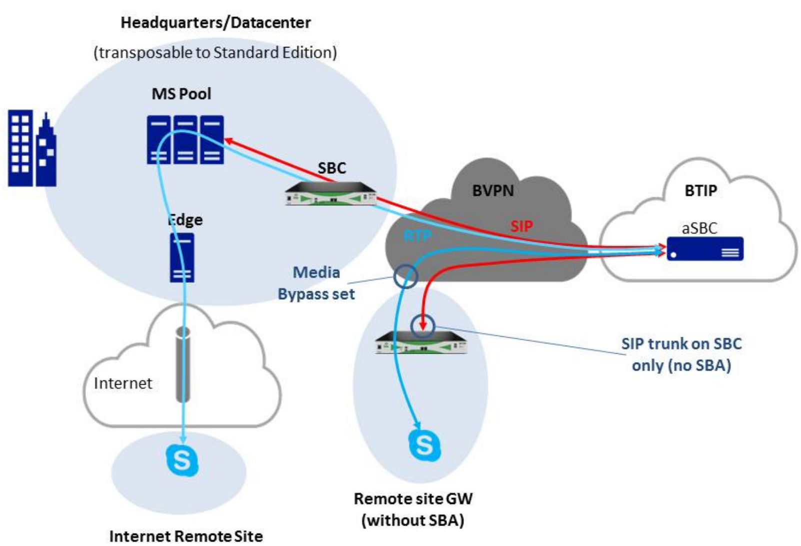
1.3 Centralized architecture with "GW aboard"



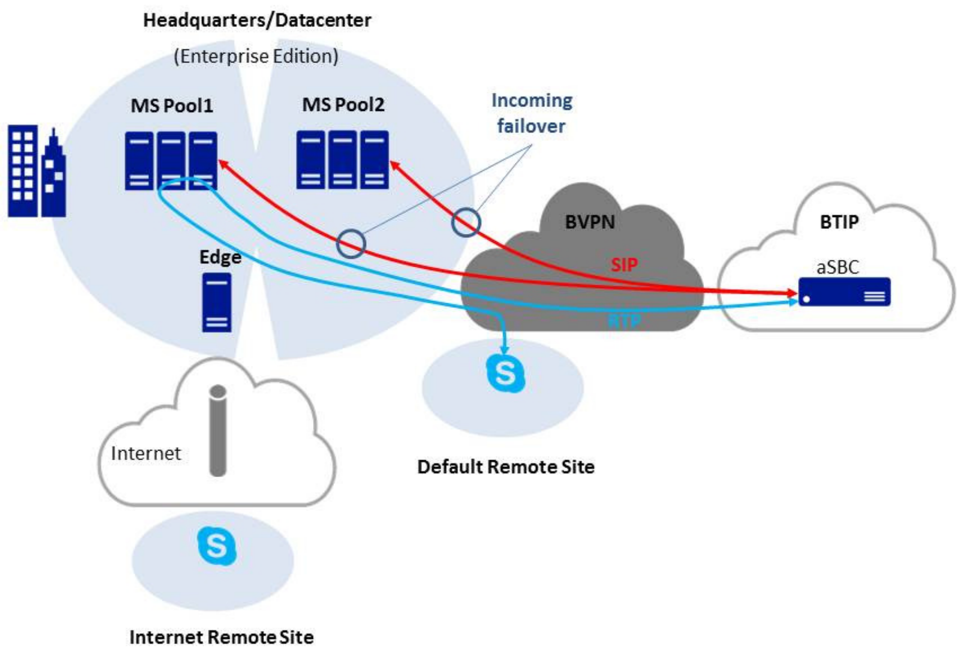
1.4 Remote site "SBA" and central site with "GW aboard"



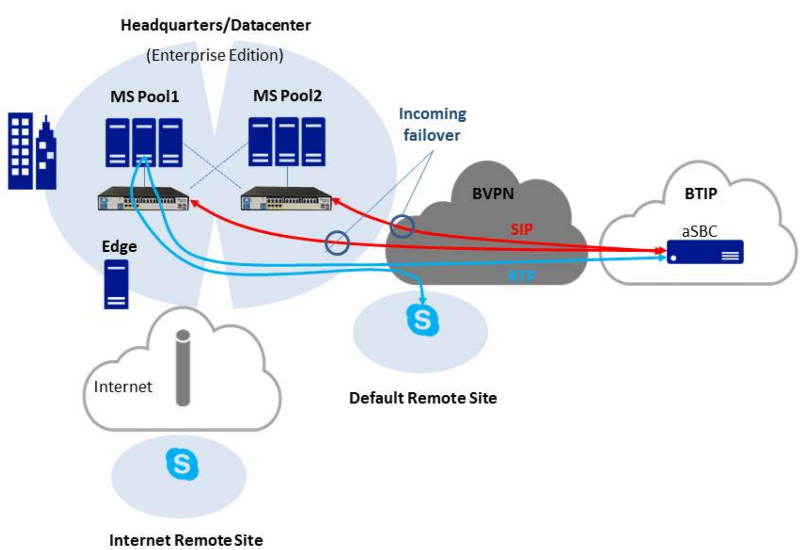
1.5 Remote site "GW" and central site with "GW aboard"



1.6 2-pool centralized architecture



1.7 2-pool architecture with “GW aboard” (Customer specific)



2 Parameters for connection to BTIP

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

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 Functionning mode: - users remain registered to HQ - SIP trunk is handled by local MS - 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 Functionning mode: - SIP trunk is handled by SBA (not GW part) or SBS - Nominal ougoing and incoming traffic goes through SBA/SBS - In Case of GW-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 Functioning mode: - SIP trunk is handled by a-SBC part of the appliance (not MS part) - Nominal outgoing and incoming traffic goes through a-SBC - In case of GW-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	GW 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) - 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

3 Lync 2013 Configuration Checklist

Menu	Value
DNS requirements	
From the DNS interface: ✓ Start > Administrative Tools > DNS	FQDNs of each server (**DNS A** record)
From the DNS interface: ✓ Start > Administrative Tools > DNS	FQDNs of both nominal and backup aSBC on each site (**DNS A** record)
From the DNS interface: ✓ 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: ✓ 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: ✓ 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: ✓ ip helper-address *"IP@ of the DHCP Server"*
From the Microsoft Lync Server Management Shell interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	**Following command** has to **be typed**: ✓ Set-CsRegistrarConfiguration **–EnableDHCPServer $True**
From the DHCP interface: ✓ Start > Administrative Tools > DHCP > "select a scope" > Scope Options	DHCP Option **006 DNS Servers** has to **be activated**
From the DHCP interface: ✓ 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 *(*) DHCP Options **120 / 43** have to **be configured** (only if required by the type of endpoints deployed)*
From command prompt from the DHCP server: ✓ Start > Run… > cmd	**Following command** has to **be typed***: ✓ \\<FileShare>\DHCPUtil.exe -SipServer *"SipServer"* -WebServer *"WebServer"* –RunConfigScript *(*) DHCP Options **120 / 43** have to **be configured** (only if required by the type of endpoints deployed)*
From the DHCP interface: ✓ Start > Administrative Tools > DHCP > "select a scope" > Scope Options	DHCP Option **042 NTP Servers** has to **be activated*** *(*) only if required by the type of endpoints deployed*
AD requirements	
From the AD interface: ✓ 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: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder ✓ Lync Server 2013 > "select a Central	**TCP listening port** has to be set to **5060**

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

Menu	Value
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Route	4 **Routes** have to **be created** for 2 Standalone Mediation Servers*: ✓ **from Standalone Mediation Server 1a to nominal aSBC (=FQDN of the nominal aSBC from the Mediation Server 1a)** ✓ **from Standalone Mediation Server 1b to backup aSBC (=FQDN of the backup aSBC from the Mediation Server 1b)** ✓ **from Standalone Mediation Server 2a to nominal aSBC (=FQDN of the nominal aSBC from the Mediation Server 2a)** ✓ **from Standalone Mediation Server 2b to backup aSBC (=FQDN of the backup aSBC from the Mediation Server 2b)** A **gateway (=FQDN of the nominal aSBC from the Mediation Server 1a)** has to **be associated to First Route** A **gateway (=FQDN of the backup aSBC from the Mediation Server 1b)** has to **be associated to Second Route** A **gateway (=FQDN of the nominal aSBC from the Mediation Server 2a)** has to **be associated to Third Route** A **gateway (=FQDN of the backup aSBC from the Mediation Server 2b)** has to **be associated to Fourth Route** A **PSTN Usage** has to **be associated to each Route** *(*) 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: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder ✓ 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: ✓ **to nominal aSBC (=FQDN of the nominal aSBC)** ✓ **to backup aSBC (=FQDN of the backup aSBC)** **Check** that **2 Trunks** were created while creating PSTN gateways **Listening port** has to be set to **5060** for each PSTN gateways **SIP transport protocol** has to be set to **TCP** for each PSTN gateways
From the Microsoft Lync Server Topology Builder interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Topology Builder ✓ Lync Server 2013 > "select a Branch Sites" > Mediation pools	A **Mediation pools** has to **be configured** for the Standalone Mediation Server: ✓ **One single computer pool (=FQDN of the Mediation Server)** 2 **PSTN Gateways** have to **be associated** to the Standalone Mediation Server: ✓ **FQDN of the nominal aSBC** ✓ **FQDN of the backup aSBC** **Use all configured IPv4 IP addresses** has to **be checked**: **Listening port** has to be set to **5060**
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Dial Plan	A **Site dial plan** has to **be created for each Remote site with a Standalone Mediation Server** A **New Normalization Rule** for extension numbers has to **be associated**: ✓ **Pattern to match** has to **be edited** ✓ **Translation rule** has to **be edited** ✓ **Internal extension** has to **be checked** **Normalization Rule** for extension numbers has to **be moved up** before the **existent Normalization Rule** for Prefix All
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Voice Policy	An **User policy** has to **be created for each Remote site with a Standalone Mediation Server** **Enable call park** has to **be checked** **Enable PSTN reroute** has to **be unchecked** **A PSTN Usage** has to **be associated to each User policy**
From the Microsoft Lync Server Control Panel	The **specific voice policy** has to **be assigned to each RS (with a Standalone**

Menu	Value
interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Users > "select an user of Remote Site with a Standalone Mediation Server"	**Mediation Server) user**
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Route	2 **Routes** have to **be created for each Remote site with a Standalone Mediation Server** : ✓ **to nominal aSBC** ✓ **to backup aSBC** A **gateway (=FQDN of nominal aSBC)** has to **be associated to First Route** A **gateway (=FQDN of backup aSBC)** has to **be associated to Second Route** A **PSTN Usage** has to **be associated to each Route**
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Trunk Configuration	A **Site trunk** has to **be created for each Remote site with a Standalone Mediation Server** **Enable refer support** has to **be unchecked** **Encryption support level** has to be set to **Optional** 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: ✓ 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**: ✓ Set-CsTrunkConfiguration –Identity *"Site"* **–RTCPActiveCalls $False** ✓ Set-CsTrunkConfiguration –Identity *"Site"* **–RTCPCallsOnHold $False**
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Route	A **PSTN Usage of Branch Sites** has to **be associated to each Route of Headquarter** Note that routes must be in the following order: 1) **Route of Branch Sites to nominal aSBC** 2) **Route of Branch Sites to backup aSBC** 3) **Route of Headquarter to nominal aSBC** 4) **Route of Headquarter to backup aSBC**
Users Configuration	
From the AD interface: ✓ Start > Administrative Tools > Active Directory Users and Computers ✓ New > User	**User information** (the user logon name) has to **be filled**
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Users > Enable users > Add… > Find	Each user has to **be assigned to a pool** **Format <SAMAccountName>@<*SIP domain*>** has to **be selected** **Telephony** has to be set to **Enterprise Voice** An ***E164 telephone number*** format followed by **an extension number** has to be entered in the **line URI**
Routing mechanisms for Microsoft Lync Server 2013	
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Dial Plan	A **Site dial plan** has to **be created for each site** A **New Normalization Rule** for extension numbers has to **be associated**: ✓ **Pattern to match** has to **be edited** ✓ **Translation rule** has to **be edited** ✓ **Internal extension** has to **be checked** **Normalization Rule** for extension numbers has to **be moved up** before the **existent Normalization Rule** for Prefix All

Menu	Value
	(*) Site dial plan for a site Headquarter includes its Remote Sites without MGW
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Voice Policy	A **Site policy** has to **be created for each site*** **Enable call park** has to **be checked** **Enable PSTN reroute** has to **be unchecked** **A PSTN Usage** has to **be associated to each Site policy** *(*) Site policy for a site Headquarter includes its Remote Sites without MGW*
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Route	2 **Routes** have to **be created for each site*** : ✓ **to nominal aSBC** ✓ **to backup aSBC** A **gateway (=FQDN of nominal aSBC)** has to **be associated to First Route** A **gateway (=FQDN of backup aSBC)** has to **be associated to Second Route** A **PSTN Usage** has to **be associated to each Route** *(*) Routes for a site Headquarter includes its Remote Sites without MGW*
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Trunk Configuration	A **Site trunk** has to **be created for each site*** **Enable refer support** has to **be unchecked** **Enable forward call history** has to **be checked** **Encryption support level** has to be set to **Optional** A **Translation Rule** (to remove digit "+" for outbound calls to BTIP SIP) has to **be associated to each Site trunk** *(*) Site trunk for a site Headquarter includes its Remote Sites without MGW*
From the Microsoft Lync Server Management Shell interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	**Following commands** have to **be typed for each site***: ✓ Set-CsTrunkConfiguration –Identity *"Site"* **–RTCPActiveCalls $False** ✓ Set-CsTrunkConfiguration –Identity *"Site"* **–RTCPCallsOnHold $False** *(*) A Site Headquarter includes its Remote Sites without MGW*
From the Microsoft Lync Server Management Shell interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	**Following command** has to **be typed**: ✓ Set-CsMediaConfiguration **–EncryptionLevel SupportEncryption**
Specific Normalization Rule	
Voice Mail Feature : From the Microsoft Lync Server Control Panel interface: Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Dial Plan	**A Normalization Rule** has to **be associated to each Site dial plan*** *(*) to be adapted according the client architecture*
Call Park Feature : From the Microsoft Lync Server Control Panel interface: Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel Voice Routing > Dial Plan	**A Normalization Rule** has to **be associated to each Site dial plan*** *(*) to be adapted according the client architecture*
Music On Hold	

Menu	Value
From the Microsoft Lync Server Management Shell interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell **Note:** The customized MoH is played **For Softphone Devices** The embedded firmware MoH is played **For Lync Phone Edition Devices**	**The global clientpolicy is used:** **Following commands** have to **be typed for Softphones** ✓ New-CsClientPolicy –Identity **global –**EnableClientOnHold **$True** –MusicOnHoldAudioFile *<FILE PATH>* **Note:** 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: https://exchangeserverIPaddress/ecp logon using administrator credential ✓ Select **Unified Messaging** ✓ 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: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	**Following command** has to **be typed** Set-UMservice –Identity *<ExchangeServer>* **–UMStartUpMode TLS**
From the Exchange Server Administration Url: https://exchangeserverIPaddress/ecp logon using administrator credential ✓ Select **Unified Messaging** ✓ 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: https://exchangeserverIPaddress/ecp logon using administrator credential ✓ Select **Unified Messaging** ✓ 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): ✓ C:\Program Files\Microsoft\Exchange Server\V15\Bin\MSExchangeUM	<add key="MinimumRtpPort" value="**49152**" /> <add key="MaximumRtpPort" value="**57500**" />
From the Exchange UM server (Local Group Policy Editor): ✓ Start > Run… > gpedit.msc	**Audio Policy-based QoS** is configured Source port: **49152:57500** Protocol: **TCP and UDP** DSCP: **46**
From the Front End Server: ✓ C:\Program Files\Common Files\Microosft Lync Server 2013\Support\OcsUmUtil.exe ✓ On the OcsUmUtil tool: ▪ Click **Load Data** ▪ 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:	An **User policy** has to **be created for each site with Analog Devices**

Menu	Value
✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Voice Policy	**Enable call park** has to **be checked** **Enable PSTN reroute** has to **be unchecked** An **Existent PSTN Usage** has to **be associated** by selecting it
From the Microsoft Lync Server Management Shell interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	**Following command** has to **be typed for each Analog Device :** ✓ New-CsAnalogDevice *"LineURI"* –DisplayName *"DisplayName"* –RegistrarPool *"RegistrarPool"* **–AnalogFax $False** –Gateway *"Gateway"* –OU *"OU"*
From the Microsoft Lync Server Management Shell interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	**Following command** has to **be typed for each Analog Device :** ✓ Set-CsAnalogDevice -Identity *"Identity"* –DisplayNumber *"DisplayNumber"* ✓ Set-CsAnalogDevice -Identity *"Identity"* –LineURI *"LineURI"* ✓ Grant-CsVoicePolicy -Identity *"Identity"* –PolicyName *"PolicyName"*
From the Sonus (NET) (UX 1000/2000 SBA)	
From the UX Web User interface: ✓ Settings Tab > Media > Media List	A **Media List** has to **be created**: Media List for Analog Devices: **Media Profiles List** has to match the **Voice Codec Profile G711 A-Law** ➢ *Digit Relay* **Digit (DTMF) Relay Type** has to be set to **RFC 2833** **Digit Relay Payload Type** has to be set to **101**
From the UX Web User interface: ✓ Settings Tab > CAS > CAS Signaling Profiles	A **FXS CAS Signaling Profiles** has to **be created**
From the UX Web User interface: ✓ Settings Tab > Signaling Groups	A **CAS Signaling Group** has to **be created**: CAS Signaling Group for Analog Devices connectivity: ➢ *CAS Protocol* **CAS Signaling Profile** has to match the **CAS Signaling Profile** for Analog Devices ➢ *Channels and Routing* **Channel Hunting** has to be set to **Own Number** **Tone Table** has to match the **Analog Device Tone Table** **Call Routing Table** has to match the **Analog Device Call Routing Table** for routing calls received from Analog Devices** ➢ *Assigned Channels* **Channel Phone Number** has to match the **Analog Device phone number** *(**) Please note that Call Routing Table must be added later (after specific Call Routing Tables configuration)*
From the UX Web User interface: ✓ Settings Tab > Transformation	A **Transformation Table** has to **be created**: Transformation Table for Lync to Analog Device calls: ➢ *Input Field* **Value** has to match **the Analog Device telephone number E.164 format** ➢ *Output Field* **Value** has to be set to **\1**
From the UX Web User interface: ✓ 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)

Menu	Value
	Call Routing Entry for Lync to Analog Device calls: ➢ *Route Details* **Number/Name Transformation Table** has to match the **Transformation Table for Lync to Analog Device calls** ➢ *Destination Information* **Destination Signaling Groups** has to match the **Signaling Group for Analog Device connectivity** ➢ *Media* **Media List** has to match the **Media List for Analog Device** A **Call Routing Table** has to **be created** for calls received from the Analog Devices Call Routing Entry Tenor to Lync calls: ➢ *Route Details* **Number/Name Transformation Table** has to match the **Transformation Table used to send a phone number without modification** ➢ *Destination Information* **Destination Signaling Groups** has to match the **Signaling Group for Lync connectivity** ➢ *Media* **Media List** has to match the **Media List for Analog Device** *(**) 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: ✓ Configuration Tab (full) >VoIP menu > TDM submenu > Select TDM Bus Settings	**PCM Law Select** has to be set to **A-Law** **TDM Bus Clock Source** has to be set to **Network**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > Media submenu > Select Voice Settings From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > Media submenu > Select Analog Settings	**CAS Transport Type** has to be set to **CASRFC2833Relay** Check that **Analog Settings** are **filled** with default value
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > Coders and Profiles submenu > Select Analog Coders	**Coder Name** has to be set to **G711 A-Law** **Packetization Time** has to be set to**20ms** **Payload Type** has to be set to **8**
From the AudioCodes Web User interface: ✓ 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: **Module** has to be set to **Module 2 FXS** **Channels** has to be set to the **Analog Device** port on the gateway **Phone Number** has to match the **Analog Device** phone number **Trunk Group ID** has to match the **Analog Device** Trunk Group ID **Tel Profile ID** has to match the **Tel Profile ID** if configured else the **default profile 0** has to be associated **Trunk Group ID** has to match the **Analog Device** Trunk Group ID

Menu	Value
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Trunk Group > Select Trunk Group Settings	**Channel Select Mode** has to be set to **By Dest Phone Number**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Manipulation > Select Dest Number IP -> Tel From the AudioCodes Web User interface: ✓ 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** **Source Trunk Group** has to match the **Analog Device Trunk Group** already created **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: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Routing > Select Tel to IP Routing From the AudioCodes Web User interface: ✓ 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** **Src IP Group ID** has to be set to **-1** **Src Trunk Group ID** has to match the **Analog Device Group ID** **Dest IP Group ID** has to match the **Lync Server Group ID** **IP toTel Routing Mode** has to be set to **Route Calls before manipulation** **Dest Phone Prefix** has to match the **Analog Device** phone number **Trunk Group ID** has to match the **Analog Device Trunk Group ID** **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	
From the Sonus (NET) (UX 1000/2000 SBA) with FXS ports	
From the UX Web User interface: ✓ Settings Tab > Signaling Groups	An **ISDN Signaling Group** has to **be created**: ISDN Signaling Group for E1/T1 connectivity: ➢ *Port and Protocol* **Port Name** has to be **selected** **Switch Variant** has to be set to **Euro ISDN** ➢ *Channels and Routing* **Tone Table** has to match the **Tone Table** if configured else the **Default Tone Table** has to be selected **Call Routing Table** has to match the **E1/T1 Call Routing Table** for routing calls received from E1/T1 access** *(**) Please note that Call Routing Table must be added later (after specific Call Routing Tables configuration)*
From the UX Web User interface: ✓ Settings Tab > Transformation	Transformation Table for T2 to Lync calls A **Transformation Table** has to **be created**: Transformation Entry for T2 to Lync calls (Called): ➢ *Input Field* **Type** has to be set to **Called Address/Number** **Value** has to match the **T2 number** ➢ *Output Field* **Type** has to be set to **Called Address/Number**

Menu	Value
	Value has to match the **E.164 Lync number** Transformation Entry for T2 to Lync calls (Calling): ➢ *Input Field* **Type** has to be set to **Calling Address/Number** **Value** has to be **filled** ➢ *Output Field* **Type** has to be set to **Calling Address/Number** **Value** has to be **filled**
From the UX Web User interface: ✓ Settings Tab > Transformation	**Transformation Table for Lync to T2 calls** A **Transformation Table** has to **be created**: Transformation Entry for Lync to T2 calls (Called): ➢ *Input Field* **Type** has to be set to **Called Address/Number** **Value** has to be **filled** ➢ *Output Field* **Type** has to be set to **Called Address/Number** **Value** has to be **filled** Transformation Entry for Lync to T2 calls (Calling): ➢ *Input Field* **Type** has to be set to **Calling Address/Number** **Value** has to be **filled** ➢ *Output Field* **Type** has to be set to **Calling Address/Number** **Value** has to be **filled**
From the UX Web User interface: ✓ Settings Tab > Call Routing Table	**Call Routing Table for Lync to T2 calls** 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) Call Routing Entry for Lync to T2 calls: ➢ *Route Details* **Number/Name Transformation Table** has to match the **Transformation Table for Lync to T2 calls** ➢ *Destination Information* **Destination Signaling Groups** has to match the **Signaling Group for E1/T1 connectivity** ➢ *Media* **Media List** has to match the **Media List without crypto** **Call Routing Table for T2 to Lync calls** A **Call Routing Table** has to **be created** for calls received from E1/T1 access Call Routing Entry for T2 to Lync calls: ➢ *Route Details* **Number/Name Transformation Table** has to match the **Transformation Table T2 to Lync calls**

Menu	Value
	➢ *Destination Information* **Destination Signaling Groups** has to match the **Signaling Group for Lync connectivity** ➢ *Media* **Media List** has to match the **Media List without crypto** *(**) 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: ✓ Configuration Tab (full) >VoIP menu > PSTN submenu > Select Trunk Settings	**Protocol Type** has to be set to **E1 Euro ISDN** **Line Code** has to be set to**HDB3** **Framing Method** has to be set to **E1 FRAMING MFF CRC4 EXT**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Trunk Group > Select Trunk Group From the AudioCodes Web User interface: ✓ 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: **Module** has to be set to **Module 1 PRI** **Channels** has to be set to **T2 line number of channels** **Phone Number** has to match the **T2** phone number **Trunk Group ID** has to match the **T2** Trunk Group ID **Tel Profile ID** has to match the **Tel Profile ID** if configured else the **default profile 0** has to be associated **Trunk Group ID** has to match the **T2 Trunk Group ID** **Channel Select Mode** has to be set to **Cyclic Ascending**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > Control Network submenu > Select Proxy Set Table	A **Proxy Set Table** has to be created with the following parameters: **Proxy Set ID** has to be **filled** **Proxy Address** has to match the **SBA FQDN** **Transport Type** has to be set to **TLS** **Enable Proxy Keep Alive** has to be set to **Using Options**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > Control Network submenu > Select IP Group Table	An **IP Group Table** has to be created with the following parameters: **Index** has to be **filled** **Type** has to be set to **Server** **Proxy Set ID** has to match the **SBA proxy Set ID** already created
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Manipulation > Select Dest Number IP -> Tel From the AudioCodes Web User interface: ✓ 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 **Source IP Address** has to match the **SBA IP Address** **Stripped Digits from Left** has to be **filled** **Prefix to Add** has to be **filled** **Source Trunk Group** has to match the **T2 Trunk Group** already created **Destination Prefix** has to match the **T2 Line number** **Stripped Digits from Left** has to be **filled** **Prefix to add** has to match the corresponding Lync device on **E.164 format number**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu >	**Tel to IP Routing Mode** has to be set to **Route Calls after manipulation** **Src IP Group ID** has to be set to **-1**

Menu	Value
Routing > Select Tel to IP Routing	**Src Trunk Group ID** has to match the **T2 Group ID**
From the AudioCodes Web User interface: ✓ Configuration Tab (full) >VoIP menu > GW and IP to IP submenu > Routing > Select IP to Tel Routing	**IP toTel Routing Mode** has to be set to **Route Calls before manipulation** **Source IP Address** has to match the **Gateway IP Address** **Trunk Group ID** has to match the **T2 Trunk Group ID** **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: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ 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: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Routing > Trunk Configuration	A specific **translation Rule** has to **be associated to each Site trunk** *(*) to be adapted to the client architecture* *(**) first priority before translation rule removing the « + » digit*
Call Park feature	
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel ✓ Voice Features	A Number range has to be created **for each Site** *(*) to be adapted to the client architecture*
CALL ADMISSION CONTROL	
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel Network Configuration > Global	Edit Global Setting –Global Check **Enable call admission control**
From the Microsoft Lync Server Control Panel interface: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Control Panel Network Configuration > Bandwidth Policy	Create Bandwidth Policy for **CAC "from site to WAN"** **New** *"name"* Audio limit: **according to site sizing** Audio session limit: **100** Create Bandwidth Policy for **CAC "from Edge to WAN"** **New** *"name"* Audio limit: **according to site sizing** Audio session limit: **9999999999** Create Bandwidth Policy for **CAC "from site to SIP Trunk"** **New** *"name"* Audio limit: **according to site sizing** Audio session limit: **97** Create Bandwidth Policy for **CAC "0"** **New** *"name"* Audio limit: **0** Audio session limit: **40**

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

Menu	Value
From the Microsoft Lync Management Shell interface:: ✓ Start > All Programs > Microsoft Lync Server 2013 > Lync Server Management Shell	Enable client media port range: **Set-CsConferencingConfiguration –ClientMediaPortRangeEnabled $true –ClientAudioPort 50060 –ClientAudioPortRange 48**
Configuring Clients ports range for VVX	
✓ Using VVX Web UI	Navigate through the VVX Web Interface: http:<VVX_IP_Address> Go to Settings tab > Network menu > RTP Configure the Port Range Start to: 50060
✓ Using VVX configuration file (.cfg)	Configure the following line in the VVX configuration file : `tcpIpApp.port.rtp.mediaPortRangeStart="50060"` 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.

4 Skype for Business 2015 Configuration Checklist

Menu	Value
Skype for Business Configuration (Topology Builder)	
On the Topology builder interface: ✓ Central Site > skype for business 2015 > **Mediation Pools**, right click and Edit properties	**Enable TCP** port has to be **checked** **Listening port** has to be set to **5060** for each Mediation Server in skype for Business topology
On the Topology builder interface: ✓ Central Site > Skype for Business 2015 > Shared components > Trunks, right click edit properties	**FQDN** of **nominal aSBC** for BT/BTIP traffic Specify **nominal aSBC BT/BTIP trunk name** Listening port for IP/PSTN gateway**: 5060** SIP Transport protocol: **TCP** Associated Mediation Server: **Mediation Server FQDN** Associated Mediation Server port: **5060**
On the Topology builder interface: ✓ Central Site > Skype for Business 2015 > Shared components > Trunks, right click edit properties	**FQDN** of **backup aSBC** for BT/BTIP traffic Specify **backup aSBC BT/BTIP trunk name** Listening port for IP/PSTN gateway**: 5060** SIP Transport protocol: **TCP** Associated Mediation Server: **Mediation Server FQDN** Associated Mediation Server port: **5060**
Skype for Business Configuration (Control Panel)	
Dial Plan On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Dial Plan	Type: **Dial Plan** type Name: **Dial Plan** name
Voice Policy On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Voice Policy	Name: **Voice Policy** name Enable call park: **Checked** Enable PSTN reroute: **Unchecked**
PSTN usage On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Voice Policy	New PSTN Usage record Name: **BT/BTIP PSTN Usage name**
Routes (aSBC nominal route) On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Voice Policy	Edit PSTN Usage record Associated routes → New Name: **aSBC nominal Route name** Associated Trunks → Add **Select** corresponding **aSBC nominal Trunk** from drop down list
Routes (aSBC backup route) On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Voice Policy	Edit PSTN Usage record Associated routes → New Name: **aSBC backup Route name** Associated Trunks → Add **Select** corresponding **aSBC backup Trunk** from drop down list
Trunk configuration On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Trunk configuration	New Name: **BT/BTIP Trunk name** Encryption support level : **Optional** Refer support : **None**

Menu	Value
	Enable forward call History : **Checked**
Trunk configuration (SFB PowerShell) On the Skype for Business PowerShell Interface: ✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPActiveCalls $False*** ✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPCallsOnHold $False***	**-Site:** The name of the site

Configuration Checklist in case of Sonus 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 sonus SBC BT/BTIP configuration	
PSTN usage On the Skype for Server Control Panel Interface: ✓ Voice Routing > Voice Policy	New sonus SBC BT/BTIP PSTN Usage record Name: sonus SBC **BT/BTIP PSTN Usage name**
Route (sonus SBC BT/BTIP) On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Voice Policy	Edit PSTN Usage record Associated routes → New Name: **sonus SBC for BT/BTIP route name** Associated Trunks → Add **Select** corresponding **sonus SBC Trunk** from drop down list
Trunk configuration On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Trunk configuration	New Name: **sonus SBC for BT/BTIP Trunk name** Encryption support level : **Optional** Refer support : **None** Enable forward call History : **Checked**
Trunk configuration (SFB PowerShell) On the Skype for Business PowerShell Interface: ✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPActiveCalls $False*** ✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPCallsOnHold $False***	**-Site:** The name of the remote site
Sonus SBC BT/BTIP configuration	
SIP Profile	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Profile > Default SIP Profile	Session Timer: Session Timer: **Disabled** Header Customization: UA Header: **Sonus SBC** Calling Info Source: **RFC Standard** Options Tags: 100rel: **Supported** Update: **Supported** SDP Customization: Send Number of Channels: **True**

Menu	Value
	Connection Info In Media Section: **True** Digit Transmission Preference: **RFC 2833/Voice**
Media	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Media > Media System Configuration	Port Range: Start Port: **16384** Number of Port pairs: **600** Echo Canceller Type Option: **Standard** Echo Cancel NLP Option: **Mild** Send STUN Packets: **Enabled** Music On Hold: Music on Hold Source: **File**
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Media > Media Profiles	Default G711a: Codec: **G711 A-law** Payload Size: **20 ms** Default G711µ: Codec: **G711 µ-law** Payload Size: **20 ms**
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Media > Media List	Default Media List: Media Profiles List: **G711a** **G711µ** Crypto Profile ID: **None** Media DSCP: **46** RTCP Mode: **RTCP** Dead Call Detection: **Disabled** Silence Suppression: **Disabled**
Secondary interface (only for RS SBA)	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Node Interfaces > Logical Interfaces > Ethernet 1 IP	Configure Secondary Interface: **Enabled** Secondary Address: **IP address of the secondary interface of the Sonus gateway (dedicated for BT/BTIP traffic)** 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 On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Server Tables > Create SIP Server	Host: **SBA or MS Pool IP address** Port: **5060** Protocol: TCP Monitor: **SIP Options**
From/To BT/BTIP-SBA or From/To MS Pool –BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Server Tables > Create SIP Server	1st Entry: ACME aSBC nominal Host: **ACME aSBC nominal IP address** Port: **5060** Protocol: TCP Monitor: SIP Options 2nd Entry: ACME aSBC backup Host: ACME aSBC backup IP address Port: 5060 Protocol: TCP Monitor: SIP Options
Transformation Rules	

Menu	Value
SBA to BT/BTIP or MS Pool to BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number Output Field Value: depend on transformation need Called Entry: Input Field Type: **Called Address/Number** Input Field Value: depend on transformation need Output Field Type: Called Address/Number Output Field Value: depend on transformation need
BT/BTIP to SBA or BT/BTIP to SBA On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number Output Field Value: depend on transformation need Called Entry: Input Field Type: **Called Address/Number** Input Field Value: must normalize received number on Skype for Business E.164 number format Output Field Type: Called Address/Number Output Field Value: depend on transformation need
Call Routing Tables	
From SBA or From MS Pool On the Sonus SBC gateway WebUi Interface: ✓ Settings >Call Routing Table > Create	**SBA to BT/TIP** or **MS Pool to BT/TIP** entry: Description: **SBA to BT/BTIP or MS pool to BT/BTIP** Route Priority: **1** Number/Name Transformation Table: SBA to BT/BTIP or MS Pool to BT/BTIP Destination Signalling Group: (SIP) From/To BT/TIP-SBA or From/To BT/TIP-SBA Media Transcoding: Enabled (If licenced)
From BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >Call Routing Table > Create	**BT/TIP to SBA** or **BT/TIP to MS Pool** entry: Description: **BT/BTIP to SBA or BT/BTIP to MS Pool** Route Priority: **1** Number/Name Transformation Table: BT/BTIP to SBA or BT/BTIP to MS Pool Destination Signalling Group: **(SIP) From/To SBA-BT/BTIP or From/To MS Pool-BT/BTIP** Media Transcoding: **Enabled** (If licenced)
Signaling Groups	
(SIP) From/To SBA – BT/BTIP or From/To MS Pool – BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >Signaling Group > SIP Signaling Group	Description: **SIP From/To SBA – BT/BTIP or From/To MS Pool – BT/BTIP** Call Routing Table: **From SBA or From MS Pool** SIP Server Table: From/To SBA –BT/BTIP or MS Pool –BT/BTIP Signalling/Media Source IP :Sonus BT/BTIP

Menu	Value
	interface IP address Listen Ports:5060 /TCP Federated IP/FQDN: SBA or MS Pool FQDN
(SIP) From/To BT/BTIP-SBA or From/To BT/BTIP-MS Pool On the Sonus SBC gateway WebUi Interface: ✓ Settings >Signaling Group > SIP Signaling Group	Description: **SIP From/To BT/BTIP-SBA or From/To BT/BTIP-MS Pool** Call Routing Table: **From BT/BTIP** SIP Server Table: From/To BT/BTIP -SBA or From/To BT/BTIP-MS Pool Signalling/Media Source IP: Sonus BT/BTIP interface IP address Listen Ports:5060 /TCP Federated IP/FQDN: ACME aSBC nominal IP address ACME aSBC backup IP address
From/To SFB <-> Offnet routing E1/T1 traffic (only for RS SBA)	
System Companding Law	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >System > System companding law	Companding law: **A-Law**
SIP Server Table	
From/To SBA –PSTN On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Server Tables > Create SIP Server	Host: **SBA IP** Port: example **5060 (must be the same as defined on Skype for Business topology builder)** Protocol: TCP Monitor: **SIP Options** Note: If using same protocol and port as BT/BTIP the same SIP Server table can be used
Transformation Rules	
SBA to PSTN On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number Output Field Value: depend on transformation need Called Entry: Input Field Type: **Called Address/Number** Input Field Value: depend on transformation need Output Field Type: Called Address/Number Output Field Value: depend on transformation need
PSTN to SBA On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number Output Field Value: depend on transformation need Called Entry: Input Field Type: **Called Address/Number** Input Field Value: must normalize received number on Skype for Business E.164 number format

Menu	Value
	Output Field Type: Called Address/Number Output Field Value: depend on transformation need
Call Routing Tables	
From SBA On the Sonus SBC gateway WebUi Interface: ✓ Settings >Call Routing Table > Create	SBA to PSTN entry: Description: **SBA to PSTN** Route Priority: **1** Number/Name Transformation Table: SBA to PSTN Destination Signalling Group: (ISDN) From/To PSTN-SBA Media Transcoding: Enabled (If licenced)
From PSTN On the Sonus SBC gateway WebUi Interface: ✓ Settings >Call Routing Table > Create	PSTN to SBA entry: Description: **PSTN to SBA** Route Priority: **1** Number/Name Transformation Table: PSTN to SBA Destination Signalling Group: **(SIP) From/To SBA-PSTN** Media Transcoding: **Enabled** (If licenced)
Signaling Groups	
(SIP) From/To SBA – PSTN On the Sonus SBC gateway WebUi Interface: ✓ Settings >Signaling Group > SIP Signaling Group	Description: **SIP From/To SBA – PSTN** Call Routing Table: **From SBA** SIP Server Table: From/To SBA –PSTN Signalling/Media Source IP :Sonus E1/analog interface IP address Listen Ports:5060 /TCP Federated IP/FQDN: SBA IP address
(ISDN) PSTN On the Sonus SBC gateway WebUi Interface: ✓ Settings >Signaling Group > Signaling Group > ISDN Signaling Group	Description: **ISDN PSTN** Switch variant: **Euro ISDN** Call Routing Table: **From PSTN**
From/To SFB <-> Offnet routing Analog Devices traffic	
SIP Server Table	
From/To SBA –Analog Device On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Server Tables > Create SIP Server	Host: **SBA FQDN/IP address** Port: example **5060 (must be the same as defined on Skype for Business topology builder)** Protocol: TCP Monitor: **SIP Options** 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 On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number

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

Menu	Value
Skype for Business– RS GW BT/BTIP configuration	
PSTN usage On the Skype for Server Control Panel Interface: ✓ Voice Routing > Voice Policy	New sonus SBC BT/BTIP PSTN Usage record Name: sonus Gateway **BT/BTIP PSTN Usage name**
Route (sonus SBC BT/BTIP) On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Voice Policy	Edit PSTN Usage record Associated routes → New Name: **BT/BTIP Sonus GW route name** Associated Trunks → Add **Select** corresponding **sonus GW Trunk** from drop down list
Trunk configuration On the Skype for Business Server Control Panel Interface: ✓ Voice Routing > Trunk configuration	New Name: **sonus SBC for BT/BTIP Trunk name** Encryption support level : **Optional** Refer support : **None** Enable forward call History : **Checked** Enable media bypass : **Checked**
Trunk configuration (SFB PowerShell) On the Skype for Business PowerShell Interface: ✓ ***Set-CsTrunkConfiguration* –Identity *<Site>* –RTCPActiveCalls *$False*** ✓ ***Set-CsTrunkConfiguration* –Identity *<Site>* –RTCPCallsOnHold *$False***	**-Site:** The name of the site
Sonus GW BT/BTIP configuration	
SIP Profile	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Profile > Default SIP Profile	Session Timer: Session Timer: **Disabled** Header Customization: UA Header: **Sonus SBC** Calling Info Source: **RFC Standard** Options Tags: 100rel: **Supported** Update: **Supported** SDP Customization: Send Number of Channels: **True** Connection Info In Media Section: **True** Digit Transmission Preference: **RFC 2833/Voice**
Media	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Media > Media System Configuration	Port Range: Start Port: **16384** Number of Port pairs: **600** Echo Canceller Type Option: **Standard** Echo Cancel NLP Option: **Mild** Send STUN Packets: **Enabled** Music On Hold: Music on Hold Source: **File**
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Media > Media Profiles	Default G711a: Codec: **G711 A-law** Payload Size: **20 ms** Default G711µ:

Menu	Value
	Codec: **G711 µ-law** Payload Size: **20 ms**
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Media > Media List	Default Media List: Media Profiles List: **G711a** **G711µ** Crypto Profile ID: **None** Media DSCP: **46** RTCP Mode: **RTCP** Dead Call Detection: **Disabled** Silence Suppression: **Disabled**
TLS Profile	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Security > TLS Profiles	Create TLS Profile: TLS Protocol: **TLS 1.2 Only** Mutual Authentication: **Enabled** Allow Weak Cipher: **Disable** Handshake Inactivity Timeout: **10** The Client Cipher List is **automatically updated** to display only the ciphers supported for the selected TLS version Validate Server FQDN: **Disabled** Validate Client FQDN: **Disabled**
Secondary interface	
On the Sonus SBC gateway WebUi Interface: ✓ Settings >Node Interfaces > Logical Interfaces > Ethernet 1 IP	Configure Secondary Interface: **Disabled** Primary address dedicated for BT/BTIP traffic
From/To SFB <-> Offnet routing BT/BTIP traffic	
SIP Server Table	
From/To MS Pool –BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Server Tables > Create SIP Server	Host: **MS Pools FQDN/IP address** Port: **5067** Protocol: TLS TLS Profile: Select the TLS Profile created above Monitor: **SIP Options**
From/To BT/BTIP-MS Pool On the Sonus SBC gateway WebUi Interface: ✓ Settings >SIP > SIP Server Tables > Create SIP Server	1st Entry: ACME aSBC nominal Host: **ACME aSBC nominal IP address** Port: **5060** Protocol: TCP Monitor: SIP Options 2nd Entry: ACME aSBC backup Host: ACME aSBC backup IP address Port: 5060 Protocol: TCP Monitor: SIP Options
Transformation Rules	
MS Pool to BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number Output Field Value: depend on transformation need Called Entry:

Menu	Value
	Input Field Type: **Called Address/Number** Input Field Value: depend on transformation need Output Field Type: Called Address/Number Output Field Value: depend on transformation need
BT/BTIP to MS Pool On the Sonus SBC gateway WebUi Interface: ✓ Settings >Transformation > New Transformation Table > New Transformation Entry	Calling Entry: Input Field Type: **Calling Address/Number** Input Field Value: depend on transformation need Output Field Type: Calling Address/Number Output Field Value: depend on transformation need Called Entry: Input Field Type: **Called Address/Number** Input Field Value: must normalize received number on Skype for Business E.164 number format Output Field Type: Called Address/Number Output Field Value: depend on transformation need
Call Routing Tables	
From MS Pool On the Sonus SBC gateway WebUi Interface: ✓ Settings >Call Routing Table > Create	MS Pool to BT/TIP entry: Description: **MS Pool to BT/BTIP** Route Priority: **1** Number/Name Transformation Table: MS Pool to BT/BTIP Destination Signalling Group: (SIP) From/To BT/TIP-MS Pool Media Transcoding: Enabled (If licenced) Media List: Select the Media List created above
From BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >Call Routing Table > Create	BT/TIP to MS Pool entry: Description: **BT/BTIP to MS Pool** Route Priority: **1** Number/Name Transformation Table: BT/BTIP to MS Pool Destination Signalling Group: **(SIP) From/To MS Pool-BT/BTIP** Media Transcoding: Enabled (If licenced) Media List: Select the **Media List created** above
Signaling Groups	
(SIP) From/To MS Pool – BT/BTIP On the Sonus SBC gateway WebUi Interface: ✓ Settings >Signaling Group > SIP Signaling Group	Description: **SIP From/To MS Pool – BT/BTIP** Call Routing Table: **From MS Pool** No. of Channels: **60 (Default)** SIP Server Table: From/To MS Pool –BT/BTIP Signalling/Media Source IP :Sonus BT/BTIP interface IP address Listen Ports:5067 /TLS TLS Profile: Select the TLS Profile created above Federated IP/FQDN: MS Pools IP/FQDN

Menu	Value
(SIP) From/To BT/BTIP-MS Pool On the Sonus SBC gateway WebUi Interface: ✓ Settings >Signaling Group > SIP Signaling Group	Description: **SIP Froom/To BT/BTIP-MS Pool** Call Routing Table: **From BT/BTIP** No. of Channels: **60 (Default)** SIP Server Table: From/To BT/BTIP –MS Pool Signalling/Media Source IP :Sonus BT/BTIP interface IP address Listen Ports:5060 /TCP Federated IP/FQDN: ACME aSBC nominal IP address 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: ✓ Branch Site > SfB Server > **Mediation Pools**, right click and Edit properties	Listening ports **TLS: 5067 – 5067** **Note:** **When both VISIT and B2G offer:** Listening ports TLS must be**: 5069**
On the Topology builder interface: ✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for BT/BTIP Then click Next to define **root trunk**	**FQDN** of dedicated gateway **for BT/BTIP traffic** Specify **BT trunk name** Listening port for IP/PSTN gateway**: 5067** SIP Transport protocol: **TLS** Associated Mediation Server: **Mediation Pool FQDN** Associated Mediation Server port: **5067** **Note:** **When both VISIT and B2G offer:** Listening ports TLS must be**: 5069**
Skype for Business Configuration in case of RS-SBA (Topology Builder)	
On the Topology builder interface: ✓ Branch Site > SfB Server > **Mediation Pools**, right click and Edit properties	Listening ports **TCP: 5060 – 5060**
On the Topology builder interface: ✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for BT/BTIP Then click Next to define **root trunk**	**FQDN** of dedicated gateway **for BT/BTIP traffic** Specify **BT trunk name** Listening port for IP/PSTN gateway**: 5060** SIP Transport protocol: **TCP** Associated Mediation Server: **SBA FQDN** Associated Mediation Server port: **5060**
On the Topology builder interface: ✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for E1/analog **PSTN & Analog Trunk:** ✓ Branch Site > SfB Server > Shared Components > Trunks, right click and **New Trunk**	**FQDN** of dedicated gateway **for E1/Analog traffic** Specify **PSTN&Analog trunk name** Listening port for IP/PSTN gateway**: 5060** SIP Transport protocol: **TCP** Associated Mediation Server: **SBA FQDN** Associated Mediation Server port: **5060**
Skype for Business Configuration in case of HQ with GW aboard (Topology Builder)	

Menu	Value
On the Topology builder interface: ✓ Branch Site > SfB Server > **Mediation Pools**, right click and Edit properties	Listening ports **TCP: 5060 – 5060**
On the Topology builder interface: ✓ Branch Site > SfB Server > Shared components > PSTN gateways, right click and **New IP/PSTN Gateway** dedicated for BT/BTIP Then click Next to define **root trunk**	**FQDN** of dedicated gateway **for BT/BTIP traffic** Specify **BT trunk name** Listening port for IP/PSTN gateway**: 5060** SIP Transport protocol: **TCP** Associated Mediation Server: **MS Pool FQDN** Associated Mediation Server port: **5060**
AudioCodes Mediant 800/1000 E-SBC configuration	
TLS Context	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) ✓ System > TLS Context	Links Tab TLS Context Certificate TLS Context Trusted Certificates
Media	
Voice Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) ✓ Configuration >VoIP > Media > Voice Settings	Silence Suppression: **Disable** DTMF Transport Type: **RFC 2833 Relay DTMF**
Media Security	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration >VoIP > Media > Media Security	Media security: **Enable**
RTP / RTCP Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration >VoIP > Media > RTP / RTCP Settings	RTP Base UDP Port: **16400**
Application Enabling	
Application Enabling	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration >VoIP > Application Enabling > Application Enabling	SBC Application: **Enable**
Coders and Profiles	
Coders	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration >VoIP > Coders and Profiles > Coders	Coders Table Coder Name : **G711A-law** Packetization time : **20** Rate : **64** Payloed Type : **8** Silence Suppression : **Disabled** Coder Name : **G711U-law** Packetization time : **20** Rate : **64** Payload Type : **0** Silence Suppression : **Disabled**
Coders Group Settings	

Menu	Value
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration >VoIP > Coders and Profiles > Coders Group Settings	Coders Group ID Coder Name : **G711A-law** Packetization time : **20** Rate : **64** Payloed Type : **8** Silence Suppression : **Disabled** Coder Name : **G711U-law** Packetization time : **20** Rate : **64** Payload Type : **0** Silence Suppression : **Disabled**
IP Profile Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration >VoIP > Coders and Profiles > IP Profile Settings	SBA or SfB IP Profile ID (GW tab) Early Media : **Enable** Hold : **Enable** (SBC Media tab) Extension Coders : **Coders Group** Allowed Audio Coders : **Coders Group** Allowed Coders Mode : **Restriction and Preference** BTIP IP Profile ID (GW tab) Early Media : **Enable** Hold : **Enable** (SBC Media tab) Extension Coders : **Coders Group** Allowed Audio Coders : **Coders Group** Allowed Coders Mode : **Restriction and Preference**
VoIP Network	
Media Realm Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > Media Realm Table	Skype Media Realm (SBA or SfB) Name : **MRm for Skype** IPv4 Interface Name : **Mediant IPv4 Interface** Port Range Start : **16900** Number of Media Session Legs : **50** Port Range End : **Filled automatically** Default Media Realm : **Yes** BTIP Media Realm Name : **MRm for BTIP** IPv4 Interface Name : **Mediant IPv4 Interface** Port Range Start : **16400** Number of Media Session Legs : **50** Port Range End : **Filled automatically** Default Media Realm : **No** **This range is used to accept incoming traffic from SBC in case of BTIP incoming calls, the defined range respects the OBS infra recommandations**

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

Menu	Value
	SRD : **DefaultSRD** Network Interface : **Mediant IPv4 Interface** SBC IPv4 SIP Interface : **SIP Interface for BTIP Traffic** Proxy Load Balancing Method : **Round Robin** Proxy Keep-Alive Time : **60** Proxy Keep-Alive : **Using OPTIONS** (Proxy Address Table) 2 Entries : **FQDN or @IP of aSBC ACME:5060 TCP**
IP Group Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > IP Group Table	IP Group Table for Skype traffic (SBA or SfB) Name : **IP Profile for Skype Traffic** Type : **Server** Proxy Set : **Proxy Set for Skype Traffic** IP Profile : **IP Profile for Skype Traffic** Media Realm : **Media Realm for Skype traffic** IP Group Table for BTIP traffic Name : **IP Profile for BTIP Traffic** Type : **Server** Proxy Set : **Proxy Set for BTIP Traffic** IP Profile : **IP Profile for BTIP Traffic** Media Realm : **Media Realm for BTIP traffic**
SIP Definitions	
General Parameters	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > SIP Definitions > General Parameters	PRACK Mode : **Supported** Channel Select Mode : **Cyclic Ascending** Enable Early Media : **Enable**
SBC	
Allowed Audio Coders Group	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > SBC > Allowed Audio Coders Group	Allowed Audio Coders Group ID Coder Name 1 : **G711A-Law** Coder Name 2 : **G711U-Law**
IP-to-IP Routing Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > SBC > IP-to-IP Routing Table	SIP Options rule Name : **SIP Options** Alternative Route Options: **Route Row** Source IP Group : **Any** Request Type : **OPTIONS** Destination Type : **Dest Address** Destination IP Group : **None** Destination SIP Interface : **None** Destination Address : **internal** Skype to BTIP rule Name : **Skype to BTIP** Alternative Route Options: **Route Row** Source IP Group : **Skype IP Group**

Menu	Value
	Request Type : **All** Destination Type : **IP Group** Destination IP Group : **BTIP IP Group** Destination SIP Interface : **BTIP SIP Interface** BTIP to Skype rule Name : **BTIP to Skype** Alternative Route Options: **Route Row** Source IP Group : **BTIP IP Group** Request Type : **All** Destination Type : **IP Group** Destination IP Group : **BTIP IP Group** 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: (Advanced mode) Configuration > VoIP > Gateway > Trunk Group	Configure Group Index Module : **PRI** From/To Trunk : **1** Channels : **1-31** Phone Number : **Phone number used for the Trunk** Trunk Group ID : **Trunk Group ID associated**
Trunk Group Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Gateway > Trunk Group Settings	Add Trunk Group Settings Name : **E1 PSTN** Trunk Group ID : **Trunk Group ID associated** Channel Selected Mode : **Cyclic Descending** Registration Mode : **Don't Register**
Trunk Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > PSTN > Trunk Settings	Protocol Type : **E1 EURO ISDN** Line Code : **HDB3** Framing Method : **Extend super Frame**
VoIP Network Configuration	
Media Realm Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > Media Realm Table	Can be the same as Skype Media Realm Name : **MRm for Skype** IPv4 Interface Name : **Mediant IPv4 Interface** Port Range Start : **16900** Number of Media Session Legs : **50** Port Range End : **Filled automatically** Default Media Realm : **Yes**
SRD Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > SRD Table	Same as Skype SRD Table Name : **DefaultSRD**
SIP Interface Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > SIP	SIP Interface Table Name : **SIPInterface_PSTN** SRD : **DefaultSRD**

Menu	Value
Interface Table	Network Interface : **Mediant IPv4 Interface for E1/Analog** Application Type : **GW** TCP Port : **5060**
Proxy Set Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > Proxy Set Table	Proxy Set Table for PSTN traffic Name : **ProxySet for PSTN Traffic** SRD : **DefaultSRD** Network Interface : **Mediant IPv4 Interface for E1/Analog** SBC IPv4 SIP Interface : **SIP Interface for PSTN Traffic** Proxy Load Balancing Method : **Round Robin** Proxy Keep-Alive Time : **60** Proxy Keep-Alive : **Using OPTIONS** (Proxy Address Table) 1 Entry : **FQDN or @IP of SBA:5060 TCP**
IP Group Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > IP Group Table	IP Group Table for Skype traffic Name : **IP Profile for PSTN Traffic** Type : **Server** Proxy Set : **Proxy Set for PSTN Traffic** IP Profile : **IP Profile for Skype Traffic** Media Realm : **Media Realm for Skype Traffic**
Routing	
General Parameters	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Gateway > Routing > General Parameters	Enable Alt Routing Tel to IP : **Enable**
IP To Trunk Group Routing	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Gateway > Routing > IP To Trunk Group Routing	Skype To PSTN rule Name : **Skype To PSTN** Source IP Group : **Skype IP Group** Source SIP Interface : **PSTN SIP Interface** Trunk Group ID : **PSTN Trunk Group ID** Destination Type : **Trunk Group**
TEL To IP	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Gateway > Routing > TEL To IP	PSTN To Skype rule Name : **PSTN To Skype** Source Trunk Group ID : **PSTN Trunk Group ID** Destination IP Group : **Skype IP Group** SIP Interface : **PSTN SIP Interface** IP Profile : **Skype IP Profile**
Gateway for Analog calls (Annex 2)	
Trunk Group	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Gateway > Trunk Group	Configure Group Index Module : **FXS** Channels : **1** Phone Number : **Analog number in e164 format** Trunk Group ID : **Trunk Group ID for Analog**

Menu	Value
Trunk Group Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Gateway > Trunk Group Settings	Add Trunk Group Settings Name : **Analog** Trunk Group ID : **Trunk Group ID for Analog** Channel Selected Mode : **By Dest Phone Number** Registration Mode : **Don't Register**
Analog Settings	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > Media > Analog Settings	Analog Metering Type : **12 Khz Sinusoidal bursts** FXS Coefficient Type : **Europe**
VoIP Network Configuration	
Media Realm Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > Media Realm Table	Can be the same as Skype Media Realm Name : **MRm for Skype** IPv4 Interface Name : **Mediant IPv4 Interface** Port Range Start : **16900** Number of Media Session Legs : **50** Port Range End : **Filled automatically** Default Media Realm : **Yes**
SRD Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > SRD Table	Same as Skype SRD Table Name : **DefaultSRD**
SIP Interface Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > SIP Interface Table	SIP Interface Table Name : **SIPInterface_Analog** SRD : **DefaultSRD** Network Interface : **Mediant IPv4 Interface for E1/Analog** Application Type : **GW** TCP Port : **5060**
Proxy Set Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > Proxy Set Table	Proxy Set Table for Analog traffic Name : **ProxySet for Analog Traffic** SRD : **DefaultSRD** Network Interface : **Mediant IPv4 Interface for E1/Analog** SBC IPv4 SIP Interface : **SIP Interface for Analog Traffic** Proxy Load Balancing Method : **Round Robin** Proxy Keep-Alive Time : **60** Proxy Keep-Alive : **Using OPTIONS** (Proxy Address Table) 1 Entries : **FQDN or @IP of SBA:5060 TCP**
IP Group Table	
On the AudioCodes Mediant WebUi Interface: (Advanced mode) Configuration > VoIP > VoIP Network > IP Group Table	IP Group Table for Skype traffic Name : **IP Profile for Analog Traffic** Type : **Server** Proxy Set : **Proxy Set for Analog Traffic** IP Profile : **IP Profile for Skype Traffic** Media Realm : **Media Realm for Skype Traffic**

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

CAC Configuration Checklist

CAC Configuration	
Enable CAC	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***Set-CsNetworkConfiguration -EnableBandwidthPolicyCheck*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration >Global	SFB PowerShell **EnableBandwidthPolicyCheck** parameter has to be set to **1** SFB Control Panel **Enable call admission control** parameter has to be **checked**
Media bypass configuration (In case of RS SBA and/or RS Default)	
SFB PowerShell	SFB PowerShell ✓ ***AlwaysByPass*** parameter has to be

Menu	Value
On the Skype for Business PowerShell Interface: ✓ ***$a=*** ***New-CsNetworkMediaBypassConfiguration -alwaysByPass $false -Enabled $false*** ✓ ***Set-CsNetworkConfiguration –MediaBypassSettings $a*** **SFB Control Panel** On the Skype for Business control panel interface: Network Configuration >Global	set to **false** ✓ ***Enable*** parameter has to be set to **false** SFB Control Panel ✓ **Enable media bypass** parameter must not be **checked**
Media bypass configuration (In case of RS GW or a mix of RS GW, RS SBA and RS Default)	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***$a=*** ***New-CsNetworkMediaBypassConfiguration -alwaysByPass $ false -Enabled $true*** ✓ ***Set-CsNetworkConfiguration –MediaBypassSettings $a*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration >Global	SFB PowerShell ✓ ***AlwaysByPass*** parameter has to be set to **false** ✓ ***Enable*** parameter has to be set to **true** SFB Control Panel ✓ **Enable media bypass** parameter has to be **checked** ✓ **Choose** "Use sites and region configuration"
Media bypass Trunk Configuration (Only in case of RS-GW)	
SFB Control Panel On the Skype for Business Control panel interface ✓ Voice Routing > Trunk Configuration And then select the RS-GW Trunk to edit Trunk configuration	SFB Control Panel ✓ **Enable media bypass** parameter has to be **checked**
Trunk configuration (SFB PowerShell) On the Skype for Business PowerShell Interface: ✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPActiveCalls $False*** ✓ ***Set-CsTrunkConfiguration –Identity <Site> –RTCPCallsOnHold $False***	**-Site:** The name of the site
Network Region	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkRegion –Identity <XdsIdentity> -CentralSite <Central_Site> –AudioAlternatePath $False -Description "All Locations"*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration >Global	SFB PowerShell **-Identity:** The name of the network region **-Central site:** The name of the central site as defined on SFB topology builder SFB Control Panel **Identity:** The name of the network region **Central site:** The name of the central site as defined on SFB topology builder **Audio alternate path**: Recommended to disable
Bandwidth Policy profiles	
CAC Onnet – Network sites and Network Region CAC	

Menu	Value
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –Description "*Descr Name*" -AudioBWLimit *<AudiototalBW>* -AudioBWSessionLimit *<AudiosessionBW>* -VideoBWLimit *<VideototalBW>* -VideoBWSessionLimit *<VideoSessionBW>*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration >Bandwidth Policy	SFB PowerShell **-Identity:** The name of the bandwidth region (eg: **CAC_basse**) **-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy **-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **100** **-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) SFB Control Panel **Identity:** The name of the bandwidth region (eg: **CAC_basse**) **AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy **AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to 100 **VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) on SFB topology builder
CAC SIP Trunk – Inter site CAC	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –Description "*Descr Name*" -AudioBWLimit *<AudiototalBW>* -AudioBWSessionLimit *<AudiosessionBW>* -VideoBWLimit *<VideototalBW>* -VideoBWSessionLimit *<VideoSessionBW>*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration >Bandwidth Policy	SFB PowerShell **-Identity:** The name of the bandwidth region (eg: **CAC_SIPTrunk**) **-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy **-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **97** **-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) SFB Control Panel **Identity:** The name of the bandwidth region (eg: CAC_SIPTrunk) **AudioBWLimit:** The total bandwidth allowed for BT/BTIP calls on network sites associated to this BW profile policy **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 **97** **VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) on SFB topology builder
CAC Zero – BT/BTIP network site to Network region CAC	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –Description “*Descr Name*” -AudioBWLimit *<AudiototalBW>* -AudioBWSessionLimit *<AudiosessionBW>* -VideoBWLimit *<VideototalBW>* -VideoBWSessionLimit *<VideoSessionBW>*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration >Bandwidth Policy	SFB PowerShell **-Identity:** The name of the bandwidth region (eg: **CAC_Zero**) **-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy → parameter has to be set to **0** **-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **40** **-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) SFB Control Panel **Identity:** The name of the bandwidth region (eg: **CAC_Zero**) **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** **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 **40** **VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) on SFB topology builder
CAC Edge – Edge network site to Network region CAC	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkBandwidthPolicyProfile* -Identity *<BWname>* –Description “*Descr Name*” -AudioBWLimit *<AudiototalBW>* -AudioBWSessionLimit *<AudiosessionBW>* -VideoBWLimit *<VideototalBW>* -VideoBWSessionLimit *<VideoSessionBW>***	SFB PowerShell **-Identity:** The name of the bandwidth region (eg: **CAC_Edge**) **-AudioBWLimit:** The total bandwidth allowed for calls on network sites associated to this BW profile policy → parameter has to be set to **9999999999** **-AudioBWSession Limit:** The session bandwidth allowed for one call on network site associated to this BW profile policy → has to be set to **100**

Menu	Value
SFB Control Panel On the Skype for Business control panel interface: ✓ Network Configuration >Bandwidth Policy	**-VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **-VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) SFB Control Panel **Identity:** The name of the bandwidth region (eg: **CAC_Edge**) **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** **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** **VideoBWLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) **VideoBWSessionLimit:** Not applied with BT/BTIP (used for onnet calls refer to B2G documentation) on SFB topology builder
Network Sites	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkSite-NetworkSIteID <NSname> –Description "Descr Name" -NetworkRegionID <NRname> -BWPolicyProfileID <BWPname>*** **SFB Control Panel** On the Skype for Business control panel interface: ✓ Network Configuration > Site	SFB PowerShell **-NetworkSiteID:** The name of the network site **-Description:** Optional **-NetworkRegionID:** Select the network region to associate to created network site **-BWPolicyProfileID:** Select the bandwidth profile policy to associate to created network site SFB Control Panel **-NetworkSiteID:** The name of the network site **-Description:** Optional **-NetworkRegionID:** Select the network region to associate to created network site **-BWPolicyProfileID:** Select the bandwidth profile policy to associate to created network site
Inter Site Policy	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkInterSitePolicy–Identity <NetworkInterSitename>-BWPolicyProfileID <SIPTRUNK_BWPname> -NetworkSiteID1 <NS1name>-NetworkSiteID2 <BTIP_NS_name>***	SFB PowerShell **-Identity:** The name of the network inter site policy **-BWPolicyProfileID:** Select the bandwidth profile policy to associate to created network inter site policy **-NetworkSiteID1:** parameter has to correspond to the network site 1 (SFB component) to associate to BTIP using inter site policy **-NetworkSiteID2:** parameter has to

Menu	Value
	correspond to the BT/BTIP network site name **WARNING:** NO Inter site for Remote site Gateway
Subnets	
SFB PowerShell On the Skype for Business PowerShell Interface: ✓ ***New-CsNetworkSubnet*-SubnetID <firstsubnetIPaddress>-MaskBits <maskwo/> -NetworkSiteID <associated NS_name>*** **SFB Control Panel** On the Skype for Business control panel interface: Network Configuration > Subnet	SFB PowerShell **-SubnetID:** The first IP address of the corresponding subnet **-MaskBits:** The subnet mask to associate to subnet to create without / (eg:**32**) **-NetworkSiteID:** Select the network site name from the drop down list to associate to this subnet (eg: **BTIP**) SFB Control Panel **-SubnetID:** The first IP address of the corresponding subnet **-MaskBits:** The subnet mask to associate to subnet to create without / (eg:**32**) **-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	
SFB PowerShell On the Skype for Business PowerShell Interface ***Set-CsConferencingConfiguration –ClientMediaPortRangeEnabled $true –ClientAudioPort 50060 –ClientAudioPortRange 48***	SFB PowerShell -**ClientMediaPortRangeEnable :** must be enabled in order to use the specific range -**ClientAudioPort :** corresponds to the first port used for audio -**ClientAudioPortRange :** corresponds to the audio range
Configuring Clients ports range for VVX	
✓ Using VVX Web UI : - Navigate through the VVX Web Interface: http:<VVX_IP_Address> - Go to Settings tab > Network menu > RTP - Configure the Port Range Start to: 50060	VVX WebUI
✓ Using VVX configuration file (.cfg) - Configure the following line in the VVX configuration file : tcpIpApp.port.rtp.mediaPortRangeStart="50060" - Import the new configuration file to the VVX using the WebUI or through the IIS server	VVX WebUI or IIS Server
Others Devices	
✓ Check that the audio range port respect the OBS recommendations The default audio range is: 50060-50107.	