Business Talk & BTIP for Alcatel-Lucent Enterprise OmniPCX Enterprise (OXE)

versions addressed in this guide : R101.x

Information included in this document is dedicated to customer equipment (IPBX, TOIP ecosystems) connection to Business Talk & BTIP service : it shall not be used for other goals or in another context.

Latest edition: 05/01/2026

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1. Goal of this document

The aim of this document is to list technical requirements to ensure the interoperability between Alcatel-Lucent Enterprise IPBX with Business Talk (International) or BTIP (France) service from Orange Business, hereafter so-called “service”.

2. Certified architectures

2.1. Introduction to architecture components and features

This document describes “only” the main supported architectures either strictly used by our customers or that are used as reference to add specific usages often required in enterprise context (specific ecosystems, redundancy, multi-codec and/or transcoding, recording…)

Concerning fax communications, Orange supports the following usage :

- fax servers connected to the IPBX* -and sharing same dial plan-, or as sperate ecosystems -and separate dial plan-
- analog fax machines, usually connected on specific gateways* (seen as IPBX ecosystem or not)

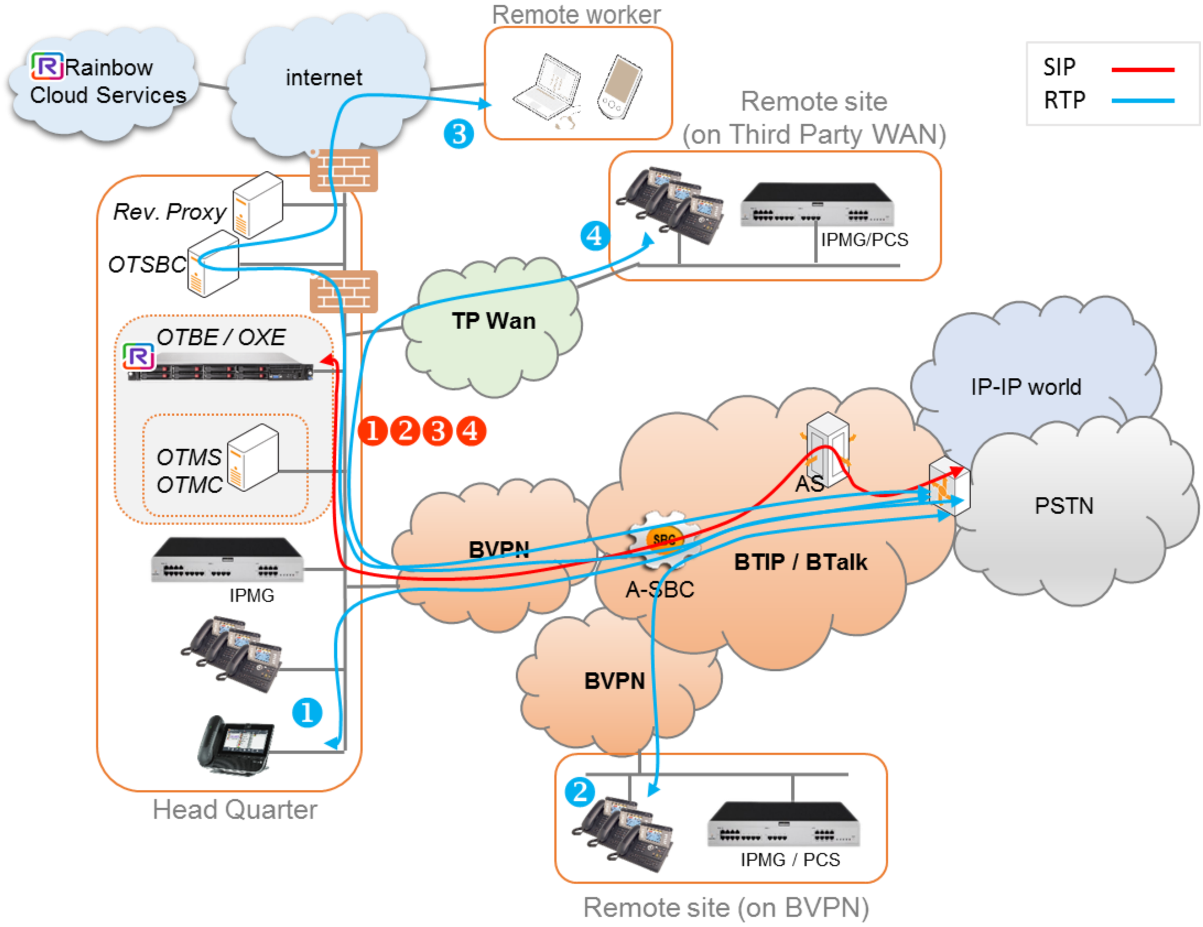
Fax flows are handled via T.38 transport only through BTIP and Business Talk.

Concerning the Quality of Service, Business VPN and BTIP/Btalk networks trust the DSCP (Differenciated Services Code Point) values sent by customer voice equipment. That’s why Orange strongly recommends to set the IPBX, IP phones and other voice applications with a DiffServ/TOS value** = 46 (or PHB value = EF) at least for media.

BTIP in French overseas departments are also supported. Dedicated aSBC pairs in Caribbean and Indian Ocean zones are deployed for local calls. For a trunking point of view, the mechanism is similar to “Business Talk French customers”, the IPBX must support international dial plans and route local calls to the dedicated local aSBC pair.

*cf fax servers and gateways listed in “Certified software and hardware versions section” and dedicated parameters in the “Configuration Checklist → SIP / SIP Ext Gateway”.
**cf QoS parameters in the “Configuration Checklist → IP / IP Quality of Service Category”. This value applies to IP phones, IP boards, IP domains and SIP messages from the OXE Communication Server.

2.2. SIP trunk on OXE over BVPN



Notes :

- in the diagram above, the SIP, proprietary and Rainbow internal flows are hidden.
  - ❶ call from/to head quarter
  - ❷ call from/to remote site (on Business VPN)
  - ❸ call from/to remote worker (on Internet)
  - ❹ call from/to remote site (on Third Party WAN)
- call flows will be the similar with or without OXE Call Server redundancy (duplicated or spatial)

In this architecture :

- all 'SIP trunking' signaling flows are carried by the OXE server and routed on the main BVPN connection.
- Media flows are direct between endpoints and the Business Talk/BTIP but IP routing differs from one site to another :
  - For the Head Quarter site, media flows are just routed on the main BVPN connection
  - For Remote sites on BVPN, media flows are just routed on the local BVPN connection (= **distributed architecture**),

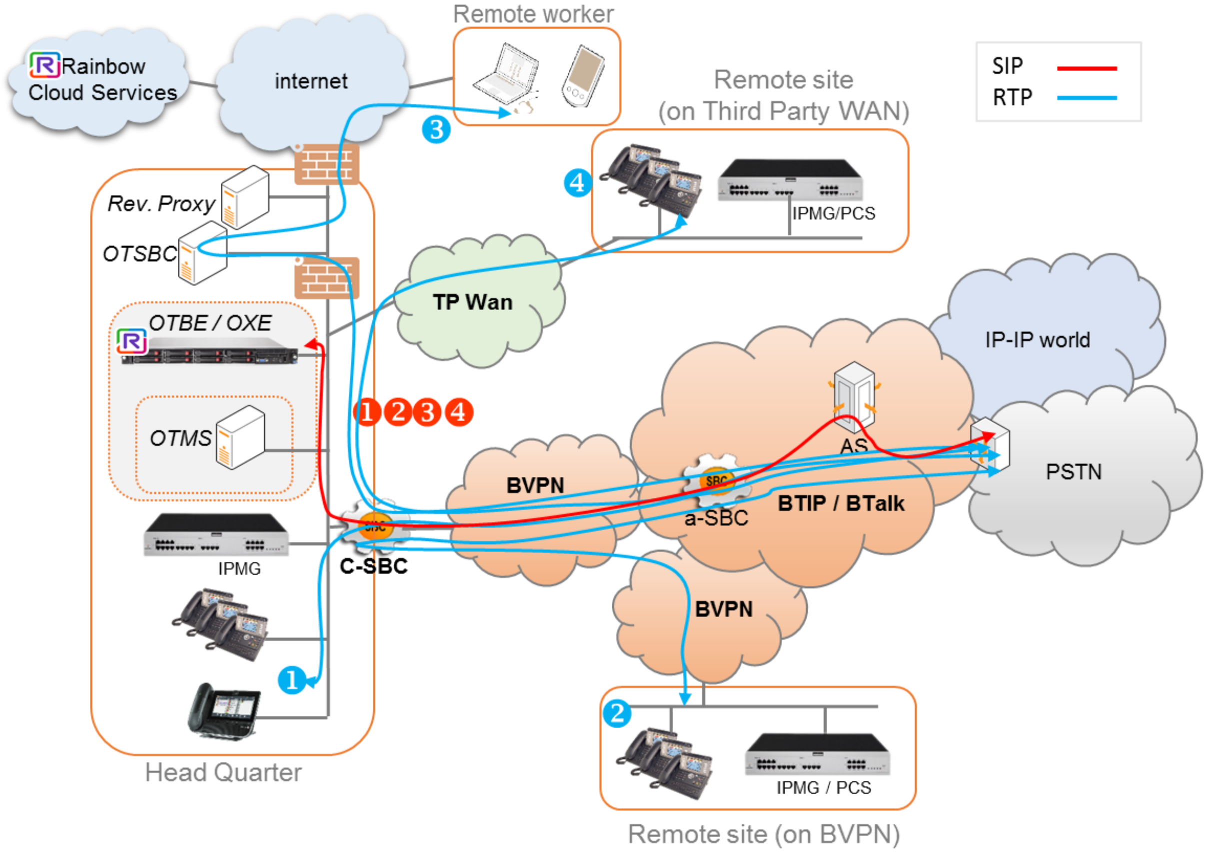
- For Remote sites on Third Party WAN, media flows are routed through the Head Quarter (but not through the IPBX) and use the main BVPN connection (= **centralized architecture**, cf sizing below).

Here below a table with a few sizing elements :

Call scenario	**nb of voice channels/media resources used**		
	IPBX	**WAN router***	**BTIP**
1 offnet call from/to the head quarter (HQ)	**1** in HQ	**1** in HQ	**1** in HQ
1 offnet call from/to a remote site (RS) on BVPN	0 in HQ **1** in RS	0 in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ **1** in RS	**1** in HQ BVPN **1** in HQ TPWan **1** in RS TPWan	0 in HQ **1** in RS
1 offnet call from/to a remote site **with put on hold**	**1** in HQ **1** in RS	**1** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site **after transfer/forward to BTIP**	0 in HQ **0** in RS	0 in HQ **0** in RS	0 in HQ **2** in RS
1 **forced onnet** call **from head quarter to a remote site** (= through Business Talk infrastructure)	**2** in HQ **2** in RS	**1** in HQ **1** in RS	**0** in HQ **0** in RS

*on the WAN router, 1 voice channel = 80Kb/s

2.3. SIP trunk on OTSBC over BVPN



Notes :

- in the diagram above, the SIP, proprietary and Rainbow internal flows are hidden :
  - ❶ call from/to head quarter
  - ❷ call from/to remote site (over Business VPN)
  - ❸ call from/to remote worker (over Internet)
  - ❹ call from/to remote site (over Third Party WAN)

In this architecture, both ‘SIP trunking’ and RTP media flows between endpoints and the Business Talk/BTIP are anchored by the enterprise SBC* :

- for the Head Quarter site, media flows are routed through the SBC and the BVPN access
- for Remote Sites either on BVPN or Third Party WAN, media flows transit **through the Head Quarter SBC** and use the central BVPN connection (= **centralized architecture**, cf sizing below).

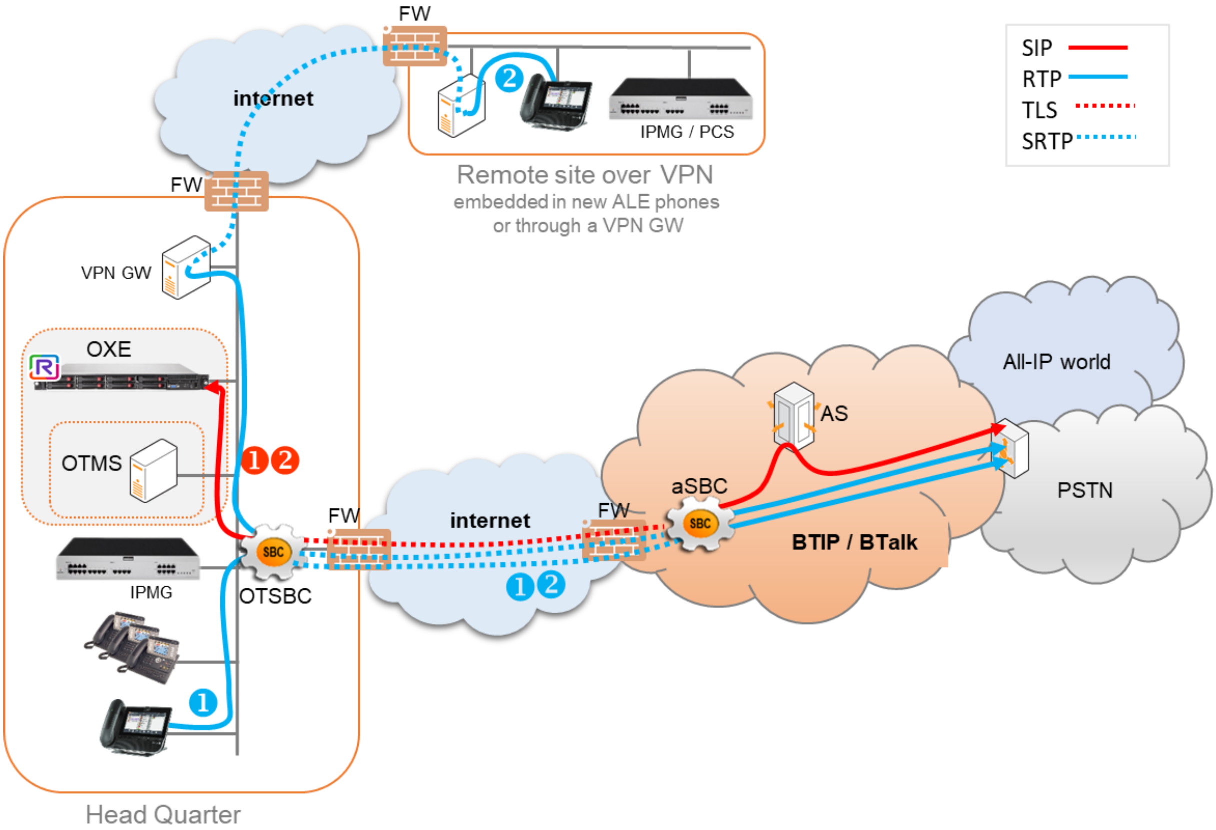
*AudioCodes Mediant (VE or appliance versions) and Alcatel OTSBC are standard, so don’t need any other specific implementation request.

Warning : with an enterprise SBC architecture, site access capacity has to be sized adequately on the Head Quarter. Here below a table with a few sizing elements :

Call scenario	**nb of voice channels/media resources used**		
	IPBX	**WAN router***	**BTIP**
1 offnet call from/to the head quarter (HQ)	**1** for HQ	**1** in HQ	**1** in HQ
1 offnet call from/to a remote site (RS) on BVPN	0 for HQ **1** for RS	**2** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site (RS) on TP Wan	0 in HQ **1** in RS	**1** in HQ BVPN **1** in HQ TPWan **1** in RS TPWan	0 in HQ **1** in RS
1 offnet call from/to a remote site **with put on hold**	**1** for HQ **1** for RS	**3** in HQ **1** in RS	0 in HQ **1** in RS
1 offnet call from/to a remote site **after transfer/forward to BTIP**	0 for HQ **0** for RS	0 in HQ*/3 in HQ** **0** in RS	0 in HQ **2** in RS
1 **forced onnet** call **from head quarter to a remote site** (= through Business Talk infrastructure)	**2** for HQ **2** for RS	**3** in HQ **1** in RS	**0** in HQ **0** in RS

*on the WAN router, 1 voice channel = 80Kb/s **if media release is activated on the cSBC ***if media release is not activated on the cSBC

2.4. SIP trunk on OTSBC over Internet



Notes : in the diagram above, the SIP, proprietary and Rainbow internal flows are hidden

❶ call from/to head quarter

❷ call from/to remote site over Internet (with the VPN client embedded in the ALE phones or with a VPN gateway)

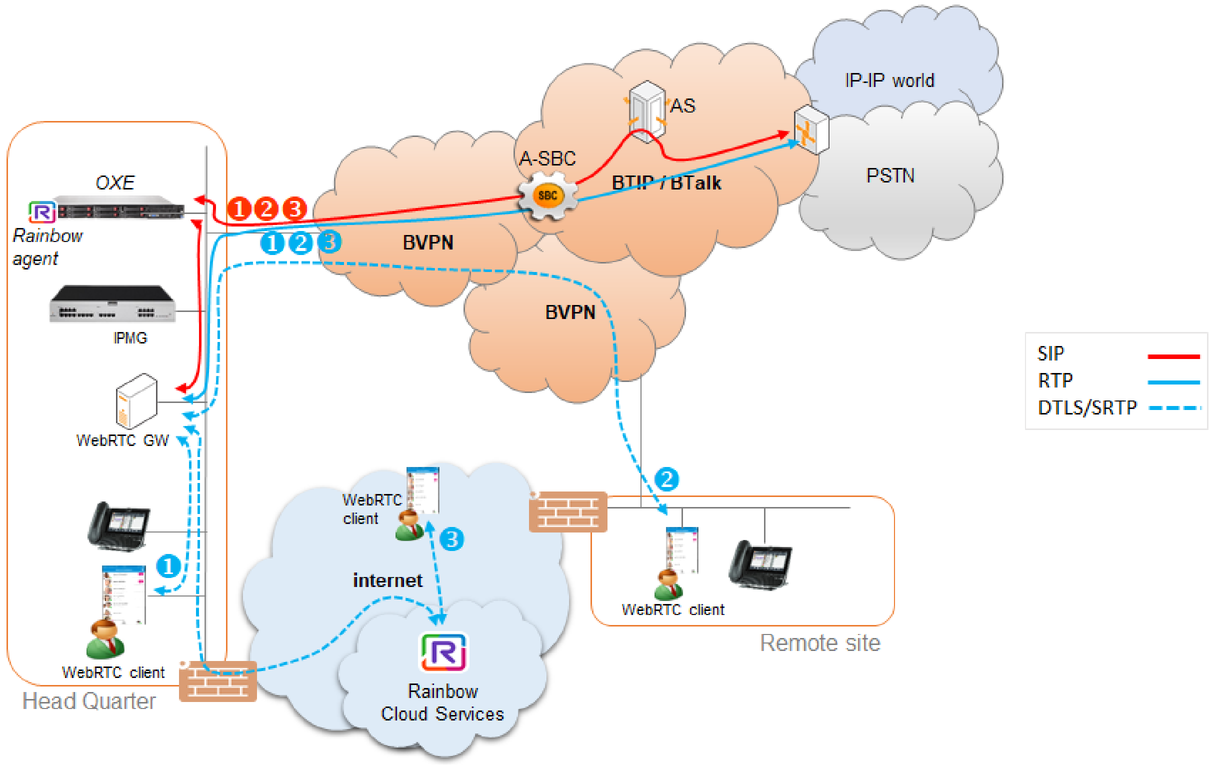
SIP TLS + Secured RTP: all SIP messages and media packets are encrypted on the public internet between Orange and the customer endpoints. This is the level of encryption recommended by default by Orange to ensure security & privacy. Refer to the dedicated configuration section chapter 6.4 for more details.

In this architecture, both ‘SIP trunking’ and RTP media flows between endpoints and the Business Talk/BTIP are anchored by the enterprise SBC* :

- for the Head Quarter site, media flows are routed through the SBC and the Internet access
- for Remote Sites either on Internet, media flows transit **through the Head Quarter SBC** and use the BTIP/BTalk connection (= **centralized architecture**).

* Alcatel OTSBC and AudioCodes Mediant (VE or appliance versions) are standard, so don’t need any other specific implementation request.

2.5. Architecture with Rainbow WebRTC gateway



Notes :

- in the diagram above, data flows (HTTPS/XMPP/Jingle/REST) between the clients/OXE/WebRTC GW and Rainbow services on the internet are hidden :
    - ❶ call from/to head quarter
    - ❷ call from/to remote site (over Business VPN)
    - ❸ call from/to remote worker (over Internet)

Since OXE R12.4 MD4, architectures with WebRTC media gateway support the direct RTP feature (with codec G711 only).

Prior to that version media flows to/from Business Talk/BTIP are anchored on an IPMG. In that case, IPMG resources have to be sized adequately on the Head Quarter. There isn't any impact on Business Talk/BTIP.

3. Parameters to be provided by customers to access to the service

IP addresses marked in red have to be indicated by the Customer, depending on Customer architecture scenario

Head Quarter (HQ) – **architecture without OTSBC**	**Level of Service**	**@IP used by service**	
Single Call Server	No call server redundancy	call server @IP	
Duplicated Call Server	Local call server redundancy	call server @IP (virtual)	
Spatial Redundancy **warning:** - Site access capacity to be sized adequately on the site carrying the 2nd call server - DNS server feature must be activated on both CS (OXE)	Site redundancy: 2 call servers (active/standby) hosted by 2 different physical sites	nominal call server @IP	backup call server @IP
Remote Site (RS) – **architecture without OTSBC**	**Level of Service**	**@IP used by service**	
Remote site without survivability	No survivability, no trunk redundancy	N/A	
PCS for one remote site	Local user survivability and SIP trunk redundancy for the remote site hosting the PCS in case of non-access to HQ	PCS @IP	
PCS for several remote sites **warning:** Site access capacity to be sized adequately on the site carrying the PCS	Local user survivability and SIP trunk redundancy for the remote site hosting the PCS in case of non-access to HQ	PCS @IP	

Head Quarter (HQ) – **architecture with OTSBC over BVPN**	**Level of Service**	**@IP used by service**	
1 Customer SBC	No redundancy	OTSBC @IP	
2 Customer SBC Nominal / Backup mode	- **Local redundancy**: both SBC are hosted on the same site OR - **Geographical redundancy** both SBC are hosted on 2 different sites	OTSBC1 @IP	OTSBC2 @IP
2 Customer SBC HA mode	- **Local redundancy:** both SBC are hosted on the same site OR - **Geographical redundancy** both SBC are hosted on 2 different sites **warning:** Link level 2 between SBC with max delay 50ms required for geo-redundancy	OTSBC VIP @IP	
Remote Site (RS) – **architecture with OTSBC over BVPN**	**Level of Service**	**@IP used by service**	
SBC on Head Quarter – all modes (see above)	No redundancy	N/A	

Head Quarter (HQ) – **architecture with OTSBC over Internet**	**Level of Service**	**@IP used by service**	
1 customer SBC	No redundancy	OTSBC public FQDN(1) DNS type A	
2 customer SBC Nominal / Backup mode (DNS resiliency model)	- **Local redundancy**: both SBC are hosted on the same site OR - **Geographical redundancy** both SBC are hosted on 2 different sites	OTSBC public FQDN(1) DNS type SRV	
2 customer SBC Nominal / Backup mode (SIP resiliency model)	- **Local redundancy**: both SBC are hosted on the same site OR - **Geographical redundancy** both SBC are hosted on 2 different sites	OTSBC1 public FQDN(1) DNS type A	OTSBC2 public FQDN(1) DNS type A
2 customer SBC HA mode (IP resiliency model)	- **Local redundancy:** both SBC are hosted on the same site OR - **Geographical redundancy** both SBC are hosted on 2 different sites **warning:** Link level 2 between SBC with max delay 50ms required for geo-redundancy	OTSBC public FQDN(1) DNS type A	
Remote Site (RS) – **architecture with OTSBC over Internet**	**Level of Service**	**@IP used by service**	
SBC on Head Quarter – all modes (see above)	No redundancy	N/A	

(1) FQDN is mandatory for BTIP over Internet (France), while public IP or FQDN is supported for Busines Talk over Internet (International)

Refer to the ‘Business Talk IP over Internet pre-requesites’ document provided by your sales team for more details about the encrypted architecture, the certificate management, the firewall rules, etc…

In the TLS context, the full chain Root and Intermediate Certificate (PEM format) must be transmitted to Orange

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4. Business Talk & BTIP certified versions

4.1. Global Release Policy

Orange supports the last 2 major IPBX versions and will ensure Business Talk and BTIP infrastructure evolutions will rightly interwork with the related architectures. Orange will assist customers running supported IPBX versions and facing issues.

Please refer to the latest Alcatel-Lucent MLV Release Policy Information and MLE Cross Compatibility documents for more details about the supported versions:

- 'MLV_ReleasePolicy_PhaseOutProduct_October_2025_Ed56' or more recent
- 'MLE_CrossCompatibility_October_2025_Ed48' or more recent

4.2. Alcatel-Lucent Entreprise IPBX and SBC

ALE IPBX & SBC – software versions

Reference product	Software version	Certification	Certified "Loads"
OmniPCX Enterprise OpenTouch Business Edition	**OXE R101**	✓	n3.320.9b (TR), n3.321.2a (MD1), n3.321.4 (MD2), n3.510.7a (MD3), n3.521.8a (MD4), n3.521.10a (MD5), n3.521.11a (MD6**),** n3.521.12 (MD7), n3.521.13 (MD8), n3.521.14a (MD9), **n3.521.14a (MD10)**
	OXE R101.1	✓	n4.205.19 (TR), n4.205.36a (MD1), n4.205.38 (MD2), **n4.513.13 (MD3)**
OpenTouch SBC or AudioCodes Mediant VE generic version	**OTSBC 7.4**	✓	7.40A.500.017, 7.40A.500.781, **F7.40A.600.203**

4.3. Alcatel-Lucent Entreprise endpoints and applications

ALE IPBX - endpoints and applications

Reference product		Software version	Certification	OXE version	Comments
Attendant	Visual Automated Attendant	-	Not tested		
	4059 EE	2.4.6 min	✓	R101/R101.1	
Voice Mail	4645	-	✓	R101/R101.1	
	OTMC	2.6 min	Not tested		End of Life
Mobility	Desk Sharing	-	Not tested		
	OTSBC	7.4	✓	R101/R101.1	Remote workers with NGInx reverse proxy
UC	Open Touch MS	2.6.1	Not tested		End of Life
	Open Touch Fax Center	9.0	✓	R101/R101.1	
	Rainbow / Cloud Connect	-	✓	R101/R101.1	call control only. No voice impact
	Rainbow WebRTC GW	3.0 min	✓	R101/R101.1	
Recording	OmniPCX Record	2.5 min	NA	R101/R101.1	No impact
Call Center	OTCC Standard	10.15 min	✓	R101/R101.1	CCAgent
	Business Contact	-	Not tested	R101/R101.1	Customer specific (ALE Professional Services)

ALE IPBX - endpoints and applications

Reference product		Software version	Certification	OXE version	Comments
Alcatel-Lucent endpoints	Pleiades Enterprise series Pleiades Essential series 80x8s/80x8/80x9 series	-	✓	R101/R101.1	Refer to ALE release policy
	IP-DECT xBS AP + handsets	-	Not tested		
	IP Desktop Softphone	13 min	✓	R101/R101.1	
	ALE softphone (ALES) for PC	2.5 min	✓	R101/R101.1	
	8135S Conference phone	2.0.10 min	Not tested		'SIP device' mode
Third-party endpoints & applications	AudioCodes GW MP11x	MP11x v07 ≧ 6.60.x	✓	R100.x	Voice gateway to be declared as 'SIP device'
	ISI-COM Interact	8.x	Not tested		Contact Center
	Conecteo Kiamo	8.x	Not tested		Contact Center
	Fortigate 30E/60F VPN Gateways	-	Not tested		for remote users over internet
	Others		On demand		
Fax (T.38)	Open Touch Fax Center	9.0	✓	R101/R101.1	New parameter impact from OXE R12.2 (see configuration checklist below)
	Analog fax on ALE IPMG (SLI-x, MIX-x, Z-x)	-	✓		
	Analog fax on Mediatrix 4102 or C7/S7 serie	Dgw ≧ 42.2.1669	✓		
	Analog fax on AudioCodes MP11x	MP11x v07 ≧ 6.60.x	On demand		

5. OXE SIP trunking configuration checklist

The checklist below presents all the **required** configuration parameters for interoperability between Business Talk/BTIP and IPBX OXE. New parameters since the previous edition are highlighted.

5.1. Common parameters

Menu	Value
MEDIA PARAMETERS	
Media Gateway > "select an instance MGw" Média Gateway > "sélectionner une instance MGW"	**Law** has to be set to **Default** **Loi de quantification** doit être configurée à **Défaut**
System > Timers Installation > Temporisations	**Timer 42** should be set to **5** **Timer 299** can be set up to 150 in case of DTMF detection issue only **Temporisation 42** devrait être configurée à **5** **Temporisation 299** peut être augmentée jusqu'à **150** en cas de problème de détection des DTMF uniquement
System > Other System Param. Installation > Autres param. Install.	**DTMF on Alert** has to be **False** **No End of Dialing** has to be set to **True** **DTMF on Alert** doit être configuré à **Non** **Pas de fin de numérotation** doit être configuré à **Oui**
System > Other System Param. > System Parameters > Law Installation > Autres param. Install. > Paramètres Système > Loi de quantification	**Law** has to be set to **A Law** **Loi de quantification** doit être configurée à **Loi A**
System > Other System Param. > Compression Parameters Installation > Autres param. Install. > Paramètres Compression	**Voice Activity Detect (Comp Bds)** has to be **False** **Voice Activity Detection on G711** has to be set to **False** **Compression Type** has to be set to **G729** **Suppression Silence (Cartes Comp.)** doit être configuré à **Non** **Suppression Silence sur G711** doit être configuré à **Non** **Type de compression** doit être configuré à **G729**
System > Other System Param. > SIP Parameters Installation > Autres param. Install. > Paramètres SIP	**Packetization times per codec** has to be set to **False** **Via Header Inbound Calls Routing** has to be set to **True** **Enhanced codec renegotiation** has to be set to **Network type** **SIP diversion info for incoming** has to be set to **True** **Packetization time par codec** doit être configure à **Non** **Entête Via pour appels entrants** doit être configure à **Oui** **Négociation optimisée des codecs** doit être configuré à **Type réseau** **Sip diversion pour appels entrants** doit être configuré à **Oui**
System > Other System Param. > Signaling String Installation > Autres param. Install. > Signalisation, para. non num	**Country Code** has to be configured with the **Country Code of site** **Préfixe international** doit être configuré avec le **code pays du site**

Menu	Value
System > Other System Param. > External Signaling Parameters Installation > Autres param. Install. > Paramètres Signalisation Externe	**NPD for external forward** has to be set to **a value different from -1** except in Multivendor / Hybrid or non-ABC architectures **NPD pour renvoi extérieur** doit être configuré à **une valeur différente de -1** sauf dans une architecture Multivendeur / Hybrid ou non-ABC
Users > TSC IP Users > “select a user” Usagers > Usagers TSC IP > “sélectionner un usager”	**Voice Coding Algorithm** has to be set to **Default** **Algorithme de codage** doit être configuré à **Défaut**
IP > IP Parameters IP > Paramètres IP	**Fast Start** has to be set to **True** **G711 VOIP Framing** has to be set to **20 ms** **G729 VOIP Framing** has to be set to **20 ms** **Round trip delay request** has to be set to **False** **Fast Start** doit être configuré à **Oui** **Framing VOIP pour G711** doit être configuré à **20 ms** **Framing VOIP pour G729** doit être configuré à **20 ms** **Délais round trip requis** doit être configuré à **Non**
IP > IP Quality Of Service COS IP > Catégorie Qualité Service IP	**TOS/Diffserv** has to be set to **46** *(default value)* SIP Diff. Service can be set to **40** *(default value)* **TOS/DiffServ** doit être configuré à **46** *(valeur par défaut)* **SIP Diff. Service** peut être configuré à **40** *(valeur par défaut)*
IP > IP Domain IP > Domain IP	**Intra domain bandwidth** has to be set to **High bandwidth** **Extra domain bandwidth** has to be set to **High bandwidth** **FAX/MODEM Intra domain call transp** has to be set to **Yes** **FAX/MODEM Extra domain call transp** has to be set to **Yes** **Bande-passante intra domaine** doit être configuré à **Large Bande** **Bande-passante extra domaine** doit être configuré à **Large Bande** **FAX/MODEM Appel Intra domain trans** doit être configuré à **Oui** **FAX/MODEM Appel Extra domain trans** doit être configuré à **Oui**
IP > IP Domain > IP Domain Address IP > Domain IP > Zone de domaine IP	IP addresses have to be set for: ▪ IP Address Low ▪ IP Address High ▪ IP NetMask Les adresses IP suivantes doivent être configurées : ▪ Adresse IP basse ▪ Adresse IP haute ▪ NetMask IP

If UDP lost is manage IP > IP Quality of Service COS > "select CoS QoS number 0" IP > IP Domain > "select an IP Domain" Shelf > Board > Ethernet Parameters > "select an INTIP or GD/GA board" Si UDP lost est géré IP > Catégorie de qualité de service IP > "sélectionner l'instance CoS QoS 0" IP > Domain IP > "sélectionner un Domaine IP" Alvéole > Carte-Interface > Paramètres Ethernet > "sélectionner une carte INTIP ou GD/GA"	**UDP Lost** should be set to **45s** **IP Quality of Service** has to be set to **0** **IP Quality of Service** has to be set to **0** **UDP Lost** devrait être configuré à **45s** **Qualité de service IP** doit être configurée à **0** **Qualité de service IP** doit être configurée à **0**
Call Allowance Control (CAC)	
IP > IP Domain > "Select an IP Domain" IP > Domain IP > "sélectionner un Domaine IP"	**Domain Max Voice Connection** has to be set to a limitation call number (–1 is the default value = no limitation) **Nb Max de connexions / domaine** doit être configuré avec un nombre limite d'appel (–1 est la valeur par défaut = aucune limitation)
Diversion parameter for Remote Extension users	
Applications > Remote Extension Parameter > Redirecting IE number available Applications > Paramètres Remote Extension > No de réacheminement EI dispo.	Redirecting IE number available has to be set to YES **No de réacheminement EI dispo.** doit être à **OUI**

ROUTE MECANISM ON OXE Off-net calls	
SIP > SIP Gateway SIP > Passerelle SIP	**Cac SIP-SIP** has to be set to **True** **Cac SIP-SIP** doit être configuré à **Oui** *Note: for BTalk, the session timers, dynamic payload and SDP in 180 are now managed in the SIP External gateways* *Note : pour BTIP les timers de sessions, de payload dynamique et de SDP dans les 180 sont maintenant gérés dans passeerelles SIP externes*
SIP > SIP Proxy SIP > Proxy	**SIP initial time-out** has to be set to **500** **Retransmission number for INVITE** has to be set to **4** **TCP when long messages** has to be set to **False** **Only authenticated incoming calls** has to be set to **True** **Tempo. initiale** doit être configurée à **500** **Retransmission number for INVITE** doit être configurée à **4** **TCP lors de longs messages** doit être configurée à **non** **Seulement appel arriv. Authentifié** doit être configuré à **oui**
Private SIP Trunk Group (for internal SIP gateway): Trunk Groups Faisceau privé pour passerelle SIP interne: Faisceaux	**Trunk Group Type** has to be set to **T2** **T2 Specification** has to be set to **SIP** **Q931 Signal variant** has to be set to **ABC-F** **Type faisceau** doit être configuré à **T2** **Spécificité T2** doit être configurée à **SIP** **Variante signalisation Q931** doit être configurée à **ABC-F**
Public SIP Trunk Group (for external SIP gateways): Trunk Groups Faisceaux publiques pour passerelles SIP externes: Faisceaux	**Trunk Group Type** has to be set to **T2** **T2 Specification** has to be set to **SIP** **Q931 Signal variant** has to be set to **ISDN all countries** **Type faisceau** doit être configuré à **T2** **Spécificité T2** doit être configurée à **SIP** **Variante signalisation Q931** doit être configurée à **RNIS tout pays**
Trunk Groups > *"select a SIP Trunk Group ID"* > Trunk Group Faisceaux > "*sélectionner le faisceau SIP*" > Faisceau	**Entity Number** has to match to **Entity of site** **Trunk COS** has to be set to **31** **IE External Forward** has to be set to **Diverting leg info** **No Entité** doit correspondre à l'**Entité du site** **Classe de service ARS** doit être configuré à **31** **Transfert Externe IE** doit être configuré à **Diverting leg info**
Trunk Groups > Trunk Group > Virtual accesses for SIP > "select a Trunk Group ID" Faisceaux > Faisceau > Accès Virtuel pour SIP > "sélectionner le faisceau SIP"	**Number of SIP Accesses** has to define between **2** (=60 simultaneous calls) and **32** (=960 simultaneous calls) **Nombre d'accès SIP** à définir entre **2** (=60 appels simultanés) et **32** (=960 appels simultanés)

Public numbering plan *(by default/par défaut):* Translator > External Numbering Plan > Numbering Plan Description (NPD) > "select a NPD identifier" Traducteur > Plan de numérotation externe > Description de plan de num. > "sélectionne un identificateur de description"	**Calling Numbering Plan ident.** has to be set to **NPI/TON ISDN international** **Install. number source** has to be set to **None used** **Default number source** has to be set to **None used** **Called DID identifier** has to be set to **0** **Identifiant plan de num appellant** doit être configuré à **Plan/Type num RNIS international** **Origine num. installation** doit être configuré à **Aucun** **Numéro par défaut** doit être configuré à **Aucun** **Numéro de SDA pour appelé** doit être configuré à **0**
Private numbering plan *(optional/en option):* Translator > External Numbering Plan > Numbering Plan Description (NPD) > "select a NPD identifier" Traducteur > Plan de numérotation externe > Description de plan de num. > "sélectionner un identificateur de description"	**Install. number source** has to be set to **None used** **Default number source** has to be set to **None used** **Origine num. installation** doit être configuré à **Aucun** **Numéro par défaut** doit être configuré à **Aucun**
Translator > External Numbering Plan > DID Numbering Translator Traducteur > Plan de numérotation externe > Traducteur numéro SDA	**DID num. transl. Identifier** has to be set to **0** **Numéro de traducteur SDA** doit être configuré à **0**
Translator > External Numbering Plan > DID Numbering Translator > DID Number Translator rules Traducteur > Plan de numérotation externe > Traducteur numéro SDA > Traducteur SDA : règles	**First External Number** has to be set to **CCNSN** (international number)**,** **First Internal Number** has to match the **private short number** (if private numbering used) **1er numéro extérieur** de la tranche doit avoir le format 33ZABPQMCDU **1er numéro intérieur** de la tranche doit avoir le **format de numéro privé** (si plan de num privé utilisé)
Trunk Groups > Trunk group NPD selector > "select a Trunk Group ID" Faisceaux > Sélecteur de Plan de Num Faisceau > "sélectionner le faisceau SIP"	**Public NPD ID** has to match the **NPD identifier** **Management Mode** has to be set to **Normal** **No Desc.Plan.Num public** doit correspondre à l'**identificateur de description** **Mode de gestion** doit être configuré à **Normal**
External Services > Trunk COS > "select a Trunk Group COS" Services extérieurs > Catégories de joncteurs > "sélectionner la classe de service ARS du faisceau"	**Overflow Timer on No Answer** should be set to **300** (default value). Could be reduced for remote sites without Media Gateway using the BTIP/BT DTO mechanism. **Timer T310** has to be set to a value **greater than 110** (>**11s)** to wait for the Ring Back Tone. For example: **200 (=20s)** **Débordement sur non-réponse** devrait être configuré à **300** (valeur par défaut). Cette valeur pourrait être réduite pour des sites distants sans Media Gateway et utilisant le mécanisme de DTO BTIP. **Tempo. T310** doit être configuré avec une valeur **plus grande que 110** (>**11s)** pour attendre le retour de sonnerie. Par exemple : **200 (=20s)**.

SIP > SIP Ext Gateway	Remote domain has to match IP address of the BTalk SBC (or OTSBC for BTalk over internet) PCS IP Address should to be set to 255.255.255.255 SIP Port Number has to be set to 5060 Transport Type has to be set to UDP […] Supervision timer has to be set to 380 Trunk group number has to match the public SIP trunk Gp […] RFC 3325 supported by the distant has to be set to True […] SDP in 18x has to be set to False Minimal authentication method has to be set to None […] Contact with IP address has to be set to True Dynamic Payload type for DTMF has to be set to 101 Outbound Calls 100 REL has to be set to Supported Incoming Calls 100 REL has to be set to Not Requested Gateway Type has to be set to Standard Type Re-Trans No. for REGISTER/OPTIONS has to be set to 4 P-Asserted-ID in Calling Number has to be set to False Trusted P-Asserted-ID header has to be set to True Trusted From header has to be set to False Diversion Info to provide via has to be set on Diversion Support Re-Invite without SDP has to be set to True Proxy identification on IP address has to be set to False Outbound calls only has to be set to False SDP relay on Ext. Call Fwd should be set to 180 only RFC 5009 supported/Outbound call has to be set to Mode1 Nonce Caching activation has to be set to False Fax Procedure Type has to be set to T38 only DNS SRV/Call retry on busy server has to be set to 0 Unattended Transfer for RSI has to be set to No Redirection functionality has to be set to No Attended Transfer has to be set to No Support Redirection Response has to be set to No OPTIONS required has to be set to YES Support UTF8 characters has to be set to YES Support CSTA User-to-User should be set to YES (!!! If license subscribed only !!!) DDI destination number has to be set to ReqURI UPDATE in Allow header/INIVTE has to be set to Mandatory RFC 4904 supported has to be set to No RFC3264 m-line has to be set to False *(for Fax T.38 support)* Sendonly for Hold has to be set to False In Band DTMF has to be set to No SIP Trunk Recording has to be set to No Send user name in SIP has to be set to User name else nuber

	Session Timer should be set to 21499 (but can be reduced to non-multiple of 1800) Min Session Timer has to be set to 900 Session Timer Method has to be set to RE_INVITE Allow Direct RTP has to be set to YES IP Domain must be configured according to the location of MGW resources (domain number to force, or -1 by default) P-ANI header has to be set to None *(not used by BTalk)* Support G722 has to be set to YES Support G711 has to be set to YES Support G729 has to be set to YES

SIP > Passerelles Externes	Domaine distant doit correspondre à l'adresse IP du SBC BTIP ou de l'OTSBC dans le cas de BTIP sur Internet Adresse IP PCS devrait être configuré à 255.255.255.255 Numéro de port doit être configuré à 5060 Type de transport doit être configuré à UDP […] Timer de supervision doit être configuré à 380 Numéro de faisceau doit correspondre au faisceau SIP public [...] RFC 3325 supporté par le distant doit être configuré à Oui […] SDP dans 18x doit être configuré à Non Authentification minimale doit être configurée à Aucun […] Contact avec adresse IP doit être configuré à Oui Type de payload dynamique (dtmf) doit être configuré à 101 Outbound Calls 100 REL doit être configuré à Supporté Incoming Calls 100 REL doit être configuré à Non demandé Type de Gateway doit être configuré à Type standard Re-Trans No. for REGISTER/OPTIONS doit être configuré à 4 P-Asserted-ID dans No Appelant doit être configuré à Non Entête P-Asserted-ID certifié doit être configuré à Oui Entête From certifié doit être configuré à Non Info. de renvoi ext. fourni par doit être configuré à Diversion Supporte le Re-invite sans SDP doit être configuré à Oui Identificat. Proxy sur Adresse IP doit être configuré à Non Spécialisé Départ doit être configuré à Non Relai SDP sur renvoi ext. devrait être configuré à 180 seulement Transparence DSP en Transit doit être configuré à Non RFC5009 supporté/Appels sortants doit être configuré à Mode1 Activation du Nonce Caching doit être configuré à Non Type procédure FAX doit être configuré à T38 DNS SRV/Réemission sur Serveur OCC doit être configuré à 0 Transfert non supervisé pour RSI doit être configuré à Non Fonctionnalité de Redirection doit être configuré à Non Transfert supervisé doit être configuré à Non […] Envoi OPTIONS doit être configuré à Oui Support des caractères UTF8 doit être configuré à Oui Support User-to-User CSTA devrait être configuré à Oui (!!! si licence souscrite uniquement !!!) Numéro de destination doit être configuré à ReqURI UPDATE présent dans Allow/INIVTE doit être configuré à Obligatoire RFC 4904 supporté doit être configuré à Non RFC3264 m-Line doit être configuré à Non (pour le Fax T.38 Sendonly sur mise en garde doit être configuré à Non DTMF dans la Bande doit être configuré à Non

	Enregistrement Trunk SIP doit être configuré à Non Envoi du Nom Utilisateur en SIP doit être configuré à 1 (Nom d'usager sinon numéro) Session Timer devrait être configuré à 21499 (mais peut être réduit à une valeur non-multiple de 1800) Min Session Timer doit être configuré à 900 Session Timer Méthode doit être configuré à RE_INVITE Autoriser RTP direct doit être configuré à Oui Domaine doit être configuré en fonction de la localisation des ressources MGW (n° de domaine pour forcer, ou -1 par défaut) P-ANI header doit être configuré à Aucun *(non utilisé par BTIP)* Support G722 doit être configuré à Oui Support G711 doit être configuré à Oui Support G729 doit être configuré à Oui
Translator > External Numbering Plan > Numbering Discriminator > Discriminator Rule > "select a Discriminator No." Traducteur > Plan de numérotation externe > Discrimination numérotation > Règle de discrimination > "sélectionner un numéro de discrimination"	ARS Route List Number has to match the ARS Route for SIP Trunking Number of Digits has to be set to the exact expected number No Table De Routage ARS doit correspondre à la Route ARS pour SIP Trunking Nombre de chiffres doit être configuré avec le nombre exact de chiffres attendus
Translator > Automatic Route Selection > ARS Route list > ARS Route > "select an ARS Route list" Traducteur > Tables de routage ARS > Table de routage ARS > Routage ARS > "sélectionner une table de routage ARS"	Trunk Group has to match the public SIP Trunk Numbering Command Table. ID has to match a SIP External Gateway Quality has to be set to Speech and Fax Faisceau doit correspondre au faisceau SIP public No table commande num. doit correspondre à une Passerelle Externe Qualité doit être configurée à Voix et Télécopie
Translator > Automatic Route Selection > ARS Route list > Time-based Route List > "select an ARS Route list" Traducteur > Tables de routage ARS > Table de routage ARS > Liste des routes temporelles > "sélectionner une table de routage ARS"	In ARS Route menu, 2 routes have to be created: ▪ 1 (route SIP to nominal BTIP/BT aSBC) ▪ 2 (route SIP to backup BTIP/BT aSBC) Dans le menu Routage ARS, 2 routes doivent être créées: ▪ 1 (route SIP vers l'aSBC BTIP/BT nominal) ▪ 2 (route SIP vers l'aSBC BTIP/BT secours)
Translator > Automatic Route Selection > ARS Route list > Numbering Command Table > "select a Numbering Command Table" Traducteur > Tables de routage ARS > Table commande num. > "sélectionner une table de commande num."	Carrier Reference has to be set to 0 (=not used) Associated Ext SIP gateway has to match the SIP Ext Gateway associated for SIP Trunking. The parameter Command has to be configured to "I" ([I]nsert). Réf.Opérateur réseau doit être configurée à 0 (=non utilisé) Gateway SIP associée doit correspondre à la Passerelle Externe associé pour SIP Trunking. Le paramètre Commande doit être renseigné à "I" ([I]nsert).
Translator > Prefix Plan Traducteur > Plan des préfixes	Prefix Meaning has to be set to Local Features Local Features has to be set to PCX address in DPNSS Signification préfixe doit être configurée à Exploitations en local Exploitations en local doit être configuré à Adresse PABX en DPNSS

Overflow through PSTN (optional)	
Translator > Automatic Route Selection > ARS Route list > Time-based Route List > "select an ARS Route list" Traducteur > Tables de routage ARS > Table de routage ARS > Liste des routes temporelles > "sélectionner une table de routage ARS"	In ARS Route menu, a third route have to be created: ▪ 1 (route SIP to nominal BTIP/BT aSBC) ▪ 2 (route SIP to backup BTIP/BT aSBC) ▪ 3 (route to PSTN - only if T0/T2 on the site) Dans le menu Routage ARS, une 3eme route doit être créée: ▪ 1 (route SIP vers l'aSBC BTIP/BT nominal) ▪ 2 (route SIP vers l'aSBC BTIP/BT secours) ▪ 3 (route vers le PSTN – seulement si T0/T2 sur le site)
Passive Communication Server configuration (optional)	
Keep only one nominal SIP External Gateway and one backup SIP External Gateway for all sites, which will be used for all CS and PCS. The same ARS will be also used for all. Ne garder qu'une seule passerelle externe SIP nominale et une seule passerelle externe SIP de secours pour l'ensemble des sites, elles seront utilisées pour l'ensemble des CS et PCS. De même, une seule ARS sera utilisée pour chacune des 2 passerelles SIP externes.	
SIP > SIP Ext Gateway > "select the SIP external gateway" SIP > Passerelles Externes > "sélectionner la passerelle Externe"	**PCS IP Address** has to be set with the "Global IP Address" ➔ **255.255.255.255** Configurer le champ **Adresse IP PCS** avec l'adresse IP globale ➔ **255.255.255.255**
Spatial redundancy configuration (optional)	
Note: Internal OXE DNS resolver must be enabled: **netadmin -p yes**. **Note:** La résolution de nom DNS doit être active sur l'OXE: **netadmin -p yes**.	
4645 Voice Mail configuration	
IP > IP Parameters > G711 VOIP Framing for 4645 IP > Paramètres IP > Framing VOIP pour G711 (4645)	Parameter has to be set to **20 ms (only supported for Appliance Server.** For CS (Commun Hardware), parameter has to remain in the default configuration (**30ms**) Ce paramètre doit être configuré à **20 ms (seulement supporté pour Appliance Server).** Pour CS (Hardware Commun), ce paramètre doit resté dans la configuration par défaut (**30ms**)
4059 Attendant configuration	
Attendant > Attendant sets Opératrice > Postes Opératrices	**Tone Presence** has to be set to **YES** **Présence tonalité** doit être configurée à **Oui**
System > Timers Installation > Temporisations	**Timer 102** has to be set to value **different to 0** to have a welcome guide (**else 0**) **Temporisation 102** doit être configurée à une valeur **différente de 0** pour avoir un guide d'accueil (**sinon 0**)
Security configuration	
! Important ! As of R101, trusted hosts including the Btalk SBCs must also be declared in the OXE firewall through the '**Restricted Access Configuration'** of **netadmin**. **! Important !** A partir de la version R101, les @IP de confiance incluant les SBCs BTIP doivent aussi être declarées dans le firewall de l'OXE via le menu '**Configuration d'Accès Restreint'** de **netadmin**.	
SIP > Trusted IP Addresses SIP > Adresses IP de Confiance	Create IP Addresses of each SIP external eco-system **including the Business Talk SBC**). killall sipmotor command must be used to validate the creation. Créer les adresses IP de chaque application externe SIP **y compris les SBC BTIP**. La commande killall sipmotor doit être utilisée pour valider la création.

5.2. Analog Third-Party gateways for Fax (or Voice)

5.2.1. Media 5 Mediatrix 4102 or C7/S7 serie

OXE configuration	
Users > ''Create an user'' Usagers > "Créer un usager"	**Set Type** must be set to **SIP Device** **Type de poste** doit être configuré sur **Périphérique SIP**
SIP > Trusted IP Addresses SIP > Adresses IP de Confiance	**Declare the IP address of the MP11x gateway** **Déclarer l'adresse IP de la Gateway Mediatrix**
Mediatrix configuration (in english only)	
Media > Codecs	**Enable** following codecs for the voice : - **G.711A** (if supported by the customer) - **G.729** (if supported by the customer) **Enable only T.38** for the data **Deactivate** the **voice activity detection**
Media > Codecs > T.38	**Activate** the **redundancy** and set it to **the lowest value : "1"** **Keep** the **detection threshold** to **"Default"** **Keep** the **frame redundancy** to **"0"** **Activate** the **"No signal"** and set timeout to **"1"**
Media > Security	**Deactivate encryption for voice and fax**
Media > Misc.	**Configure** the **jitter buffer** to match to **"Fax / Modem"** usage **Deactivate** the **CNG fax tone detection**, to prevent the gateway to switch into fax mode upon such a signal. **Enable** the **CED** and the **V.21 modulations**, to force the gateway to switch into fax mode upon those signals. Keep the default ports for voice and fax
System > Services	Restart all required services (at least MIPT process)

Finally, connect to the Mediatrix gateway through the Command Line Interface (CLI), and set "**InteropSdpT38ParametersEncoding**" parameter to "**ItuT38AnnexD**" value in order to prevent the equipment to send non-compliant fax attributes in the T.38 reINVITE.

```
login as: public
public@6.3.19.198's password:
*********************************************************************
**                    Command Line Interface                       **
*********************************************************************

(…)

Global>SipEp.InteropSdpT38ParametersEncoding          # To check the original value of the flag
SippingRealTimeFax00InternetDraft                # Default value non-compliant  with ITU-T D.2.3.1 section

Global>SipEp.InteropSdpT38ParametersEncoding=ItuT38AnnexD     # To modify the value of the flag

Global>SipEp.InteropSdpT38ParametersEncoding          # To check the new value of the flag
ItuT38AnnexD                               # Modified value compliant with ITU-T D.2.3.1 section

Global>exit                              # To exit from the CLI interface
```

5.2.2.AudioCodes MediaPack MP11x serie

OXE configuration	
Users > ''Create an user'' SIP > Trusted IP Addresses Usagers > "Créer un usager" SIP > Adresses IP de Confiance	Set Type must be set to SIP Device Declare the IP address of the MP11x gateway Type de poste doit être configuré sur Périphérique SIP Déclarer l'adresse IP de la Gateway MP11x
AudioCodes configuration (in english only)	
VoIP > SIP Definitions > General Parameters	PRACK mode must be set to Supported. Session expires method must be set to re-INVITE. Asserted identity mode must be set to Add P-Asserted-Identity. Fax signaling method must be set to T38 Relay. Detect Fax on Answer Tone must be set to Initiate T38 on Preamble Use "user=phone" in SIP URL must be set to Yes. Play Ringback Tone to IP must be set to Don't play. Play Ringback Tone to tel must be set to Prefer IP
VoIP > SIP Definitions > Proxy & Registration	Use Default Proxy must be set to YES Redundancy mode must be set to Parking. Subscription mode must be set to Per Endpoint. Authentication mode must be set to Per Endpoint. Registar transport type must be set to UDP
VoIP > Control Network > Proxy Sets Table	Enable Proxy Keep Alive must be set to Using Register. Enable proxy hot-swap must be set to Yes.
VoIP > DTMF and Supplementary > DTMF & Dialing	Declare RFC 2833 in SDP must be set to Yes. 1st Tx DTMF Option must be set to RFC2833. RFC 2833 Payload Type must be set to 101
VoIP > GW and IP to IP > Hunt Group > Endpoint Phone Number	Declare the phone number of the OXE User
VoIP > Media > Voice Settings	Silence suppression must be set to Disable. DTMF transport type must be set to RFC2833 Relay DTMF.
VoIP > Media > RTP/RTCP Settings	RFC 2833 TX payload type must be set to 101 RFC 2833 RX payload type must be set to 101
VoIP > Media > Fax/Modem/CID Settings	Fax Transport Mode must be set to T38 Relay Fax CNG Mode must be set to Doesn't send T38 re-INVITE Fax Relay Max Rate (bps) must be set to 14400bps Fax/Modem Bypass Code Type must be set to G711Alaw_64
VoIP > Coders and Profiles > Tel Profile Settings	Fax Signaling Method must be set to T38 Relay
VoIP > GW and IP to IP > Analog Gateway > Authentication	Set the SIP user phone number and password to register it to the Registrar

5.3. BTalk French customers and BTIP DROM architectures

For a trunking point of view, mechanism is similar for both 'BTalk French customers' and 'BTIP DROM' (French overseas departments), the IPBX must support international dial plans (example: CC = +262 for La Réunion instead of +33 for France) and route local calls to the dedicated aSBC pair.

"BTIP hors de France" / "BTIP DROM" additional configuration on OXE	
Add SIP External Gateway for BT SBC SIP > SIP Ext Gateway Ajouter les Passerelles SIP Externes pour les aSBC BTalk / BTIP DROM SIP > Passerelles Externes	**Please refer to ROUTE MECANISM ON OXE** **On-net calls off-net calls** Voir **ROUTE MECANISM ON OXE** **On-net calls off-net calls**
Translator > External Numbering Plan > Numbering Discriminator > Discriminator Rule > "select a Discriminator No." Traducteur > Plan de numérotation externe > Discrimination numérotation > Règle de discrimination > "sélectionner un numéro de discrimination"	**ARS Route List Number** has to match the ARS Route for SIP Trunking **Number of Digits** has to be set to the exact expected number **No Table De Routage ARS** doit correspondre à la Route ARS pour SIP Trunking **Nombre de chiffres** doit être configuré avec le nombre exact de chiffres attendus
Translator > Automatic Route Selection > ARS Route list > ARS Route > "select an ARS Route list" Traducteur > Tables de routage ARS > Table de routage ARS > Routage ARS > "sélectionner une table de routage ARS"	**Trunk Group** has to match the public SIP Trunk **Numbering Command Table. ID** has to match a SIP External Gateway **Quality has** to be set to **Speech and Fax** **Faisceau** doit correspondre au faisceau SIP public **No table commande num.** doit correspondre à une Passerelle Externe **Qualité** doit être configurée à **Voix et Télécopie**
Translator > Automatic Route Selection > ARS Route list > Time-based Route List > "select an ARS Route list" Traducteur > Tables de routage ARS > Table de routage ARS > Liste des routes temporelles > "sélectionner une table de routage ARS"	**In ARS Route menu, 2 or 3 routes have to be created:** ▪ **1 (route SIP to nominal BTIP/BT aSBC)** ▪ **2 (route SIP to backup BTIP/BT aSBC)** ▪ 3 (route to PSTN - optional - only if T0/T2 on the site) **Dans le menu Routage ARS, 2 ou 3 routes doivent être créées:** ▪ **1 (route SIP vers l'aSBC BTIP/BT nominal)** ▪ **2 (route SIP vers l'aSBC BTIP/BT secours)** ▪ 3 (route vers le PSTN – optionnel – seulement si T0/T2 sur le site)
Translator > Automatic Route Selection > ARS Route list > Numbering Command Table > "select a Numbering Command Table" Traducteur > Tables de routage ARS > Table commande num. > "sélectionner une table de commande num."	**Carrier Reference** has to be set to **0 (=not used)** **Associated Ext SIP gateway** has to match the SIP Ext Gateway associated for SIP Trunking. The parameter Command has to be configured to "I" ([I]nsert). **Réf.Opérateur réseau** doit être configurée à **0 (=non utilisé)** **Gateway SIP associée** doit correspondre à la Passerelle Externe associé pour SIP Trunking. Le paramètre Commande doit être renseigné à "I" ([I]nsert).

6. OXE + OTSBC SIP trunking configuration checklist

The aim of this chapter is to provide steps to configure an Alcatel-Lucent OTSBC / AudioCodes Mediant SBC for interworking between the OXE and BTIP/Business Talk service.
This guide shows only the settings to be checked or changed. The other settings can remain at their default values.
The "Index" values are given as example because they can be different if a configuration already exists.

On OXE, the SIP External Gateway should point to the OTSBC instead of the BTIP/BTalk SBC.
Even if the public SIP trunk between OTSBC and BTIP/BTalk is encrypted, the local SIP trunk between OXE and OTSBC can be in clear or in native encrypted mode. Only the unencrypted mode is described here and parameters are the same as those already listed in chap. 5.

Warning ! The VIA header is used to determine the origin of incoming calls when other headers do not match with the remote domain (BTIP/BTalk infrastructure) of the SIP External Gateway (OTSBC). So, as a NAT is done by the OTSBC, that parameter must be activated:
/System/Other System Param./SIP Parameters/VIA Header_ Inbound Calls Routing +True

6.1 SBC - IP Network configuration

6.1.1 Core Entities

1. Check or create the ethernet Groups:
   - GROUP 1: that corresponds to the physical (or virtual) interface on OXE side (LAN) or also called "south" side
   - GROUP 2: that corresponds to the physical (or virtual) interface on BTIP/Business Talk side (WAN) or also called "north" side

2. Check or create two Ethernet Devices:
   - LAN_Dev : that corresponds to GROUP1
   - WAN_Dev : that corresponds to GROUP2

3. Check or create two IP Interfaces:
   - LAN_IF : Associated to OXE, used for Media, SIG and Management (depends on customer architecture, could be a dedicated interface for management)
   - WAN_IF : Associated to BTIP/Btalk, used for Media and SIG

6.2 SBC - OXE LAN side configuration

6.2.1 Coders & Profiles

Create new Coder Groups through the following menu:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > Coder Groups

Configure AudioCodersGroups_0 as follow:

Parameter	Value
AudioCodersGroup_0	
Coder Name	G722
Packetization Time	20
Rate	64
Payload Type	9
Silence Suppression	Disabled
Coder Name	G711A-law (or G711µ-law depending on countries)
Packetization Time	20
Rate	64
Payload Type	8
Silence Suppression	Disabled
Coder Name	G729 (optional, depending on customer request)
Packetization Time	20
Rate	8
Payload Type	18
Silence Suppression	Disabled

Create new IP Profile through the following menu:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > IP Profiles

Create an IP Profile for OXE as follow:

Parameter	Value
General	
Name	OXE_IPProfile
SBC Media	
Extension Coders Group	AudioCodersGroups_0
RFC2833 DTMF Payload Type	101
SBC Signaling	
Remote Update Support	Supported Only After Connect
Remote re-INVITE	Supported
Media security	
SBC Media Security Mode	RTP
SBC Media	
Extension Coders Group	AudioCodersGroups_0

6.2.2 Core entities

No changes are required on the SRDs.
SETUP > SIGNALING&MEDIA >CORE ENTITIES>SRDs

Parameter	Value
Index	1
Name	defaultSRD (#1)
Sharing Policy	Shared
SBC operation mode	B2BUA
User Security mode	Accept All

Create a Media Realm through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Media Realms

Parameter	Value
Index	1
Name	OXE_MRealm
Topology location	Down
IPV4 interface name	LAN_IF
Port range start	*6000* (depends on customer)
Number of media session legs	*100* (depends on customer)
Default media realm	Yes

Create a SIP Interface through the menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > SIP Interface

Parameter	Value
General	
Name	OXE_SIPint
Topology Location	Down
Network Interface	LAN_IF
UDP Port	5060
TCP Port	0
TLS Port	0
Enable TCP Keepalive	Disable
Media	

Create a new Proxy Set through the menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Proxy Sets

Parameter	Value
Index	1
Name	OXE_PSet
SBC ipv4 SIP interface	OXE_SIPint
Proxy Keep-Alive	Using OPTIONS
Proxy Keep-Alive time	300
Proxy Hot swap	Disable

Once the OXE Proxy Set created, navigate to the **Proxy Set addresses** parameters.

For OXE, create one proxy address that contain OXE IP address or the hostname in case of spatial redundancy (DNS client must be enable on the OTSBC):

Parameter	Value
General	
Proxy Address	*<OXE_@IP or hostname>*:5060
Transport Type	UDP

Create a new IP Group through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > IP Groups

Parameter	Value
Index	1
Name	OXE_IPgrp
Topology location	Down
Type	Server
Proxy Set	OXE_PSet
IP profile	OXE_IPProfile
Media Realms	OXE_MRealm
Classify by proxy	Enable
Inbound Message Manipulation Set	1
Outbound Message ManipulationSet	3

6.2.3 Message Manipulation

SETUP > SIGNALING&MEDIA > MESSAGE MANIPULATION > Message Manipulations

The manipulation Set ID must be mentioned in "**Inbound message manipulation Set**" on the **"IP Groups"** menu above.

A message manipulation has to be created to carry out properly the OXE user-agent in the BTIP/BTalk network. It will be concatenated with the SBC user-agent:

Parameter	Value
General	
Name	User-agent_OXE
Manipulation Set ID	1
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.user-agent exists and header.user-agent regex (.*)
Action	
Action Subject	var.session.agent
Action Type	Modify
Action Value	$1

The manipulation **Set ID** must be mentioned in "**Outbound message manipulation Set**" on the "**IP Groups**" menu above.

1. Topology hiding modifies "From host" part with SBC IP address.

Parameter	Value
General	
Name	Hide IP From
Manipulation Set ID	3
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.from.url.host !contains 'Anonymous'
Action	
Action Subject	header.from.url.host
Action Type	Modify
Action Value	header.via.host

2. Topology hiding: modifies "To host" part with remote proxy IP address.

Parameter	Value
General	
Name	Hide IP To
Manipulation Set ID	3
Row Role	Use Current Condition
Match	
Message Type	any
Condition	
Action	
Action Subject	header.to.url.host
Action Type	Modify
Action Value	param.message.address.dst.ip

3. Topology hiding: modifies "Request-URI" host part with remote proxy IP address.

Parameter	Value
General	
Name	Hide IP Request-URI
Manipulation Set ID	3
Row Role	Use Current Condition
Match	
Message Type	any.request
Condition	
Action	
Action Subject	header.request-uri.url.host
Action Type	Modify
Action Value	param.message.address.dst.ip

4. Topology hiding: modifies "Diversion host" part with SBC IP address.

Parameter	Value
General	
Name	Hide IP Diversion

Manipulation Set ID	3
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.diversion exists
Action	
Action Subject	header.diversion.url.host
Action Type	Modify
Action Value	header.via.host

6.3 SBC - BTIP/Business Talk side configuration over BVPN (unencrypted flows)

For the SBC - BTIP/Business Talk side configuration **over Internet** with encrypted flows, refer to the next chapter.

6.3.1 Coders & Profiles

Coder Groups are common on both sides OXE and BTIP/Btalk.

Create new **Allowed Audio Coder Group** through the following menu:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > Allowed Audio Coder Groups

Parameter	Value
General	
Name	BTIP

Click on **Allowed Audio Coder 0 item** and click on "**+New**" to add the coders in the right order :

Parameter	Value
Index	Coder
0	G722
1	G711 A-law (or G711 µ-law depending on country)
2	G729

Create new BTIP/Business Talk IP Profile through the following menu:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > IP Profiles

Parameter	Value
General	
Name	BTIP_IPProfile
Media security	
SBC Media Security Mode	RTP
SBC Remove Crypto LifeTime in SDP	No
SBC Media	
Extension Coders Group	AudioCodersGroups_0
RFC2833 DTMF Payload Type	101
Quality of Service	
RTP IP DiffServ	46
Signalling DiffServ	46

6.3.2 Core Entities

Note: Please do not use Index 0, if so you may encounter some configuration issues!

Create a BTIP Media Realm through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Media Realms

Parameter	Value
General	
Name	BTIP_MRealm
Topology Location	UP
IPv4 Interface Name	WAN_IF
Port Range Start	*6000* (depends on customer, authorized range on BTIP or BTalk is 6000-20000)
Number of media session legs	*100* (depends on customer)
Default Media Realms	NO

Create a new SIP Interface through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > SIP Interface

Parameter	Value
General	
Name	BTIP_SIPInt
Topology Location	UP
Network Interface	WAN_IF
UDP Port	5060
TCP Port	0
TLS Port	0
Media	
Media Realm	BTIP_MRealm

Create a new Proxy Sets through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Proxy Sets

Parameter	Value
General	
Name	BTIP_PSet
SBC IPv4 SIP Interface	BTIP_SIPInt
Keep alive	
Proxy Keep-Alive	Using Options
Proxy Keep-Alive Time	300
Redundancy	
Proxy Hot swap	Enable
Redundancy mode	*Homing* (depends on customer architecture)

Once the BTIP Proxy Set created, navigate to the **Proxy Set addresses** parameters.
For BTIP, create 2 proxy addresses one for the nominal SBC and one for the Backup SBC:

Parameter	Value
General	
index	1
Proxy Address	*<BTIP_Nominal_SBC @IP>*:5060
Transport Type	UDP
index	2
Proxy Address	*< BTIP_Backup_SBC @IP >*:5060
Transport Type	UDP

Create a new IP Group through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > IP Group

Parameter	Value
General	
Name	BTIP_IPgrp
Topology Location	UP
Proxy Set	BTIP_PSet
IP Profile	BTIP_IPProfile
Media Realm	BTIP_Mrealm
Inbound Message Manipulation Set	-1
Outbound Message ManipulationSet	2

6.3.3 Message Manipulation

A message manipulation has to be created to carry out properly the user-agents in the BTIP/BTalk network. Both OXE and SBC user-agents are concatenated with a '+' :
SETUP > SIGNALING&MEDIA > MESSAGE MANIPULATION > Message Manipulations

Parameter	Value
General	
Name	User-agent_SBC
Manipulation Set ID	2
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.user-agent regex (.*)
Action	
Action Subject	header.user-agent
Action Type	Modify
Action Value	var.session.0 + '+' + $1

The manipulation Set ID must be mentioned in "Outbound message manipulation Set" on the **"IP Groups"** menu above.

6.3.4 SBC - IP-to-IP Routing

Create 3 IP-to-IP routing rules through the following menu:
SETUP > SIGNALING&MEDIA > SBC > Routing > IP-to-IP Routing

1) For OPTIONS messages:

Parameter	Value
General	
Name	Options
Match	
Request Type	OPTIONS
Action	
Destination Type	Dest Address
Destination Address	internal

2) For OXE to BTIP

Parameter	Value
General	
Name	OXE-to-BTIP
Action	
Destination Type	IP Group
Destination IP Group	BTIP_IPgrp
Match	
Source IP Group	OXE_IPgrp

3) For BTIP to OXE

Parameter	Value
General	
Name	BTIP-to-OXE
Action	
Destination Type	IP Group
Destination IP Group	OXE_IPgrp
Match	
Source IP Group	BTIP_IPgrp

6.4 SBC - BTIP/Business Talk side configuration over Internet (encrypted flows)

6.4.1 TLS context

The encrypted architecture requires the usage of an encryption Key and Ciphers present in a TLS context in order. A specific Orange TLS Context have to be created.

This SIP signaling will be configured to be compliant with Orange BTalk specification :

- For encrypted BTIP/BTalk SIP Trunk architecture we need to configure: **TLS v1.3 (recommended)** or TLS 1.2 (alternative)
- TLS **Mutual authentication** activated
- **Key size = 2048**
- Cipher list is supported as **Cipher Client/Server** :

TLS 1.3 (recommanded):

- TLS_AES_256_GCM_SHA384 (0x1302)
- TLS_AES_128_GCM_SHA256 (0x1301)
- TLS_CHACHA20_POLY1305_SHA256 (0x1303) ***(Recommanded)***

Create the security context using the menu:
SETUP > IP Network > SECURITY > TLS Contexts

Parameter	Value
General	
Name	Orange
TLS version	TLSv1.3
Cipher Server TLS 1.3	TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1305_SHA256:TLS_AES_128_GCM_SHA256
Cipher Client TLS 1.3	TLS_AES_256_GCM_SHA384:TLS_CHACHA20_POLY1305_SHA256:TLS_AES_128_GCM_SHA256
DH Key Size	2048

TLS 1.2 (alternative):

- TLS_ECDHE_RSA_WITH_AES_256_GCM_SHA384 (0xc030) *(Recommended)*
- TLS_ECDHE_RSA_WITH_AES_128_GCM_SHA256 (0xc02f)
- TLS_ECDHE_RSA_WITH_AES_256_CBC_SHA384 (0xc028)
- TLS_ECDHE_RSA_WITH_AES_128_CBC_SHA256 (0xc027)

Create the security context using the menu:
SETUP > IP Network > SECURITY > TLS Contexts

Parameter	Value
General	
Name	Orange
TLS version	TLSv1.2
Cipher Server	ECDHE-RSA-AES256-GCM-SHA384:ECDHE-RSA-AES128-GCM-SHA256:ECDHE-RSA-AES256-SHA384:ECDHE-RSA-AES128-SHA256

Cipher Client	ECDHE-RSA-AES256-GCM-SHA384:ECDHE-RSA-AES128-GCM-SHA256:ECDHE-RSA-AES256-SHA384:ECDHE-RSA-AES128-SHA256
DH Key Size	2048

6.4.2 Certificate

The TLS Context need a Certificate signed. To obtain this Certificate Authority (CA) you must generate your CSR based on the information of the SBC and Company with **SHA-256** encryption.

Create a CSR using the menu:
SETUP > IP Network > SECURITY > TLS Contexts and select the Orange TLS context previously created and click on "**Change Certificate".** Fill in the customer information in the **Certificate Signing Request** form and Click "**Create CSR**".

After creating the CSR, copy the text (including the BEGIN/END lines) and send it to your Certification Authority for signing and **get a fullchain Certificate Authority (CA) –including both Root and Intermediate CA-.**

When you have the CA files (p7b and bundle), please load it on the TLS Context just created. Only Base64 (PEM) encoded X.509 certificates can be loaded to the ALE OTSBC / Audiocodes SBC.
For that, import the certificate using the menu:
SETUP > IP Network > SECURITY > TLS Contexts and select the Orange TLS context previously created and click on "**Change Certificate".**

Scroll down to the **Upload certificates** files from your computer group, click the **Browse** button corresponding to the '**Send Device Certificate...**' field, navigate to the **cert.txt** file, and then click **Load File.**

After the certificate successfully loaded into the device, save the configuration with a device reset.

At this step the fullchain **Root and intermediate Certificate (PEM format) must be transmitted to Orange BTIP/BTalk team.**

Reversaly, as mutual authentication is used between both parties, you must import the **Root and the Intermediate Orange CA** included in the DigiCert CA by following the procedure described below. Connect to the Digicert site : https://www.digicert.com/digicert-root-certificates.htm then download and import (pem format):

- the Root CA: **DigiCert Global Root CA**
- the Intermediate CA: **DigiCert TLS RSA SHA256 2020 CA1**

  Note: the previous intermediate CA 'DigiCert SHA2 Secure Server CA' ***is no longer used***

Import the **Root and the Intermediate Orange CA** certificates using the menu:
SETUP > IP Network > SECURITY > TLS Contexts and select the Orange TLS context previously created and click on "**Trusted Root Certificates**". Import certificates.

6.4.3 TLS Mutual Authentication

Enable TLS Mutual Authentication and TLS Client Verify Server Certificate through the following menu:

SETUP > IP NETWORK > SECURITY > Security Settings

Parameter	Value
General	
TLS Mutual Authentication	Enable
TLS Client Verify Server Certificate	Enable

6.4.4 Media Security

Enable the media encryption through the following menu:

SETUP > SIGNALING&MEDIA > MEDIA > Media Security

Parameter	Value
General	
Media Security	Enable
Media Security Behavior	Preferable
Offered SRTP Cipher Suite	All

6.4.5 Coders & Profiles

Create new **Allowed Audio Coder Group** through the following menu:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > Allowed Audio Coder Groups

Parameter	Value
General	
Name	BTIP

Click on **Allowed Audio Coder 0 item** (index "0" is an example and depends on the current configuration) and click on "**+New**" to add the coders in the right order :

Parameter	Value
Index	Coder
0	G711 A-law (or G711 µ-law depending on country)

Note : only G711 codec is supported with BTIP/BTalk over internet at this time

Create new BTIP/Business Talk IP Profile through the following menu:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > IP Profiles

Parameter	Value
General	
Name	BTIPoI_IPProfile
Media security	
SBC Media Security Mode	Secured
SBC Remove Crypto Lifetime in SDP	Yes
SBC Media	
Allowed Audio Coders	BTIP
Allowed Audio Coders Mode	Restriction
RFC2833 DTMF Mode	Extend

RFC2833 DTMF Payload Type	101
Use Silence Suppression	Remove
RTP Redundancy Mode	Disable
Quality of Service	
RTP IP DiffServ	46
Signalling DiffServ	46
SBC Forward and Transfer	
Remote REFER Mode	Handle Locally
Remote 3xx Mode	Handle Locally
SBC Signaling	
Diversion Header	As is

6.4.6 Core Entities

Note: Please do not use Index 0, if so you may encounter some configuration issues!

Create a BTIP Media Realm through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Media Realms

Parameter	Value
General	
Name	BTIP_MRealm
Topology Location	UP
IPv4 Interface Name	WAN_IF
Port Range Start	*6000* (depends on customer, authorized range on BTIPoI or BToI is 6000-38000)
Number of media session legs	*100* (depends on customer)
Default Media Realms	NO

Note : on Audiocodes SBC, the RTP UDP port spacing is “10”. It means that for example 5 SIP sessions SIP, 5*10 ports RTP from 6000 to 6050 will be reserved.

Create a new SIP Interface through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > SIP Interfaces

Parameter	Value
General	
Name	BTIP_SIPInt
Topology Location	UP
Network Interface	WAN_IF
UDP Port	0
TCP Port	0
TLS Port	5061
Classification	
Classification Failure Response Type	0
Security	
TLS Context Name	Orange
TLS Mutual Authentication	Enable
Media	
Media Realm	BTIP_MRealm

Create a new Proxy Sets through the following menu:

SETUP > SIGNALING&MEDIA > CORE ENTITIES > Proxy Sets

Parameter	Value
General	
Name	BTIP_PSet
SBC IPv4 SIP Interface	BTIP_SIPInt
TLS Context Name	Orange
Keep alive	
Proxy Keep-Alive	Using Options
Proxy Keep-Alive Time	300
Redundancy	
Proxy Hot swap	Enable
Redundancy mode	*Homing* (depends on customer architecture)

Select the "**BTIP_PSet**" Proxy Set" just created and click on "**Proxy Address 1 items**" (index 1 can be different, it depends on your current configuration). Once the For BTIP over Internet, create 2 proxy addresses one for the nominal SBC and one for the backup SBC.

Parameter	Value
General	
index	1
Proxy Address	*<BTIP_Public FQDN_Nominal>*:5061
Transport Type	TLS
index	2
Proxy Address	*<BTIP_Public FQDN_Backup>*:5061
Transport Type	TLS

Note : The Public FQDN (Type A or SRV) set in the "Proxy Address" is the "**Public FQDN**" provided by Orange for the SIP trunk BTIP/BTalk. We recommend to use primarily ours Public FQDN which required **DNS Servers must be configured in "Public" network interface.**

Create a new IP Group through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > IP Group

Parameter	Value
General	
Name	BTIP_IPgrp
Topology Location	UP
Proxy Set	BTIP_PSet
IP Profile	BTIPol_IPProfile
Media Realm	BTIP_Mrealm
Inbound Message Manipulation Set	-1
Outbound Message ManipulationSet	2

6.4.7 Message Manipulation

SETUP > SIGNALING&MEDIA > MESSAGE MANIPULATION > Message Manipulations

The manipulation **Set ID** must be mentioned in "**Outbound message manipulation Set**" on the "**IP Groups**" menu above.

Implement following message manipulations:

5. To carry out properly the user-agents in the BTIP/BTalk network. Both OXE and SBC user-agents are concatenated with a '+' :

Parameter	Value
General	
Name	User-agent_SBC
Manipulation Set ID	2
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.user-agent regex (.*)
Action	
Action Subject	header.user-agent
Action Type	Modify
Action Value	var.session.0 + '+' + $1

6. Topology hiding modifies "From host" part with SBC IP address.

Parameter	Value
General	
Name	Hide IP From
Manipulation Set ID	2
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.from.url.host !contains 'Anonymous'
Action	
Action Subject	header.from.url.host
Action Type	Modify
Action Value	header.via.host

7. Topology hiding: modifies "To host" part with remote proxy IP address.

Parameter	Value
General	
Name	Hide IP To
Manipulation Set ID	2
Row Role	Use Current Condition
Match	
Message Type	any
Condition	
Action	
Action Subject	header.to.url.host
Action Type	Modify
Action Value	param.message.address.dst.ip

8. Topology hiding: modifies "Request-URI" host part with remote proxy IP address.

Parameter	Value
General	
Name	Hide IP Request-URI
Manipulation Set ID	2

Row Role	Use Current Condition
Match	
Message Type	any.request
Condition	
Action	
Action Subject	header.request-uri.url.host
Action Type	Modify
Action Value	param.message.address.dst.ip

9. Topology hiding: modifies “P-Asserted-Identity host” part with SBC IP address.

Parameter	Value
General	
Name	Hide IP PAI
Manipulation Set ID	2
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.p-asserted-identity exists
Action	
Action Subject	header.p-asserted-identity.url.host
Action Type	Modify
Action Value	header.via.host

10. Topology hiding: modifies “Diversion host” part with SBC IP address.

Parameter	Value
General	
Name	Hide IP Diversion
Manipulation Set ID	2
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.diversion exists
Action	
Action Subject	header.diversion.url.host
Action Type	Modify
Action Value	header.via.host

6.4.8 SBC - IP-to-IP Routing

Create 3 IP-to-IP routing rules through the following menu:
SETUP > SIGNALING&MEDIA > SBC > Routing > IP-to-IP Routing

1. For OPTIONS messages:

Parameter	Value
General	
Name	Options
Match	
Request Type	OPTIONS
Action	

Destination Type	Dest Address
Destination Address	internal

2. For OXE to BTIP

Parameter	Value
General	
Name	OXE-to-BTIP
Action	
Destination Type	IP Group
Destination IP Group	BTIP_IPgrp
Match	
Source IP Group	OXE_IPgrp

3. For BTIP to OXE

Parameter	Value
General	
Name	BTIP-to-OXE
Action	
Destination Type	IP Group
Destination IP Group	OXE_IPgrp
Match	
Source IP Group	BTIP_IPgrp

Glossary

- ALE : Alcatel-Lucent Enterprise International
- OXE : OmniPCX Enterprise
- OTBE : OpenTouch Business Edition
- OTMS : OpenTouch Multimedia Services
- OTMC : OpenTouch Messenging Center
- OTSBC : OpenTouch Session Border Controller
- OTFC : OpenTouch Fax Center
- IPMG : IP Media Gateway
- PCS : Passive Communication Server
- WebRTC GW : Rainbow WebRTC gateway
- BVPN : Business Virtual Private Network (Orange Business Services)
- BTIP : Business Talk IP (France) over BVPN
- BTalk : Business Talk (International) over BVPN
- BTIPoI : Business Talk IP over Internet (France)
- BToI : Business Talk over Internet (International)
- A-SBC : access Session Border Controller (Orange Business Services infrastructure)
- C-SBC : enterprise Session Border Controller on customer side ("C-SBC" for "Customer SBC")
- AS : Application Server Business Talk / BTIP
- TP WAN : Third Party WAN (on customer side)
- CAC : Call Admission Control