Business Talk & BTIP for Alcatel-Lucent Enterprise OXE & OTBE

versions addressed in this guide : OXE 12.x et OTBE 2.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.

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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 or BTIP service from Orange Business Services, 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…)

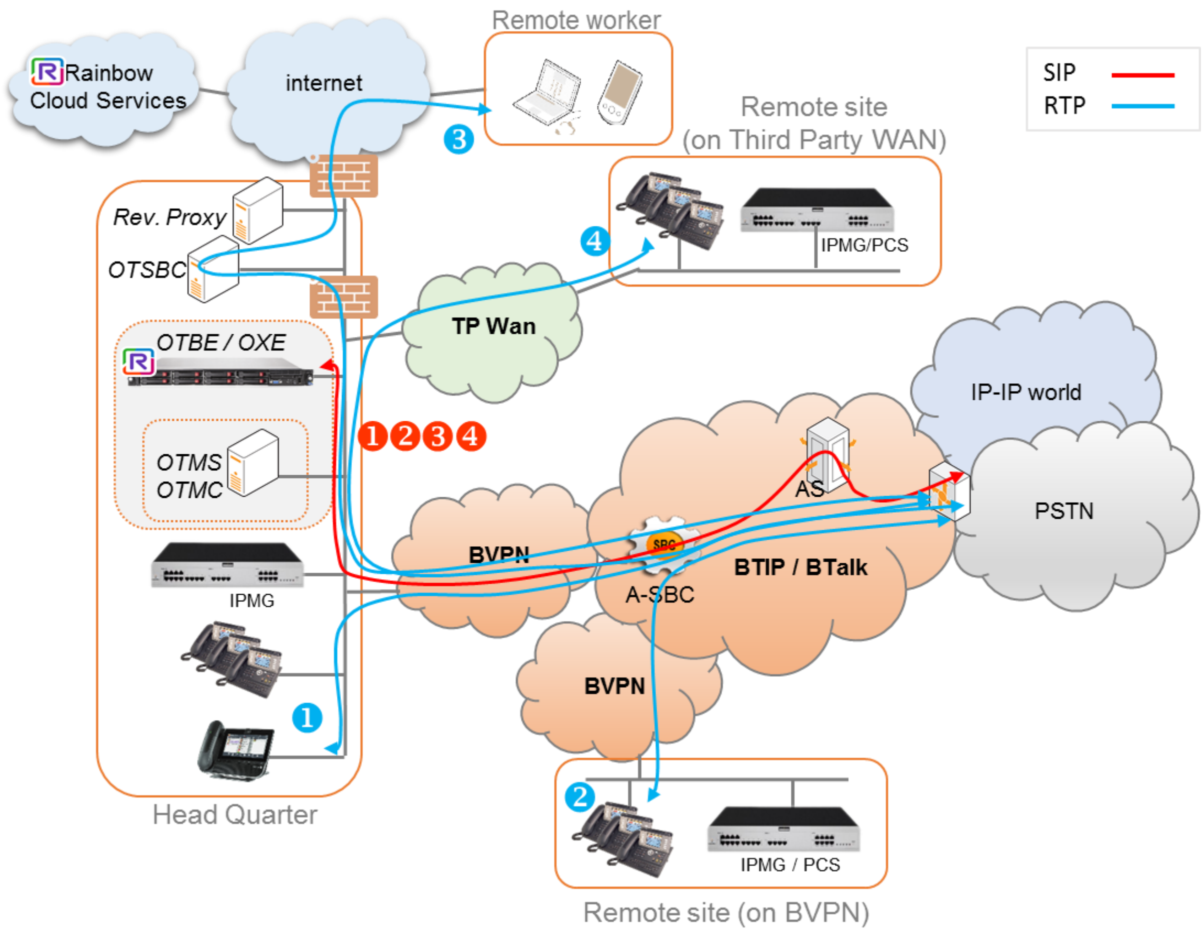
Concerning the fax support, Business talk and BTIP support the following usage :

- 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.

*cf fax servers and gateways listed in "Certified software and hardware versions section" and dedicated parameters in the "Configuration Checklist → SIP / SIP Ext Gateway""

2.2. Distributed architecture



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 :

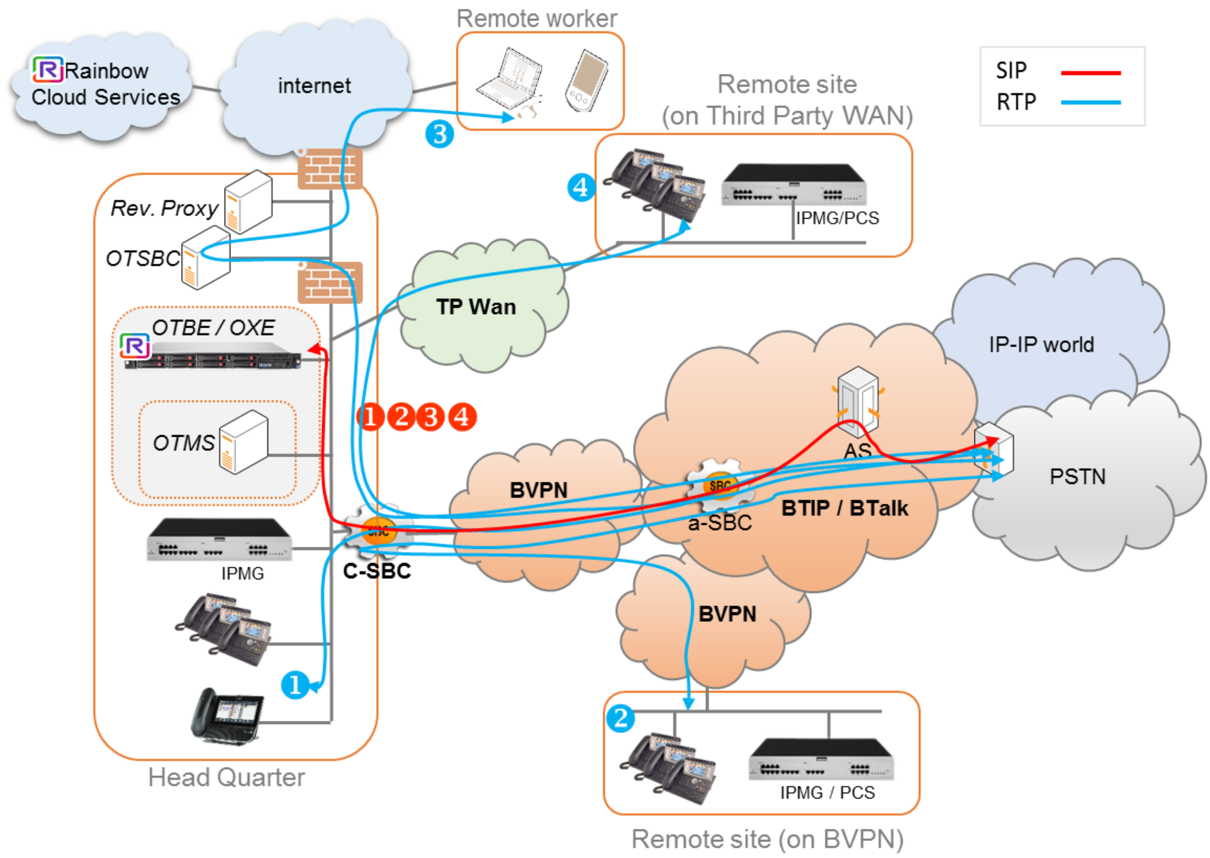
- all 'SIP trunking' signaling flows are carried by the OXE server and routed on the main BVPN connection.
- Media flows are direct between endpoints and the Business Talk/BTIP but IP routing differs from one site to another :
  - For the Head Quarter site, media flows are just routed on the main BVPN 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. Centralized architecture - with enterprise SBC



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)

In this architecture :

- Depending on the enterprise SBC equipment* we will either provide the same guidelines than the PBX ones or apply a specific “customer SBC process” to qualify the target 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).

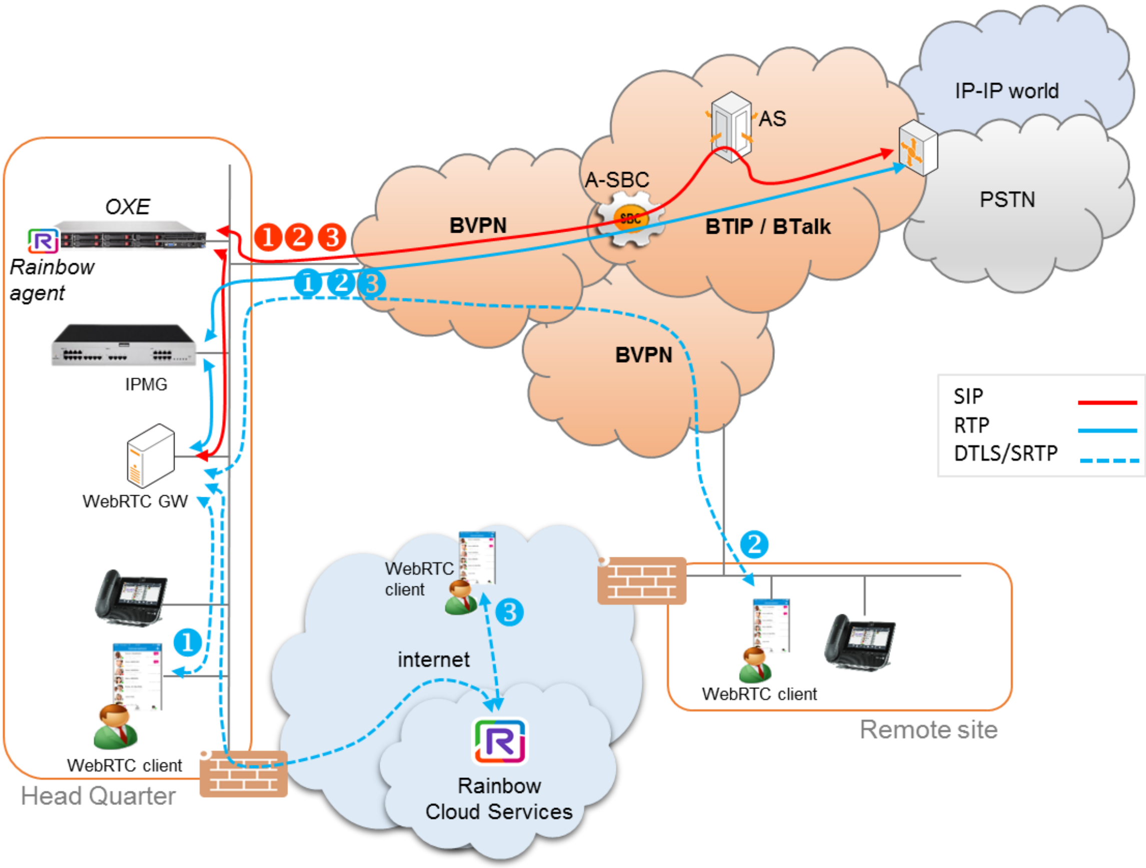
*AudioCodes Mediant (VE or appliance versions) and Alcatel OTSBC are now considered as standard, so don’t need any other specific implementation request.

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. 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 (on Business VPN)
  - ❸ call from/to remote worker (on Internet)

At this time, architectures with WebRTC media gateway do not support the direct RTP feature. The media flows to/from Business Talk/BTIP are anchored on an IPMG. Awaiting for the direct media support, 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) or Branch Office (BO) – architecture without cSBC	**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 cSBC**	**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

Customer SBC – architecture with cSBC	**Level of Service**	**@IP used by service**	
1 Customer SBC	No redundancy	cSBC @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	cSBC1 @IP	cSBC2 @IP
2 Customer SBC Load Sharing	- **Local redundancy**: both SBC are hosted on the same site OR - **Geographical redundancy** both SBC are hosted on 2 different sites	cSBC1 @IP cSBC2 @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	cSBC VIP @IP	

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 'Release Policy Information – MLV' document (ref. ENT_MLE_034198) for more details about the supported versions.

4.2. Alcatel-Lucent Entreprise IPBX

ALE IPBX – software versions			
Reference product	Software version	Certification	Certified "Loads"
OmniPCX Enterprise OpenTouch Business Edition	**OXE 12.4**	✓	m5.204. 2b, m5.204. 7c
	OXE 12.3.x	✓	m4.302.5h, m4.302.18b, m4.501.10f, m4,.501.11, m4,.501.12a, m4,.501.13, m4,.501.14, m4,.501.15a
	OTBE 2.5	✓	2.5.010.007, 2.5.015.002, 2.5.020.003

4.3. Alcatel-Lucent Entreprise endpoints and applications

ALE IPBX - endpoints and applications					
Reference product		Software version	Certification	OXE/OT version	Comments
Attendant	4068 AC	5.4.1 min	✓	12.x	
	4059 EE	2.12 min	✓	12.x	
VM	4635, 4645	-	✓	12.x	
	OTMC	2.0.100.032 min	✓	12.x	
Mobility	Desk Sharing	-	✓	12.x	
	Any Mobile	-	✓	12.x	
	OTCt smartphone & tablet (iOS, Android) & PC,	-	✓	< OT2.3	
	OTES	≧ 12.0	✓	< OT2.3	With Bluecoat or NGInx reverse proxy – no voice impact
	OTSBC	≧ 2.1	✓	OT2.x	With Bluecoat or NGInx reverse proxy
UC	Open Touch MS	2.4/2.5	✓	-	Refer to ALE TC1782 about the OTMS-OXE compatibility matrix
	Open Touch Fax Center	7.5 min	✓	12.x	
	Rainbow / Cloud Connnect	1.64 min	✓	12.x	call control only. No voice impact
	Rainbow WebRTC GW	1.74 min	✓	12.x	Restriction : no direct RTP
Recording	OmniPCX Record	2.3 min	✓	-	
Call Center	OTCC Standard	10.1 min	✓	12.x	
	OTCS / OTCS Plugin	8.2 min	✓		ACD, IVR & RSI included
	Business Contact	-	✓		Customer specific (ALE Professional Services)

ALE IPBX - endpoints and applications

Reference product		Software version	Certification	OXE/OT version	Comments
Alcatel-Lucent endpoints	OpenTouch phones (OTCv users) : 4008/4018 8002/8012 8082/8088 8001	2.11.68 min R100 2.33.1 min V2 min 3.6.09 min	✓	< OT2.3	Refer to ALE recommandations
	OXE phones (OTCt users) 8028s/8058s/8078s 40x8/80x8/40x9/80x9 8082/8088 NOE 8002/8012 SIP 8001 SIP	- - - - 3.6.09 min	✓	OT2.x OXE12.x	80x8**s** for OXE versions >12.0 Refer to ALE recommandations
	OXE IP-DECT DAP controller	R155 min 6.0.117 min	✓	12.x	Roaming included (single-node only)
	OXE DECT phones 300/400/500/500EX 8212/32/42/62/62EX	-	✓	12.x	Refer to ALE recommandations
	OXE WIFI phones (OmniTouch WLAN series)	-	✓	12.x	
	OXE softphones (IP desktop, IP desktop for iPad, MyIC desktop)	-	✓	12.x	
	4135IP	1.6.8 min	✓	12.x	'SIP device' mode
Third-party endpoints & applications	Alcatel-Business SIP phones IP100/IP150/IP2015 IP600/IP800 IP1020	1.1.0A min 15.70.0.143 min 1.2.1963 min	✓	12.x	'SIP device' mode
	VTECH SIP phones S221x/S241x	SIP_58.3.80 min	✓	12.x	Warning : G729 codec is not supported on VTECH devices
	GW AudioCodes MP11x	MP11x v07 ≧ 6.60.x	✓	12.x	Voice gateway to be declared as 'SIP device'
	Polycom SoundStation IP6000/IP7000	3.3.1 min	✓	12.x	'SIP extension' mode
	Unify Xpressions	V7 R1 FR4 min	✓	12.x	Voice Mail only
	ISI-COM Interact	7.x/8.x	✓	12.x	Contact Center
	Conecteo Kiamo	5.5/6.x/7.0	✓	12.x	Contact Center
	NGINX Reverse Proxy	R10 min	NA	OT2.x	No voice impact – mandatory with OTSBC
	Others		On demand		
Fax (T.38)	Analog fax on ALE IPMG (SLI-x, MIX-x, Z-x)	-	✓	≧ 12.2 MD3 ≧ 12.1 MD9	New parameter impact from OXE R12.2 (see configuration checklist below)
	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		
enterprise SBC	OTSBC	F7.20A min	✓	12.x	
	AudioCodes Mediant VE	7.2 min	✓	12.x	Standalone architecture

5. OXE SIP trunking configuration checklist

The checklist below presents all the **required** configuration parameters for interoperability between Business Talk/BTIP and IPBX OXE / OTBE.

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** **Enhanced codec renegotiation** has to be set to **Network type** **Packetization time par codec** doit être configure à **Non** **Négociation optimisée des codecs** doit être configuré à **Type réseau**
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**
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

Menu	Value
System > Other System param. > System Parameters Installation > Autres param.install. > Paramètres Systèmes >	**Intell Ovflw (MuAid) w/ remote ACT** has to be set to **Remote Int.Ovflw/Aid Only** (default value) *Can be different only with specific customer architecture **Entraide avec ACT distant** doit être configuré à **Entraide de l'ACT distant** (valeur par défaut) *Modifié uniquement avec une architecture client particulière
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 Domain IP > Domain IP	**Intra-domain Coding Algorithm** has to be set **Without Compression** **Extra-domain Coding Algorithm** has to be set to **Without Compression** (if g711 or g722 used) *or **With Compression** (if g729 used)* **FAX/MODEM Intra domain call transp** has to be set to **Yes** **FAX/MODEM Extra domain call transp** has to be set to **Yes** **G722/OPUS allowed in Intra-domain** has to be set to **Yes** **G722/OPUS allowed in Extra-domain** has to be set to **Yes** **Algorithme de codage intra domain** doit être configuré **Sans Compression** **Algorithme de codage Extra domain** doit être configuré **Sans Compression** (si g711 utilisé) *ou **Avec Compression** (si g729 utilisé)* **FAX/MODEM Appel Intra domain trans** doit être configuré à **Oui** **FAX/MODEM Appel Extra domain trans** doit être configuré à **Oui** **G722/OPUS autorisé en intra-domain** doit être configuré à **Oui** **G722/OPUS autorisé en extra-domain** 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
Inter-Node links > VPN Overflow Liaisons inter-Noeuds > Débordement VPN	**IP Compression Type** has to be set to **G711** if g711 used **IP Compression Type** has to be set to **Default** if g729 used **Type de compression** doit être configuré à **G711** si g711 utilisé **Type de compression** doit être configuré à **Défaut** si g729 utilisé

Menu	Value
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**
System > Other System param. > System Parameters > Intell Ovflw (MuAid) w/ remote ACT Installation > Autres param.install. > Paramètres Systèmes > Entraide avec ACT distant	**Intell Ovflw (MuAid) w/ remote ACT** has to be set to **Remote Int.Ovflw/Aid Only** (default value) *Can be different only with specific customer architecture **Entraide avec ACT distant** doit être configuré à **Entraide de l'ACT distant** (valeur par défaut) *Modifié uniquement avec une architecture client particulière
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	Session Timer has to be set to 21499 Min Session Timer has to be set to 900 Session Timer Method has to be set to RE_INVITE SDP in 180 has to be set to False Cac SIP-SIP has to be set to True Dynamic Payload type for DTMF has to be set to 101 Session Timer doit être configuré à 21499 Min Session Timer doit être configuré à 900 Session Timer Méthode doit être configuré à RE_INVITE SDP dans 180 doit être configuré à Non Cac SIP-SIP doit être configuré à Oui Type de payload dynamique (dtmf) doit être configuré à 101
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 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
Private SIP Trunk Group (for internal SIP gateway): Trunk Groups Faisceaux privés 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 > Trunk Group > “select a Trunk Group ID” Faisceaux > Faisceau > “sélectionner le faisceau SIP”	Trunk Group Type has to be set to T2 T2 Specification has to be set to SIP 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 Type faisceau doit être configuré à T2 Spécificité T2 doit être configurée à SIP 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 be 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 *(par défaut/by default):* 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 *(en option/optional):* 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 aSBC BTIP/BT 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 SIP None Ignore inactive/black hole has to be set to False 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 SDP relay on Ext. Call Fwd should be set to 180 only RFC 5009 supported/Outbound call has to be set to Mode2 Fax Procedure Type has to be set to T38 to G711 fallback Type of codec has to be set to From Domain (g722/g711 used if IP domain is set to 'Without Compression' or g729 used if IP domain is set to 'With Compression') OPTIONS required has to be set to YES Support UTF8 characters has to be set to YES Support CSTA User-to-User has to be set to YES 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 False RFC3264 m-line has to be set to False (from R12.2 MD1) for Fax T.38 Sendonly for Hold has to be set to False In Band DTMF has to be set to False

SIP > Passerelles Externes	**Domaine distant** doit correspondre à l'adresse IP du proxy distant **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** **Ignorer inactive/black hole** doit être configuré à **non** **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** **Relai SDP sur renvoi ext.** devrait être configuré à **180 seulement** **RFC5009 supporté/Appels sortants** doit être configuré à **Mode2** **Type procédure FAX** doit être configuré à **Repli T38 vers G711** **Type de négociation codec** doit être configuré à **From Domain** (g722/g711 utilisé si domaine IP configuré à 'sans Compression' ou g729 utilisé si domaine IP configuré à 'avec Compression') **OPTIONS required** doit être configuré à **Oui** **Support des caractères UTF8** doit être configuré à **YES** **Support User-to-User CSTA** doit être configuré à **YES** **Numéro de destination** doit être configuré à **ReqURI** **UPDATE in Allow header/INIVTE** doit être configuré à **Obligatoire** **RFC 4904 supporté** doit être configuré à **Non** **Conformité lignes media** doit être configuré à **Non** (à partir de R12.2 MD1) pour le Fax T.38 **Sendonly sur mise en garde** doit être configuré à **Non** **In Band DTMF** doit être configuré à **Non**

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. Remember that the PCS must NOT be set in the IP domains (they are in the default domain 0 as the CS). 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. Pour rappel, les PCS ne doivent pas être déclarés dans des domaines IP (ils seront automatiquement dans le domaine IP par défaut 0 comme les CS).	
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**.	
OXE ECOSYSTEM **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 IP 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**)
SIP Eco-systems configuration	
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. BTIP with international sites ("BTIP hors de France"/BTIP out of France)

"BTIP hors de France" additional configuration on OXE	
Add SIP External Gateway for BT SBC SIP > SIP Ext Gateway Ajouter les Passerelles SIP Externes pour les SBC vers BT 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).

5.4. Additional parameters for OTBE only

Menu	Value
Warning: G729 codec is not supported with OTCv users **Attention:** le codec G729 n'est pas supporté avec des utilisateurs ayant un profil OTCv	
COMPRESSION PARAMETERS (OXE)	
System > Other System Param. > Compression Parameters > Multi Algorithms for Compression Installation > Autres param. Install. > Paramètres Compression> Algorithmes multi. Pour Compression	This parameter has to be set to **False** Ce paramètre doit être configuré à **Non**
CALL ADMISSION CONTROL (OXE)	
IP > IP Parameters > " CAC with OTMS/OTBE" IP > IP Paramètres > "CAC avec OTMS/OTBE"	This parameter has to be set to **True** Ce paramètre doit être configuré à **Oui**
IP > CAC synchronizer IP > Synchroniseur CAC	ICE CAC Authority (primary) = **OpenTouch IP Address** Port = **2573** **ICE CAC Autorisé (principal) = Adresse Ip du serveur OpenTouch** **Port = 2573**
CALL ADMISSION CONTROL (OpenTouch)	
System services > Applications > Telephony settings > CAC > CAC configuration Services Système > Applications > Téléphonie > CAC > Configuration de la CAC	**Enable CAC feature** parameter has to be set to **True** **Bandwidth** parameters has to be set to **False** **Activer la fonction CAC** doit être configuré à **Oui** **Bande Passante** doit être configuré à **Non**
System services > Topology > OXE CS > OXE CS Network > OXE CS Sub Network > Physical Server> OXE CS > *CAC IP Link tab Services Système > Topologie > OXE CS > Réseau OXE CS > Sous-Réseau OXE CS > OXE CS > onglet *Liaison IP CAC	CAC synchronizer Port = **2573** CAC Link Keep alive = **10** Codec = **PCMA** Port de synchronisation CAC = **2573** Keep-Alive liaison CAC = **10** Codec = **PCMA**
System services > Applications > Telephony settings > CAC > Codec > PCMA Services Système > Applications > Téléphonie > CAC > Codec model > PCMA	Codec = **PCMA** Bandwidth = **64000** Codec = **PCMA** Bande passante en bit/s = **64000**

IP Domain (Open Touch)	
System services > Applications > Telephony settings > CAC > IP domain > create Services Système > Applications > Téléphonie > CAC > Domain IP > créer	Name = (example: "OpenTouch IP domain") Distribution ratio between OXE and ESS = 50 (by default) Note: according to the client site configuration, this ratio can be configured to a different value Domain ID = 100 (the same ID as IP domain ID declared on OXE) Bandwidth = Optional Video call = False Site = (example: "site 1") Specific IP and Mask address IP address = (example: 6.3.90.0) Mask = (example: 255.255.255.0) Codec = PCMA Nom = (exemple: "Domaine IP OpenTouch") Ratio de distribution entre OXE et ESS = 50 (par défaut). Note : Ce ratio peut être amené à avoir une valeur différente selon la configuration du site client. Domain ID = 100 (le même ID que celui du domaine IP déclaré sur l'OXE) Bandwidth = facultatif Video call = désactivé Site = (exemple: "site 1") Specific IP and Mask address IP address = (exemple : 6.3.90.0) Mask = (example : 255.255.255.0) Codec = PCMA
TRUSTED IP ADDRESSES (OXE)	
SIP > Trusted IP Addresses SIP > Adresses IP de Confiance	This parameter has to be filled with OpenTouch IP address Ce paramètre doit être configuré avec l'adresse IP de l'OpenTouch
SIP EXTERNAL GATEWAY (OXE) to reach OpenTouch server	
SIP > SIP Ext Gateway > Create SIP > Passerelles Externes > Créer	SIP Remote Domain = « FQDN of OpenTouch server » SIP Port Number = « 5260 » Transport type = « TCP » Belonging Domain =« » Same as SIP Remote Domain in case on Spatial redundancy, empty in any other case. Trunk group number = « Trunk group ID of OXE internal SIP Gateway » Supervision timer = 380 Trunk group number = (must match internal trunk for SIP) SIP DNS1 IP Address = « IP@ of DNS server » Minimal authentication method = « SIP None » Ignore inactive/black hole = « True » Contact with IP Address = « False » (if no spatial redundancy) « True » (if spatial redundancy) Dynamic Payload type for DTMF = 101 Outbound Calls 100 REL = Supported Incoming Calls 100 REL = « Not requested » Gateway type = « ICE Type »

	Re-Trans No. for REGISTER/OPTIONS = 4 Support Re-invite without SDP = True Proxy identification on IP address = True SDP relay on Ext. Call Fwd = 180 only FAX Procedure Type = T38 only Type of codec negotiation = From Domain Support UTF8 character set = YES Support CSTA User-to-User = YES Domaine distant = « FQDN du serveur OpenTouch » Numéro de port = « 5260 » Type de transport = « TCP » Domaine d'appartenance = « » Pareil que Domaine distant en cas de redondance spatiale, vide dans les autres cas. Timer de supervision = 380 Numéro de faisceau = « ID du faisceau de la Gateway SIP interneOXE » Adresse IP du DNS primaire = « @IP du serveur DNS » Authentification minimale = « Aucun » Ignorer Active/Black hole = « Oui » Contact avec adresse IP = « Non » (pas de redondance spatiale) « Oui » (uniquement si configuration avec redondance spatiale) Outbound Calls 100 REL = Supporté Incoming Calls 100 Rel = « non demandé » Type de Gateway = « Type ICE » Re-Trans No. for REGISTER/OPTIONS = 4 Supporte le Re-invite sans SDP doit être configuré à Oui Relai SDP sur renvoi extérieur peut être configuré à 180 seulement Type procédure FAX doit être configuré à T38 seulement Type de codec nego doit être configuré à From Domain Support des caractères UTF8 doit être configuré à YES Support User-to-User CSTA doit être configuré à YES
SIP User agent (OTBE >= 2.3)	
SIP > SIP Pproxy SIP > Proxy	User Agent Identifier = OTBE % Identification du User Agent = OTBE %

SIP EXTERNAL GATEWAY (OXE) to reach OpenTouch Voicemail	
SIP > SIP Ext Gateway > Create SIP > Passerelles Externes > Créer	SIP Remote Domain = « FQDN of OpenTouch server » SIP Port Number = « 5040 » Transport type = « TCP » Belonging Domain =« » Same as SIP Remote Domain in case on Spatial redundancy, empty in any other case. Trunk group number = « Trunk group ID of OXE internal SIP Gateway » SIP DNS1 IP Address = « IP@ of DNS server » Minimal authentication method = « SIP None » Ignore inactive/black hole = « True » Contact with IP Address = « False » (if no spatial redundancy) « True » (if spatial redundancy) Incoming Calls 100 REL = « Not requested » Gateway type = « ICE Type » Domaine distant = « FQDN du serveur OpenTouch » Numéro de port = « 5040 » Type de transport = « TCP » Domaine d'appartenance = « » Pareil que Domaine distant en cas de redondance spatiale, vide dans les autres cas. Numéro de faisceau = « ID du faisceau de la Gateway SIP interneOXE » Adresse IP du DNS primaire = « @IP du serveur DNS » Authentification minimale = « Aucun » Ignorer Active/Black hole = « Oui » Contact avec adresse IP = « Non » (pas de redondance spatiale) « Oui » (uniquement si configuration avec redondance spatiale) 100 Rel sur appel entrant = « non supporté » Type de Gateway = « Type ICE »

OpenTouch phone sets*	
Users and Devices > Audio codec > DTMF Payload Type Utilisateurs et équipements > Codec udio/Avancé Type de charge audio DTMF	This parameter has to be set to 101 Ce paramètre doit être configuré à 101
Users and Devices > Network > PRACK Utilisateurs et équipements > Réseau > PRACK	This parameter has to be set to **Disabled** Ce paramètre doit être configuré à **Désactivé**
Devices > Audio codecs équipements > Codec audio	First G722 (if supported) Second G711 Third G729 Premier G722 (si supporté) Second G711 Troisième G729

*Phones must be restarted after modification / *les postes doivent être redémarrés après la modification

5.5. OpenTouch Message Center (OTMC)

Menu	Value
COMPRESSION PARAMETERS (OXE)	
System > Other System Param. > Compression Parameters > Multi Algorithms for Compression Installation > Autres param. Install. > Paramètres Compression> Algorithmes multi. Pour Compression	This parameter has to be set to False Ce paramètre doit être configuré à Non
TRUSTED IP ADDRESSES (OXE)	
SIP > Trusted IP Addresses SIP > Adresses IP de Confiance	This parameter has to be filled with OpenTouch IP address Ce paramètre doit être configuré avec l'adresse IP de l'OpenTouch
SIP EXTERNAL GATEWAY (OXE) for OTMC	
SIP > SIP Ext Gateway > Create SIP > Passerelles Externes > Créer	SIP Remote Domain = « FQDN of OTMC » SIP Port Number = « 5040 » Transport type = « TCP » Belonging Domain =« » Same as SIP Remote Domain in case on Spatial redundancy, empty in any other case. Trunk group number = « Trunk group ID of OXE internal SIP Gateway » SIP DNS1 IP Address = « IP@ of DNS server » Minimal authentication method = « SIP None » Ignore inactive/black hole = « True » Contact with IP Address = « False » (if no spatial redundancy) « True » (if spatial redundancy) Incoming Calls 100 REL = « Not requested » Gateway type = « ICE Type » Domaine distant = « FQDN du serveur OTMC » Numéro de port = « 5040 » Type de transport = « TCP » Domaine d'appartenance = « » Pareil que Domaine distant en cas de redondance spatiale, vide dans les autres cas. Numéro de faisceau = « ID du faisceau de la Gateway SIP interne OXE » Adresse IP du DNS primaire = « @IP du serveur DNS » Authentification minimale = « Aucun » Ignorer Active/Black hole = « Oui » Contact avec adresse IP = « Non » (pas de redondance spatiale) « Oui » (uniquement si configuration avec redondance spatiale) 100 Rel sur appel entrant = « non supporté » Type de Gateway = « Type ICE »
OTMC Voicemail (OXE)	
Applications > External Voice Mail > Create Applications > Messagerie Vocale Externe > Créer	Voice Mail Dir.No. = " voicemail directory number" Sub Type = Private Directory Name = "OTMC-MEVO" as an example External Gateway Number = "sip external gateway for OTMC" Subscription on registration = "Yes"

<table>
<tr><th>Menu</th><th>Value</th></tr>
<tr><td></td><td>No annuaire Mess.Vocale = « numéro interne de la messagerie »
Type de mevo ECP
Nom d'annuaire = « OTMC-MEVO »
Numéro de gateway externe = « numéro de la gateway externe associée à l'OTMC »
Souscription sur enregistrement = « Oui »</td></tr>
<tr><td colspan="2">OTMC Automated Attendant (OXE)</td></tr>
<tr><td>Applications > External Voice Mail > Create
Applications > Messagerie Vocale Externe > Créer</td><td>Voice Mail Dir.No. = " Automated Attendant directory number"
Sub Type = Private
Directory Name = "OTMC-AA" as an example
External Gateway Number = "sip external gateway for OTMC"
Subscription on registration = "Yes"
No annuaire Mess.Vocale = « numéro interne de l'opératrice automatique »
Type de mevo ECP
Nom d'annuaire = « OTMC-AA »
Numéro de gateway externe = « numéro de la gateway externe associée à l'OTMC »
Souscription sur enregistrement = « Oui »</td></tr>
</table>

6. OXE + Enterprise SBC SIP trunking configuration checklist

The checklist below presents all the **required** configuration parameters for interoperability between Business Talk/BTIP and **AudioCodes eSBC** or **ALE OTSBC** with OXE IPBX over the Orange **Business VPN** MPLS network only. For a SIP TLS encrypted trunk, please contact your sales team.

6.1 SBC - IP Network configuration

6.1.1 Core Entities

1. Check the ethernet Groups:
   - GROUP 1: Physical (or virtual) interface that will be used for OXE (Lan)
   - GROUP 2: that correspond to the physical (or virtual) interface “BTIP” or “Busines Talk” (Wan)
2. Create two Ethernet Devices:
   - Lan_Dev : that corresponds to GROUP1
   - Wan_Dev : that corresponds to GROUP2
3. Create two IP Interfaces:
   - LAN_IF : Associated to OXE, used for Media, SIG and Management
   - WAN_IF : Associated to BTIP, used for Media and SIG

6.2 SBC - OXE customer 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	G711A-law
Packetization Time	20
Rate	64
Payload Type	8
Silence Suppression	Disabled

Note : G711A is the default supported codec. Depending on the country, G711µ is also supported. The list of codecs G722, G711(A or µ), G729 is supported too.

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

Create 2 IP Profiles for OXE and BTIP / Business Talk as follow:

Parameter	Value
General	
Name	OXE / BTIP – IP Profile
SBC Media	
Extension Coders Group	AudioCodersGroups_0
SBC Signaling	
Remote Update Support	Not Supported
Remote re-INVITE	Supported only with SDP
Remote Delayed Offer Support	Not Supported
Keep User-Agent Header	Enable

Other parameters must be kept in the default values.

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 2 Media Realms, the first for OXE and the second for BTIP through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Media Realms

Parameter	Value
Index	1
Name	OXE
IPV4 interface name	LAN_IF
Port range start	6000
Number of media session legs	5953
Port range end	65529
Default media realm	Yes

Parameter	Value
Index	1
Name	BTIP
IPV4 interface name	WAN_IF
Port range start	6000
Number of media session legs	5953
Port range end	65529
Default media realm	NO

Create 2 new SIP Interfaces through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > SIP Interface

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

Parameter	Value
General	
Name	BTIP int
Topology Location	UP
Network Interface	LAN_IF
UDP Port	5060
TCP Port	0
TLS Port	0
Enable TCP Keepalive	Disable
Media	
Media Realm	BTIP

Others parameters must be kept in the default values.

Create 2 new Proxy Sets through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Proxy Sets

Parameter	Value
Index	1
Name	OXE
SBC ipv4 SIP interface	Sipinterface1
Proxy Keep-Alive time	300
Proxy Hot swap	Disable

Parameter	Value
Index	1
Name	BTIP
SBC ipv4 SIP interface	Sipinterface2
Proxy Keep-Alive time	300
Proxy Hot swap	Enable
Redundancy mode	Homing

Create 2 new IP Groups through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > IP Groups

Parameter	Value
Index	1
Name	OXE
Type	Server
Proxy Set	OXE
IP profile	OXE
Media Realms	OXE
SIP Group Name	OXE Ip addr (Ex:6.1.51.1)
Classify by proxy	Enable
Inbound Message Manipulation Set	0
Outbound Message ManipulationSet	-1

Parameter	Value
Index	1
Name	Business Talk
Type	Server
Proxy Set	BTIP
IP profile	BTIP
Media Realms	BTIP
Classify by proxy	Enable
Inbound Message Manipulation Set	-1
Outbound Message ManipulationSet	1

Other parameters must be kept in the default values.

6.2.3 Media

Configure following parameters on the Media Security section:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > Media Security

Parameter	Value
General	
Media Security	Disable
Media Security behavior	Preferable
Master Key identifier	
Master Key Identifier (MKI) size	0
Symmetric MKI	Disable

Configure following parameters on the RTP/RTCP Settings section:
SETUP > SIGNALING&MEDIA > CODERS & PROFILES > RTP/RTCP Settings

Parameter	Value
General	
RTP Base UDP port	6000
RTCP-XR	
RTCP-XR Packet Interval	5000

Other parameters must be kept in the default values.

6.2.4 Message Manipulation

Two different message manipulations have to be created to format the SIP messages properly :

Parameter	Value
General	
Name	User-agent BTIP
Manipulation Set ID	0
Row Role	Use Current Condition
Match	
Message Type	any
Condition	header.user-agent regex (.*)
Action	

Action Subject	var.session.0
Action Type	Modify
Action Value	$1

Parameter	Value
General	
Name	User-agent BTIP
Manipulation Set ID	1
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

Others parameters must be kept in the default values.

- In OXE side:

The manipulation Set ID must be mentioned in “inbound message manipulation Set” on the **“IP Groups” menu.**

- In BTIP side

The manipulation Set ID must be mentioned in « Outbound Message Manipulation SET » on the **“IP Groups”** menu.

6.3 SBC - BTIP/Business Talk side configuration

6.3.1 Coders & Profiles

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

Parameter	Value
AudioCodersGroup_0	
Coder Name	G711A-law
Packetization Time	20
Rate	64
Payload Type	8
Silence Suppression	Disabled

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

Parameter	Value
General	
Name	OXE
Media security	
SBC Media Security Mode	RTP
Symmetric MKI	Disable
MKI Size	0
SBC Media	
Extension Coders Group	AudioCodersGroups_0
RFC2833 DTMF Payload Type	101
SBC SIGNALING	
Keep User-Agent Header	Disable

Parameter	Value
General	
Name	BTIP
Media security	
SBC Media Security Mode	RTP
Symmetric MKI	Disable
MKI Size	0
SBC Media	
Extension Coders Group	AudioCodersGroups_0
RFC2833 DTMF Payload Type	101
SBC SIGNALING	
Keep User-Agent Header	Enable

6.3.2 Core Entities

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

Parameter	Value
General	
Name	OXE
Topology Location	DOWN
IPv4 Interface Name	LAN_IF
Port Range Start	6000
Number of Media Session Legs	65529
Default Media Realms	Yes

Parameter	Value
General	
Name	BTIP
Topology Location	UP
IPv4 Interface Name	WAN_IF
Port Range Start	6000
Number of Media Session Legs	65529
Default Media Realms	NO

Other parameters must be kept in the default values.

Create 2 new SIP Interface through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > SIP Interface

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

Parameter	Value
General	
Name	SIPInterface1
Topology Location	DOWN
Network Interface	LAN_IF
UDP Port	5060
TCP Port	0
TLS Port	0
Media	
Media Realm	OXE

Parameter	Value
General	
Name	SIPInterface2
Topology Location	UP
Network Interface	WAN_IF
UDP Port	5060
TCP Port	0
TLS Port	0
Media	
Media Realm	BTIP

Other parameters must be kept in the default values.

Create 2 new Proxy Sets through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > Proxy Sets

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

Parameter	Value
General	
Name	OXE
SBC IPv4 SIP Interface	SIPInterface1
Keep alive	
Proxy Keep-Alive	Using Options
Proxy Keep-Alive Time	300
Redundancy	
Proxy Hot swap	disable

Parameter	Value
General	
Name	BTIP
SBC IPv4 SIP Interface	SIPInterface2
Keep alive	
Proxy Keep-Alive	Using Options
Proxy Keep-Alive Time	300
Redundancy	
Proxy Hot swap	Enable
Redundancy mode	Homing

Other parameters must be kept in the default values.

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

For BTIP, create 2 proxy address one fore Nominal SBC and the second for the Buckup SBC:

Parameter	Value
General	
Proxy Address	e.g. *172.22.246.41:5060*
Proxy Address	e.g. *172.22.246.81 :5060*
Transport Type	UDP

For OXE, create one proxy address that contain OXE IP address:

Parameter	Value
General	
Proxy Address	e.g. *6.1.51.1:5060*
Transport Type	UDP

Create 2 new IP Group through the following menu:
SETUP > SIGNALING&MEDIA > CORE ENTITIES > IP Group

Parameter	Value
General	
Name	OXE
Topology Location	DOWN
Proxy Set	OXE
IP Profile	OXE
Media Realm	OXE

Parameter	Value
General	
Name	BTIP
Topology Location	UP
Proxy Set	BTIP
IP Profile	BTIP
Media Realm	BTIP

Other parameters must be kept in the default values.

6.4 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

Other parameters must be kept in the default values.

2) For OXE➔BTIP

Parameter	Value
General	
Name	OXE➔BTIP
Action	
Destination Type	IP Group
Destination IP Group	BTIP
Match	
Source IP Group	OXE

3) For BTIP➔OXE

Parameter	Value
General	
Name	BTIP➔OXE
Action	
Destination Type	IP Group
Destination IP Group	OXE
Match	
Source IP Group	BTIP

6.5 OXE IPBX – External SIP Gateway configuration

Create a new SIP External Gateway on OXE: SIP>SIP Ext Gateway
Other OXE routing parameters (ARS, NPD, etc…) are the same as those described in chap. 5.

Parameter	Value	Comments
SIP External Gateway ID	1	
Gateway Name	eSBC Audiocodes/OTSBC	name of the remote access
SIP Remote domain	*6.1.51.120*	IP@ of the remote access eSBC Audiocodes/OTSBC Lan Side *(6.1.51.120 in the example)*
PCS IP Address	255.255.255.255	
SIP Port Number	5060	port of SIP messages intended for the remote gateway/proxy
Transport type	UDP	type of transport used by the gateway
Belonging Domain	--------------------	
Registration ID	--------------------	
Registration ID P_Asserted	False	
Registration timer	0	
SIP Outbound Proxy	--------------------	
Supervision timer	300	emission of OPTIONS method to the external gateway
Trunk group number	*205*	Parameter has to match the public SIP trunk *(205 in the example)*
Pool Number	-1	
Outgoing realm	--------------------	
Outgoing username	--------------------	
Outgoing Password	--------------------	
Incoming username	--------------------	
Incoming Password	--------------------	
RFC 3325 supported by the distant	True	PAI supported for outgoing calls / From: anonymous@anomymous.invalid
DNS type	DNS A	
SIP DNS1 IP Address	--------------------	
SIP DNS2 IP Address	--------------------	
SDP in 18x	False	SDP must not be present in 180 ALERTING sent by the OXE
Minimal authentication method	SIP None	no authentication requested by the proxy
INFO method for remote extension	False	
To EMS	False	
SRTP	RTP Only	
Ignore inactive/black hole	False	parameter has to be configured to False
Contact with IP address	False	
Dynamic Payload type for DTMF	101	dynamic payload proposed in RFC 2833 / required value is 101
Outbound Calls 100 REL	Supported	Since 100REL (PRACK requests) is deactivated on BTIP/BT infrastructure it should be supported for outgoing calls only
Incoming Calls 100 REL	Not Requested	
Gateway type	Standard type	Gateway type has to be set to default
Re-Trans No. for REGISTER/OPTIONS	4	
P-Asserted-ID in Calling Number	False	
Trusted P-Asserted-ID header	True	

Trusted From header	False	
Diversion Info to provide via	Diversion	
Support Re-invite without SDP	True	
Proxy identification on IP address	False	
Outbound calls only	False	
SDP relay on Ext. Call Fwd	180 only	Optional (instead of' Default' before)
SDP Transparency Override	False	
RFC5009 supported/Outbound call	Mode2	P-early-Media support for outgoing calls (in transit)
Nonce caching activation	NO	
FAX Procedure Type	T38 to G711 fallback	This parameter has to be set to « T38 to G711 fallback » Note : FAX is not provided by the OXE.
Type of codec negotiation	From Domain	For SIP Trunking, this parameter has to be set to: "From Domain"
DNS SRV/Call retry on busy server	0	
Unattended Transfer for RSI	NO	
Redirection functionality	NO	
Attended Transfer	NO	
Send BYE on REFER	YES	
Redirection response support	NO	
OPTIONS required	YES	
Support UTF8 characters set	YES	
Support CSTA User-to-User	YES	
DDI destination number	ReqURI	This parameter should be left to default value (ReqURI)
Video Support Profile	Not Supported	
UPDATE in Allow header/INVITE	Mandatory	
RFC 4904 supported	NO	
RFC3264 m-line	NO	has to be set to False (from R12.2 MD1) for Fax T. 38
Bulk registration (RFC 6140)	NO	
In Band DTMF	NO	

Glossary

- ALE : Alcatel-Lucent Enterprise
- 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
- 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)
- BVPN : Business Virtual Private Network (Orange Business Services)
- CAC : Call Admission Control
- WebRTC GW : Rainbow WebRTC gateway