TECHNICAL GUIDE to access Business Talk IP SIP IPBX Shoretel

versions addressed in this guide : 14.2

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

Document Version

Version of 10/04/2017

Table of Contents

Goal of this document

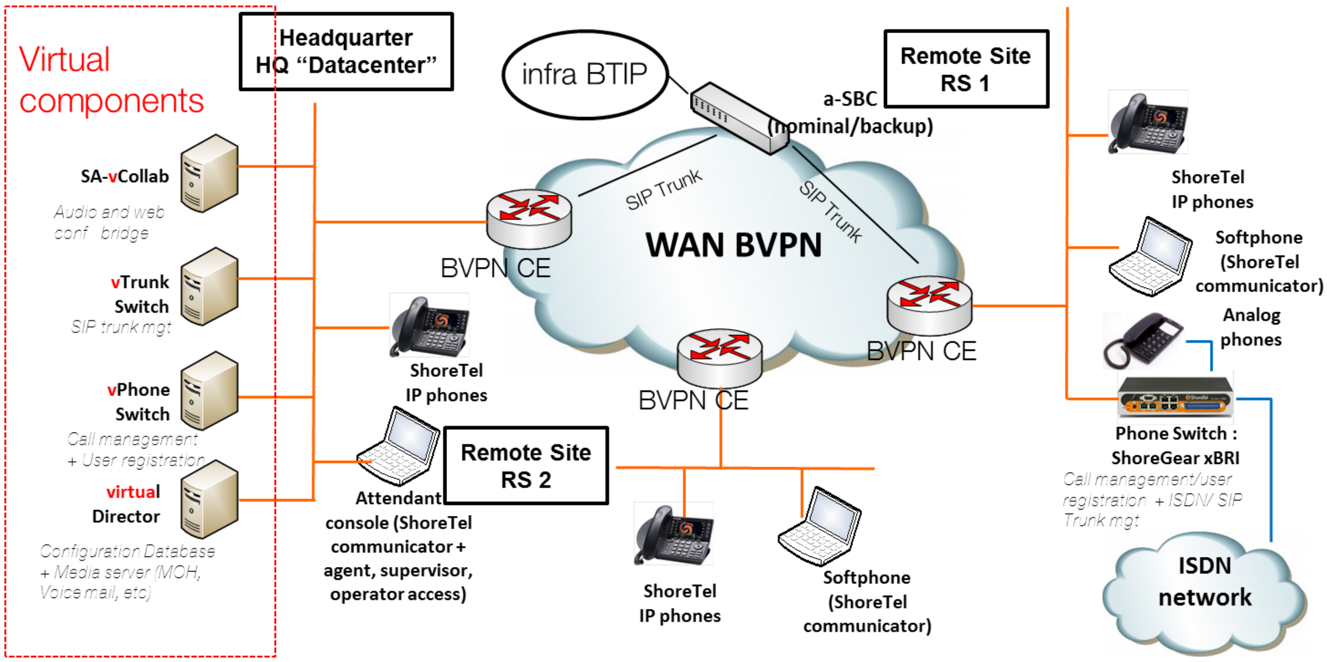
The aim of this document is to list technical requirements to ensure the interoperability between ShoreTel IPBX with OBS service Business Talk IP SIP, hereafter so-called “service”.

1 ARCHITECTURE OVERVIEW

Access to BTIP is performed through 2 a-SBC (nominal and backup).

Customer shall pay attention to get proper IPBX licencing.

2.1 Distributed architecture (virtual + hardware) components



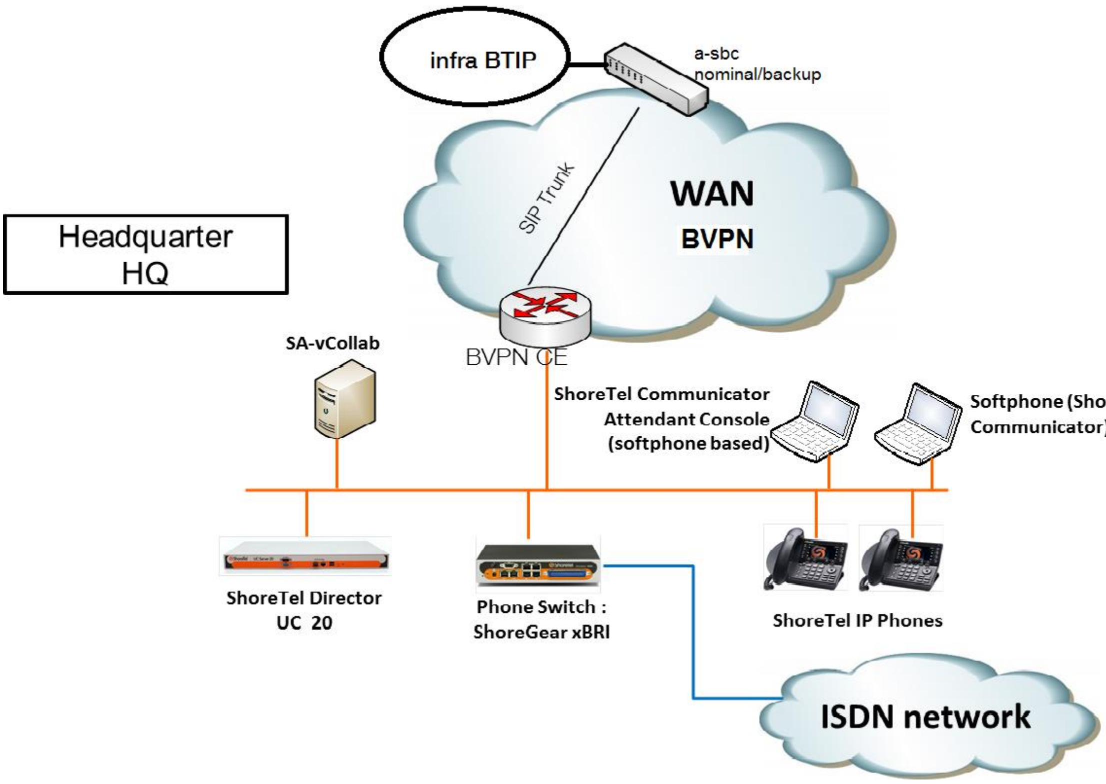
Sizing issues

For offnet calls through BTIP, media that relate to Remote sites without local BTIP SIP Trunk (case of Remote site 2 in the above picture) are anchored on vTrunk Switch. As a result, for each of Offnet calls that relate to those Remote Sites, two channels are used on Data Center hosting the VTrunk Switch.

At the other hand, for offnet calls through BTIP, media that relate to Remote sites with local BTIP SIP Trunk managed by hardware Phone Switch (Shoregear xBRI) is direct towards Orange a-sbc.

Section "Annex: Sizing rules typical example" illustrates sizing calculations on distributed architectures

2.2 Hardware based architecture



Sizing issues

For offnet calls through BTIP, media towards Orange a-sbc is direct

3 PARAMETERS to be PROVIDED by CUSTOMER to ACCESS BTIP service

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

		Customer IP addresses used by service	
Head Quarter (HQ) architecture	**Level of Service**	**Nominal**	**Backup**
Single Call Server	No call server redundancy	CS IP@	N/A
Remote Site (RS) architecture	**Level of Service**	**Nominal**	**Backup**
Remote site without ShoreGear (only IP Phones)	No survivability, no SIP trunk redundancy	N/A	N/A
Remote site with ShoreGear BRI (no SIP trunk)	Local user survivability and trunk redundancy via local PSTN only	N/A	N/A
Remote site with ShoreGear (with local SIP trunk)	Local user survivability and trunk redundancy via local PSTN only	RS Shoregear IP@	N/A

4 CERTIFIED SOFTWARE and HARDWARE versions

4.1 Release 14.2 - Load 19.47.1002.0

		VISIT SIP
		Eng. Certified
		02-juin-16

	SIP User Agent	ShoreGear/19.47.1002.0 (ShoreTel 14.2)
IP-PBX Components	Shoretel Director UC 20/25 (virtual or hardware)	19.47.1002.0
	Shoretel Virtual phone switch	19.47.1002.0
	Shoretel Virtual phone switch	19.47.1002.0
	Shoregear-30/50/90/220	19.47.1002.0
Basic	digital phones	NA
	analog phones	all
IP	Shoretel Conference Phone IP655	SWE.4.4.14
SIP	Shoretel IP Phone IP485G	802.2.1200.0
	Shoretel IP Phone IP480	802.2.1200.0
	Shoretel IP Phone IP420	802.2.1200.0
Softphone	Shoretel Communicator	19.47.1002.0
DECT	Shoretel DECT IP930D	SD1.0.0.62
	Shoretel DECT IP930D Basestation	SD1.0.0.62
WIFI		NA

Attendant	Attendant Console (Shoretel Communicator)	19.47.1002.0
Voice mail	Integrated in Shoretel Director	19.47.1002.0
Contact Center		NA
Unified Communications	ShoreTel SA-vCollab	19.47.1002.0
FMC		NA
Media GW	SG 30 BRI	19.47.1002.0
	SG 50 BRI	19.47.1002.0
	SG 90 BRI	19.47.1002.0
	SG 220 BRI	19.47.1002.0
Recorder		NA
Hybrid VoIP-TOIP archi	Cisco VoIP GW	IOS as of 15.4(3)M3
	OneAccess BLB VoIP GW	OneOS as of V5.1R5E28_FT3

Specific	FAX T38	NA
	FAX G711	NA

5 SIP TRUNKING CONFIGURATION CHECKLIST

5.1 ShoreTel Director

The checklist below presents all the steps of configuration required for interoperability between BTIP/BT and ShoreTel IPBX.

5.1.1 Trunk configuration

Trunk Configuration		
Steps	**Menu**	**Value**
1	System Parameters… > Security… > Trusted IP Ranges	Add IP Ranges authorized on the system: - Add IP ranges of different ShoreTel servers. - Add IP ranges of a-SBC servers. - If another server is added to the ecosystem, make sure that its IP address is covered by a range.
2	Trunks… > SIP Profiles	Creation of new Profile dedicated for Orange: - Set a specific name - Leave User Agent field to .* - Leave Priority to 100 - Tick '**enable**' case System Parameters have to be set to : - OptionsPing=1 - OptionsPeriod=300 (may be tuned if needed) - StripVideoCodec=1 - DontFwdRefer=1 - SendMacIn911CallSetup=1 - HistoryInfo=**diversion** - EnableP-AssertedIdentity=1 - AddG729AnnexB_NO=1 - Hairpin=1 - Register=0 - RegisterUser=BTN - RegisterExpiration=3600 - CustomRules=0 - OverwriteFromUser=0 - 1CodecAnswer=1 - IgnoreEarlyMedia=0 <u>NOTES:</u> - Custom parameters can be added to tune more specifically the SIP Profile. - The SIP Profile to apply is the same for every ShoreTel architectures.
3	Call Control… > Codec lists	Creation of two codec lists: - One list for intra-site calls - One list for inter-site calls "Orange Codecs Intra-Site" list: In 'Codec List Members', set these three codecs <u>in this order:</u> - G722/8000 (G722) - PCMA/8000 (G711 a law) - PCMU/8000 (G711 µ law) – cannot be deleted "Orange Codecs Inter-Site" list:

		In 'Codec List Members', set these three codecs in this order: - G722/8000 (G722) - PCMA/8000 (G711 a law) - PCMU/8000 (G711 µ law) – cannot be deleted
4	Sites	**Sites creation (Headquarter and Remote Sites):** Add new site in: "select the country" then click on **GO**. - In case of Remote Site creation select the **Headquarter site** name In "Parent:" field - Enter the appropriate Local Area Code (i.e. 2 for France) - Enter the appropriate Caller's Emergency Service Identification (CESID) - Select Time Zone - Set Network Time Protocol Server server - Set bandwidth value: ➔ This defines the **bandwidth** that voice streams can consume between the local site and all other sites. The caller hears a "network busy" prompt if this value is exceeded. ➔ Value have to be adjusted by operational according to site. - Select appropriate codec lists for infra-site calls and inter-sites calls (declared in step 3) - Fill the Emergency Number List (to be defined later) actually '112' - For softphone use only: Select an appropriate **Virtual IP address** according to the network addressing of the chosen **Proxy Switch**
5	Platform Hardware… > Voice Switches / Service Appliances… Primary	**Creation of ShoreTel primary switches:** Add new switch/appliance at site: "select the appropriate site" of type "select the appropriated switch type" then click on **GO**. The creation of ShoreTel voice switches directly depends on the chosen ShoreTel site architecture. Here below some examples: **Large Account architecture:** - 1 Virtual Phone switch on the Headquarter site - 1 Virtual Trunk switch on the Headquarter site - 1 Virtual SA switch on the Headquarter site - 1 ShoreGear BRI (i.e. 30BRI, 90BRI) voice switch for Remote Sites **Small and Medium Businesses architecture** - 1 ShoreGear BRI (i.e. 30BRI, 90BRI) voice switch on the Headquarter - 1 Virtual SA switch on the Headquarter site **Large Distributed architecture:** - 1 Virtual Trunk switch on the Headquarter - 1 Virtual SA switch on the Headquarter - 1 ShoreGear BRI voice switch on a Remote Site - 1 Virtual Phone switch located on Headquarter site but which will be used Remote Site without ShoreGear switch. **Parameters:** - Name = Choose a name for the created switch

		- Description = Choose a description for the created switch - Site = This field displayed the owner site - IP Address = Enter the IP address of the switch - Ethernet Address = Enter the MAC address of the switch - Server to Manage Switch = Set the site which will be managed by the switch
6	Platform Hardware… > Voice Switches / Service Appliances… Spare	Large Distributed architecture: - 1 Virtual Spare Phone switch - Name = Choose a name for the spare switch - Description = Choose a description for the spare switch - Home Site = This field displayed the owner site - Current Site = This field displayed the site currently in fail-over state - IP Address = Enter the IP address of the switch - Ethernet Address = Enter the MAC address of the switch - Server to Manage Switch = Set the site which will be managed by the switch
7	IP Phones… > IP Phone Address Map	Select the "New…" button to create and IP Address Map List for each Remote Sites (no need to create an IP Address Map List for a Headquarter site). - Site = Select the concerned site (declared in step 3) - Low IP Address = fill the first IP address of the IP network site - High IP Address = fill the last IP address of the IP network site - Enter the appropriate Caller's Emergency Service Identification (CESID)
8	IP Phones… > Options	- IP Phone Configuration Switch 1 = Enter the IP address of one ShoreTel Phone switch. - IP Phone Configuration Switch 2 = Enable IP Phone Failover = ticked to enable the N+1 ShoreTel redundancy mode on spare phone switch
9	Trunks… > Trunk groups	Creation of one SIP Trunk Group by a-SBC: - Primary SIP trunk for main a-SBC - Secondary SIP trunk for backup a-SBC Large Account architecture: - SIP Trunk Groups are created on the Virtual Trunk switch for the Headquarter site - No SIP Trunk Groups for Remote Site ShoreGear Voice switch Small and Medium Businesses architecture - SIP Trunk Groups are created on the ShoreGear Voice switch for the Headquarter site Large Distributed architecture: - SIP Trunk Groups are created on the Virtual Trunk switch for the Headquarter site - SIP Trunk Groups are created on ShoreGear Voice switch for Remote Site Add new trunk group at site: "select the appropriated site" of type:

		"select SIP trunk type" and then click on GO. - Leave box 'Enable SIP Info for G.711 DTMF Signaling' unticked - Associate trunk to concerned Site - Select Orange profile in drop-list - In Inbound part : - Number of Digits from CO : Maximal number of digits received from SIP trunk ➔ If E.164 numeration is used (+CCZABPQMCDU), then set value to 12. **Note that this numbering format is recommended.** - Tick DNIS box if DNIS Map is used - Tick DID box if DID Range is used - Tick Translation table is used - In Destination field, set Default number value ➔ This value is used to set a default destination - In Outbound part : - Billing Telephone Number field preferable order values are 1. The first number in the trunk group's DID range (the default). 2. The actual billing telephone number used by the carrier for billing purposes typed into the field by the system administrator. 3. The CESID (if the trunk group is configure to support CESID). - In Trunk services, tick those boxes Local Long Distance National Mobile International Enable Original Caller Information Caller ID not blocked by default Emergency - Note that parameter Enable Caller ID have to be unticked to avoid name transmission on SIP trunk - In trunk Digit Manipulation, tick box Dial Local Numbers in National form ➔ Trunk Digit Manipulation controls how the trunk group manipulates the telephone number before outpulsing the digits to the central office. ➔ If for outgoing calls, transmitted number of header To have to be transmitted as it was dialed (without trunk access code), then leave box Dial in E.164 format unticked. ➔ About parameter Prepend Dial Out Prefix, this

		feature is typically used when connecting the ShoreTel system to a legacy PBX system using a ShoreTel voice switch. The Dial Out Prefix enables the ShoreTel system to seize a trunk on the legacy PBX. The Dial Out Prefix is not applied to Off-System Extensions. ➔ By default, leave parameter Prepend Dial Out Prefix empty - In Trunk Group Dialing Rules (available in advanced ShoreTel menu with "CTRL+ALT+SHIFT" before login on the web interface), edit custom rules and add: ➔ The value ;65E to enable the privacy based on the user parameter « Make Number Private » of a user and E164 format in P-asserted Identity.
10	Trunks… > Individual Trunks	Creation of individual trunks for each SIP Trunk Group: Add new trunk at site: "select the appropriated site" in trunk group: "select the trunk group created in step 9" then click on GO. - Select the SIP Trunk Group created before - Set a name (HQ-SIP-Primary-Channel-00, 01, 02, … for example) - In IP Address field, set a-SBC IP@ Primary a-SBC IP@ Or secondary a-SBC IP@ ➔ Number of individual trunks are defined according license purchased
11	Users… > User Groups	User Groups creation: - Set a name (example : Executives) - Select a Class of Service - Tick boxes Send Caller ID as CESID Send DID as CESID - In 'outgoing trunk groups', select trunks created before
12	Users… > Individual Users	Users creation: - Use this menu to add new users on platform - Select site (headquarter or remote site) and click on 'Go' to add a new user - Set 'User group' to one created earlier to use the good trunk ➔ Explanation on these parameters are not directly related to SIP Trunking declaration. More information can be founded in other documentation

5.1.2 Media configuration

Media configuration		
Steps	**Menu**	**Value**
1	Call Control > Supported Codecs	Check bandwidth are defined with these values: - G722/8000 : 64kbps - PCMA/8000 : 64kbps - PCMU/8000 : 64kbps
2	Call Control > Options	Set some options to specific values: - Tick box Generate an event when a trunk is in use for 240minutes - Tick box Enable SIP Session Timer - Set Session Interval to 90sec - Set Refresher to Caller (UAC) ➔Values not used if RFC4028 is not negotiated with network (BTIP/BT case) - Set DTMF payload type to 101
3		CLIR configuration: - To dial with CLIR activated, user can use prefix *67 before number (*67 00096223284421)

5.1.3 Ecosystem configuration

Ecosystem configuration		
Steps	**Menu**	**Value**
1	System Parameters > System Extensions	Set ecosystem extension: - Voicemail - Account codes - Music on hold - Auto-Attendant - Make Me Conference - ShoreTel Conference (bridge number) - Extension - External number
2	Call Control > Options	Voice Encoding and Quality of Service: - Set a value to DiffServ / ToS Byte ➔ This parameter configures DiffServ/ToS for voicemail, workgroup, account code collection (ACC), and contact center calls. The default is 184. This setting applies to all ShoreTel servers in a ShoreTel system. To enable a new DiffServ/ToS setting, you must reboot all ShoreTel servers - Leave Media Encryption set to None Call Control Quality of Service - Set a value to DiffServ / ToS Byte ➔ This parameter configures DiffServ/ToS for call control traffic from/to ShoreTel switches, servers, and phones. The default is 96. This value should not be greater than the Voice Encoding and Quality of Service DiffServ/TOS Byte value Video Quality of Service

		- Set a value to DiffServ / ToS Byte ➔ This parameter configures the DiffServ/ToS field in the IP Packet Header of the Video Call payload packet. The default is 136. Changing this setting does not affect active video sessions. The updated value is applied to new video sessions.

NB:
The nominal/backup Orange a-sbc IP addresses are provided by Orange project manager to the customer integrator based on agreed IP addressing plan

6 MAIN SIP features and FUNCTIONAL restrictions

- Codecs:

G711 A 20 ms only (G729 A not certified)
G722 may be used for internal calls

- Encryption

TLS/SRTP used internally

- Media transport:

Note that SRTP is used for media transport in inter-sites communications. It may involve 5% increase of the required bandwidth.

- Call server /phone switch redundancy:

Not in standard with this Shoretel release

- Sizing and CAC:

For sites without local Phone switch, CAC is not managed by Shoretel system, individually, but globally for all of those sites.

Annex: Sizing rules typical example

The sizing approach can be understood through the following example:

As an example we consider a customer, which has a distributed architectures where:

- The HQ is a virtual based architecture (containing vphone switch and vtrunk switch). This involves offnet media related to devices located on sites without local SIP trunk, to be anchored on the vtrunk switch
- Some Remote Sites have their own phone switch and own SIP trunk,
- Other Remote Sites don't have dedicated SIP trunk.
  - Among them, some have a local phone switch,
  - others don't even have a local phone switch (i.e. register on the HQ)

We start from the situation where we know the customer sizing on ISDN.
We assume that the HQ is the site where the media for offnet calls related to remote sites without local SIP trunk is anchored.

Example of ISDN sizing: (*numbers in italic are just example parameters)*

On the HQ: *20* ISDN channels

Given RS i: i ISDN channels

Sum of ISDN channels on Remote Sites who don't own dedicated SIP Trunk= *10* ISDN channels

It is assumed that all those channels are used for both on net and offnet calls. Moreover, ISDN offnet trafic from Remote sites is direct : it is not anchored on HQ.

Hypothesis : *35 %* of the trafic of each site (HQ or any type of RS) relates to offnet trafic

Based on those ISDN assumptions, we propose to modelize what the sizing should be, when the customer is connected to BTIP/BT

Sizing of Voice channels on each site, on BPVN level and BTIP/BT AS :

For each RS, the number of channels should be the number of ISDN channels.

Note that in case of Remote sites without local phone switch, as indicated in section 6, proper attention should be paid to the sizing of each of this type of Remote Sites.
Indeed, in a SIP environment where dedicated Call Admission Control is not effective, a lack of bandwidth might lead to the establishment of calls without media.

For HQ, due to the offnet media anchoring of the Remote Sites that don't own a dedicated SIP trunk :
number of channels =*20* + (*35%**2**10*) = 27

On the Shoretel system declaration (following is provided for information, having in mind that proper declaration is under customer's integrator responsibility)

- For each given RS with dedicated SIP Trunk:

Number of SIP Trunk channels to be declared on **Shoretel system**= Number of ISDN channels*35%

Bandwidth to be declared on **Shoretel System** should be the G711 bandwidth multiplicated by (1-0,35) * the number of given Remote Site ISDN channels : it corresponds to the required bandwidth for intersites traffic (excluding SIP trunking traffic)

- For each given RS without dedicated SIP Trunk, but local phone switch :

Bandwidth to be declared on **Shoretel system** should be the G711 bandwidth multiplicated by the number of given Remote Site ISDN channels

- For each given RS without dedicated SIP Trunk, and without local phone switch :

As depicted in section 6, bandwidth to be declared on **Shoretel system** is global for the whole set of Remote Sites with no local phone switch.

- On the HQ :

Number of SIP Trunk channels to be declared on the **Shoretel system** : (*20* + *10*) **35*% = 10,5

Bandwitdth to be declared on the **Shoretel System** (intersites, excluding SIP trunking traffic) = (*20*-(*35%*20*)) + (*35%*10*)
=13+3,5= 16,5

Note that SRTP is used for media transport in inter-sites communications. It may involve 5% increase of the required bandwidth.