

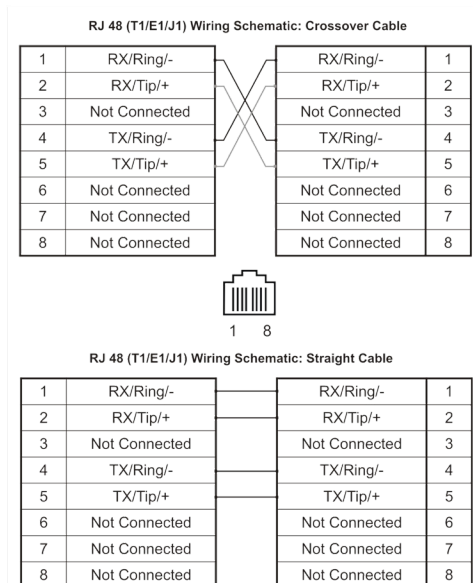
Integration Guide for Azblink SBC and Alcatel-Lucent OXO/OXE

Abstract: This document is provided as reference for the integration between Azblink SBC and Alcatel-Lucent OXO/OXE via T1/E1 link or SIP trunk.

Introduction

It is noted that deploying Azblink SBC will need a **public IP address** facing the Internet and a **private IP address** for the office network. To connect Azblink SBC with Alcatel-Lucent OXO/OXE, T1/E1 link or SIP trunk can be used. If Alcatel-Lucent OXO/OXE is with SIP trunk module, it will need an extra private IP address for itself as well. In this document, the example in the setting for the IP address part should be replaced by your own IP addresses.

For T1/E1 link, Alcatel-Lucent's OXO provides the sockets for the following schemes:



Thus, only straight cable is needed between Azblink SBC and Alcatel-Lucent's OXO for T1/E1 link. For OXE, you have to make cross-wiring cable by yourself.

And each of the Alcatel-Lucent's T1/E1 module can have its own **System Clock Timing Source** (BITS clock). Thus, it is flexible to select the

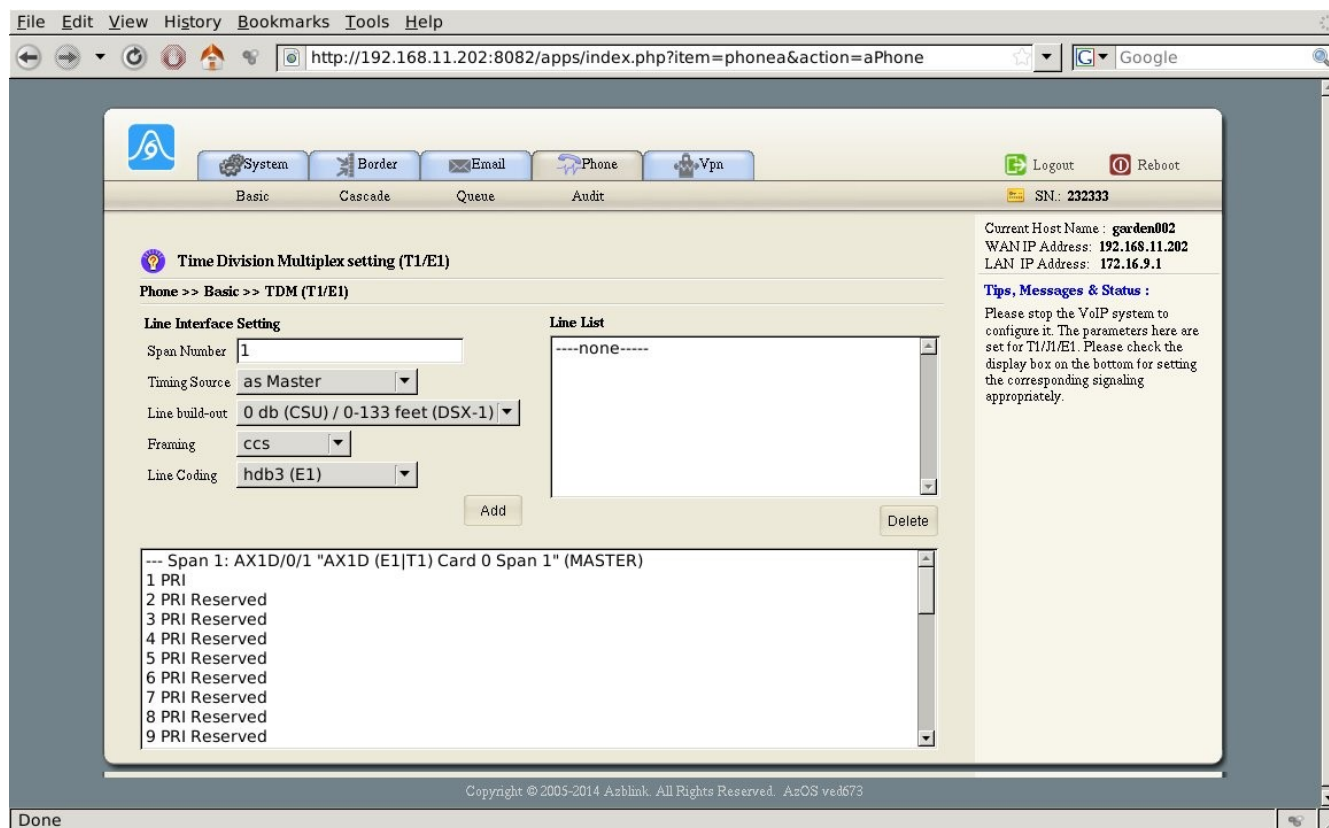
system timing source even when multiple T1/E1 modules are present on OXO/OXE system. With the description above in mind, you should adapt the changes to the deployment scenario you encounter.

T1/E1 setting

In the following, we assume that Azblink SBC's T1/E1 will be used as the System Clock Timing Source so that OXO/OXE will lock it from the other side. And we intend to run ISDN PRI over E1 on this link; OXO/OXE will act as "PRI NET" and Azblink SBC will act as "PRI CPE".

Before setting T1/E1 link on Azblink SBC, please stop VoIP process in order for the configuration to go on smoothly.

ISDN PRI belongs to CCS (Common Channel Signaling); thus, we start as follows:



After "Add" button is pressed, it takes a long while to load the parameters. After it is completed, you will find "**RED alarm**" is raised if the cable is unplugged:

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http://192.168.11.202:8082/apps/index.php?item=phonea&action=aPhone

System Border Email Phone Vpn

Logout Reboot

SN: 232333

Current Host Name : garden002
WAN IP Address : 192.168.11.202
LAN IP Address : 172.16.9.1

Tips, Messages & Status :
Please restart VoIP process to take effect.

Time Division Multiplex setting (T1/E1)

Phone >> Basic >> TDM (T1/E1)

Line Interface Setting

Span Number:

Timing Source: as Master

Line build-out: 0 db (CSU) / 0-133 feet (DSX-1)

Framing: d4 (T1/J1)

Line Coding: ami (T1/J1/E1/BR1)

Add Delete

Line List

span=1,0,0,ccs,hdb3

--- Span 1: AX1D/0/1 "AX1D (E1|T1) Card 0 Span 1" (MASTER) HDB3/CCS RED

1 PRI RED
2 PRI RED
3 PRI RED
4 PRI RED
5 PRI RED
6 PRI RED
7 PRI RED
8 PRI RED
9 PRI RED

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Done

We can move on by ignoring it at this moment.

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Phone >> Basic >> AVM/DVM

Voice Signaling

Signaling	Channel List	Group (0-63)
FXO Loop Start	<input type="text"/>	<input type="text"/>
FXO Ground Start	<input type="text"/>	<input type="text"/>
FXO Kewl Start	<input type="text"/>	<input type="text"/>
FXS Loop Start	<input type="text"/>	<input type="text"/>
FXS Ground Start	<input type="text"/>	<input type="text"/>
FXS Kewl Start	<input type="text"/>	<input type="text"/>
E&M (E1)	<input type="text"/>	<input type="text"/>
E&M	<input type="text"/>	<input type="text"/>
E&M Wink	<input type="text"/>	<input type="text"/>
Feature D MF	<input type="text"/>	<input type="text"/>
Feature D MF via Tandem Access	<input type="text"/>	<input type="text"/>
Feature B MF	<input type="text"/>	<input type="text"/>
PRI CPE	1-15,17-31	1
PRI NET	<input type="text"/>	<input type="text"/>
D-channel	<input type="text"/>	<input type="text"/>
Hardware HDLC	16	<input type="text"/>
B-channel	1-15,17-31	<input type="text"/>
Clear	<input type="text"/>	<input type="text"/>
PRI Switch type	National ISDN 2	

Voice Processing

Companding Algorithm

A-law: 1-15,17-31

mu-law:

Echo Cancellation

Note: Please fill in the Channel List as 1,2 or 3-4

Open the channel and start billing process (tagged as ANSWERED) immediately after DTMF digits are sent out

MFC/R2 (E1)

Channel List:

Group (0-63):

MFC/R2 Variant: ITU

Idle Pattern: 1101

Get ANI First:

Max ANI Digits:

Max DNIS Digits:

Caller Category: National Subscriber

SS7

Channel List:

Group (0-63):

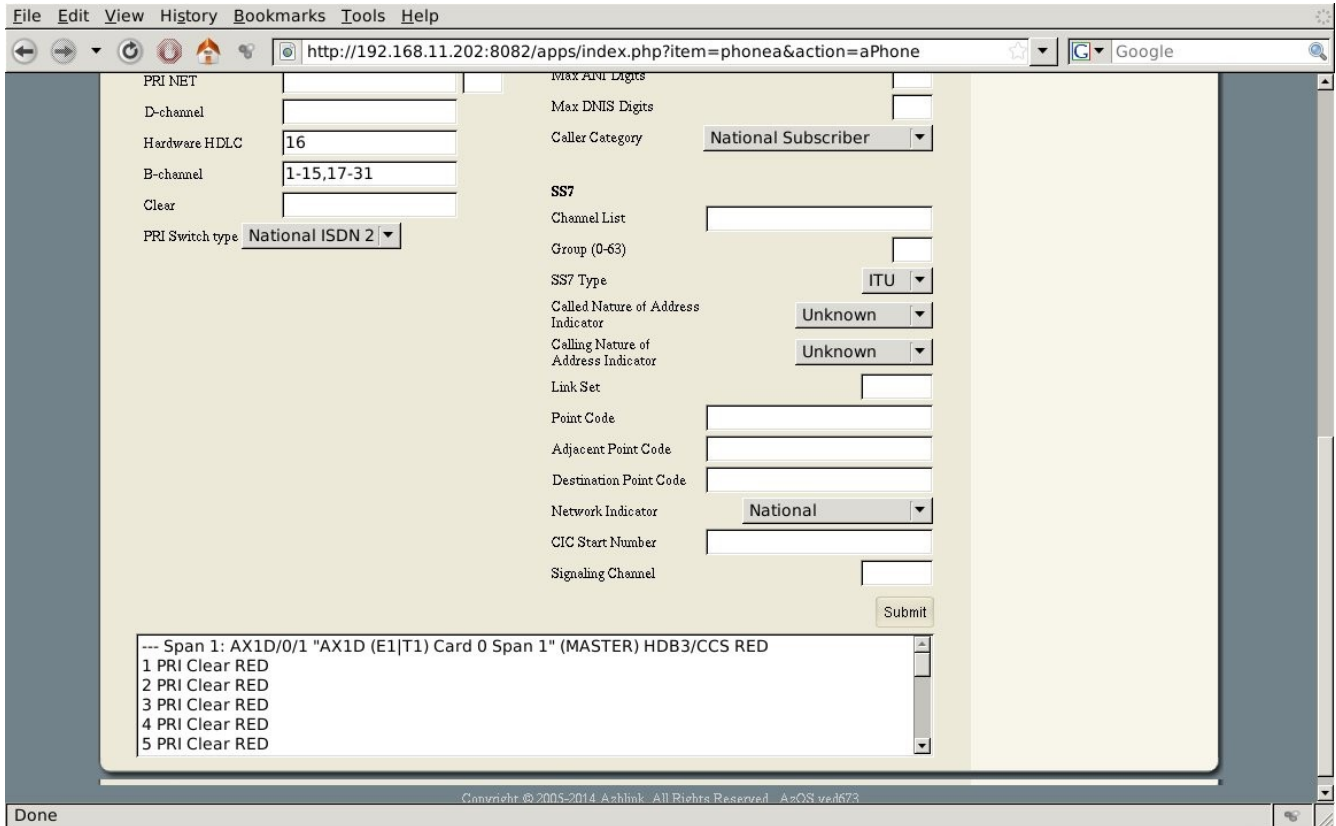
SS7 Type: ITU

Tips, Messages & Status :
The parameters here are set for Analog/Digital Voice Channel signaling, grouping, and voice processing. Please check the display box on the bottom for setting the corresponding signaling appropriately. Usually, for FXO port, you need to apply FXS signaling, for FXS port, you need to apply FXO signaling.

Done

For the signaling part, ISDN PRI is with 30 B-channels plus 1 D-channel. And we let the channels “1-15,17-31” use “**PRI CPE**” on the side of Azblink SBC.

“**Hardware HDLC**” is same as “D-channel” except that it is implemented in hardware driver. Thus, we set “Channel 16” in “**Hardware HDLC**” and just use “**National ISDN 2**” standard.



Then, we turn to “Channel Dial Plan”. Assume the numbers 1XXX and 2XXX will go to the the E1 trunk to the other end; we set as follows:

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Channel Dial Plan Setting

Phone >> Basic >> Channel Dial Plan

Setting Dial Pattern for Voice Channel or Channel Group

Channel Number: Channel Group:

Dialing Pattern: Dialing Pattern:

Number of Dial Digits to strip (empty to terminate): Number of Dial Digits to strip (empty to terminate):

Call Absolute timeout (secs): least non-busy first round-robin

Force to regenerate ringback tone Call Absolute timeout (secs):

Force to regenerate ringback tone

Delete from list

Current Host Name: garden002
WAN IP Address: 192.168.11.202
LAN IP Address: 172.16.9.1

Tips, Messages & Status :
Please restart VoIP process to take effect.

Done

File Edit View History Bookmarks Tools Help

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Channel Dial Plan Setting

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Force to regenerate ringback tone

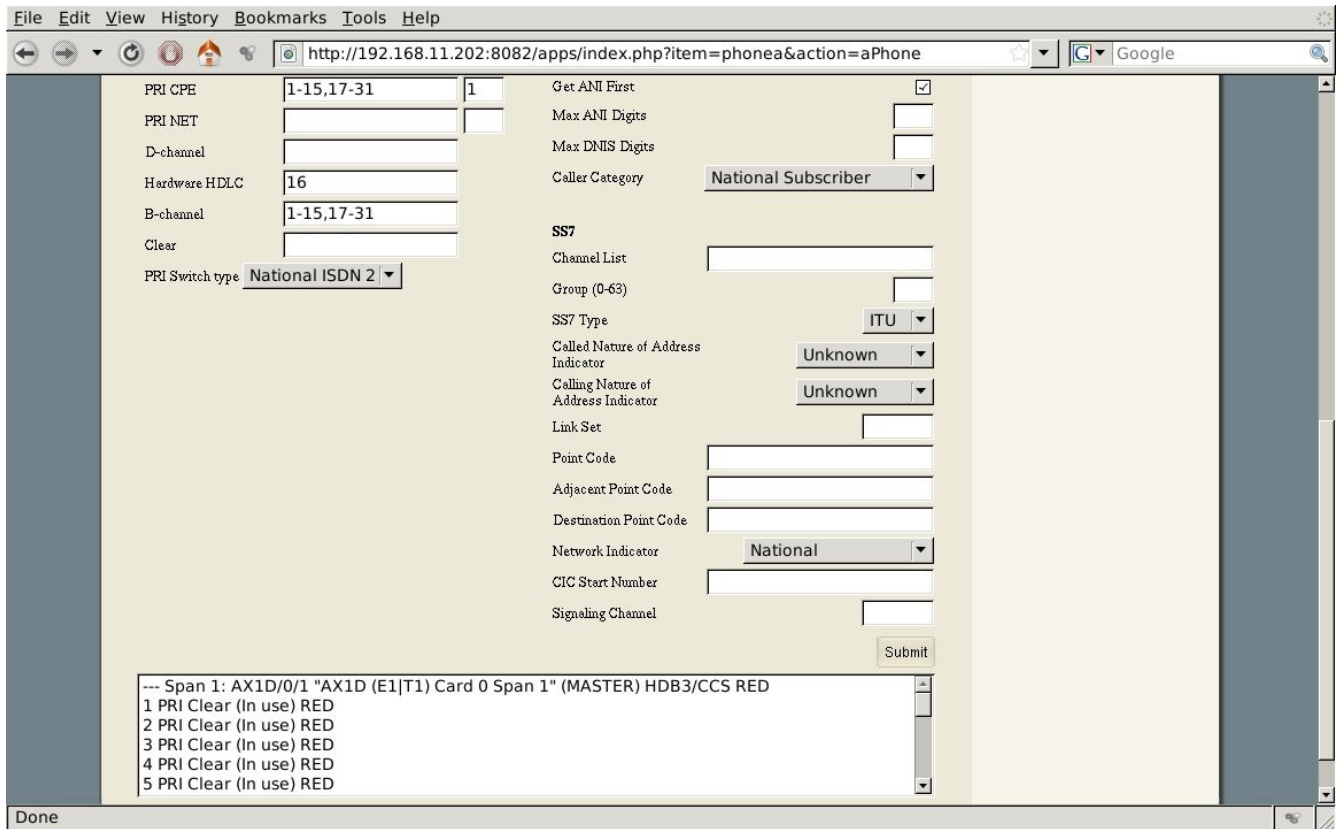
Delete from list

Current Host Name: garden002
WAN IP Address: 192.168.11.202
LAN IP Address: 172.16.9.1

Tips, Messages & Status :
Please restart VoIP process to take effect.

Done

After finishing the setting of dialing rules and restart VoIP process, we come back to check the channel status. It indicates the channels are "In Use". Once the wire is connected and the other side is configured well, the RED alarm should disappear.



SIP Trunk Setting

If there is no need to set up account authentication for SIP trunk, the setting is simple as follows:

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Basic Cascade Queue Audit

Logout Reboot

SN: 232333

Current Host Name : garden002
WAN IP Address : 192.168.11.202
LAN IP Address : 172.16.9.1

Connect to other hosts via SIP

Phone >> Cascade >> SIP Trunk

Mapping Dial Pattern to Remote Host

Trunk Identifier: toalu Transport: UDP Port: 5060

Remote hostname or IP: 192.168.11.203 Domain (From):

Dialing Pattern: _33. Outbound Proxy:

Number of Dial Digit(s) to strip: 2 Remote Host Access Account:

Prepend string (after stripping): Remote Host Access Password:

Force to regenerate ringback tone Outbound CallerID:

Add user=phone in URI Call Absolute timeout (secs):

Add

Delete from list

-----none-----

Delete

Done

Tips, Messages & Status :
The function here is to set up the dial plan and account to route the phone calls other other servers via SIP. The Dialing Pattern should be in the form like 368XXXX. In the case of _368XXXX, if you put Number of Dial Digits to strip 3, the prefix 368 will be removed and the remaining digits should be identical to the extension number on that remote machine. In other words, when you dial 3687555, 368 will route to that machine and 7555 is the user with extension number 7555 on that remote machine. The '.' can be used while specifying the dial pattern if the length of the dialing numbers is not intended to be restricted.

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WAN IP Address : 192.168.11.202
LAN IP Address : 172.16.9.1

Connect to other hosts via SIP

Phone >> Cascade >> SIP Trunk

Mapping Dial Pattern to Remote Host

Trunk Identifier: Remote hostname or IP: Dialing Pattern: Number of Dial Digit(s) to strip: Prepend string (after stripping):

Force to regenerate ringback tone

Add user=phone in URI

Transport: UDP Port: 5060

Domain (From): Outbound Proxy:

Remote Host Access Account: Remote Host Access Password:

Outbound CallerID:

Call Absolute timeout (secs):

Add

Delete from list

_33.,1,Dial(sip/\${EXTEN:2}@toalu,90)

Delete

Done

Tips, Messages & Status :
Please restart VoIP process to take effect.