

XO[®] SIP Service Customer Configuration Guide

Fonality PBXtra version 4.0/4.1 and trixbox Pro 1.4



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1.0 Overview

About This Document This document describes XO Communications SIP package 1 minimal configuration requirements for Fonality PBXtra version 4.0/4.1, and trixbox Pro version 1.4, deployed with an XO-provided Cisco 2432 IAD as the router/demarcation device.

The XO lab certification only applies to the following:

- **Fonality PBXtra** Fonality PBXtra version 4.0/4.1, which includes Fonalite hardware and software
- trixbox Pro (SE, EE and CCE editions) version 1.4 - downloadable software, only certified by XO when used with one of the Fonalite-certified servers listed at: <http://pbxtra.fonality.com/products/servers/> and <http://www.trixbox.com/products/certified-hardware/servers-and-cards>

The XO lab certification does not apply to trixbox CE or any other Fonalite products.

This document assumes the audience has a general understanding of network provisioning, connectivity requirements of XO Communications SIP service offering.

Known Issues

While XO certifies interoperability between XO SIP service and the PBX as outlined herein, the following known issues were identified during Interoperability testing. The customer should be aware that certain features and functions may not be fully supportable. In some cases, special configurations and/or PBX software patches may be available from the vendor:

- **DTMF** - Fonalite DTMF patch required in order for outbound calls to the PSTN (RFC 2833) to function. According to Fonalite, all XO SIP customers will be flagged by Fonalite's provisioning department to receive the latest PBXtra release with the patch
- **Call Center feature** - XO SIP's optional Call Center feature will not function with the Fonalite PBXtra
- **Simultaneous Calls to Hunt Group Pilot** - while 1st call to a Hunt Group is ringing, the 2nd incoming call does not ring the next available Hunt Group member. Instead, the caller hears ringing or music on hold/comfort message, depending on how the PBX is configured. The 2nd call will continue to receive this call treatment until the 1st call hangs up. According to Fonalite, a fix will be included in a future release of the PBXtra.

Registration Method

The PBXtra version 4.0 utilizes static registration between IP phones and the IP PBX.

XO SIP Service Packages Supported

The PBXtra supports XO SIP Service Package 1 only:

Pkg	Codec	DTMF	Fax
1	G.711	RFC2833 (in-band RTP DTMF fallback)	T.38; G.711 pass-through

XO SIP package 2 (G.729a compression) is not supported

2.0 Testing of Fonality PBXtra version 4.0

*Software and
Hardware Versions
Tested*

Cisco 2432 used as a router
Cisco 2432 Software Version: Cisco IOS Software, 2400 Software (C2430-IS-M),
Version 12.4(7e), RELEASE SOFTWARE (fc5)
System image file is "flash:c2430-is-mz.124-7e.bin"

Juniper Networks ERX 1400
System Release: erx_6-1-3p0-11.rel
Version: 6.1.3 patch-0.11 [BuildId 5148]

BroadSoft
System Release: 13.0

Sonus NBS
Release: V06.04.06 F001

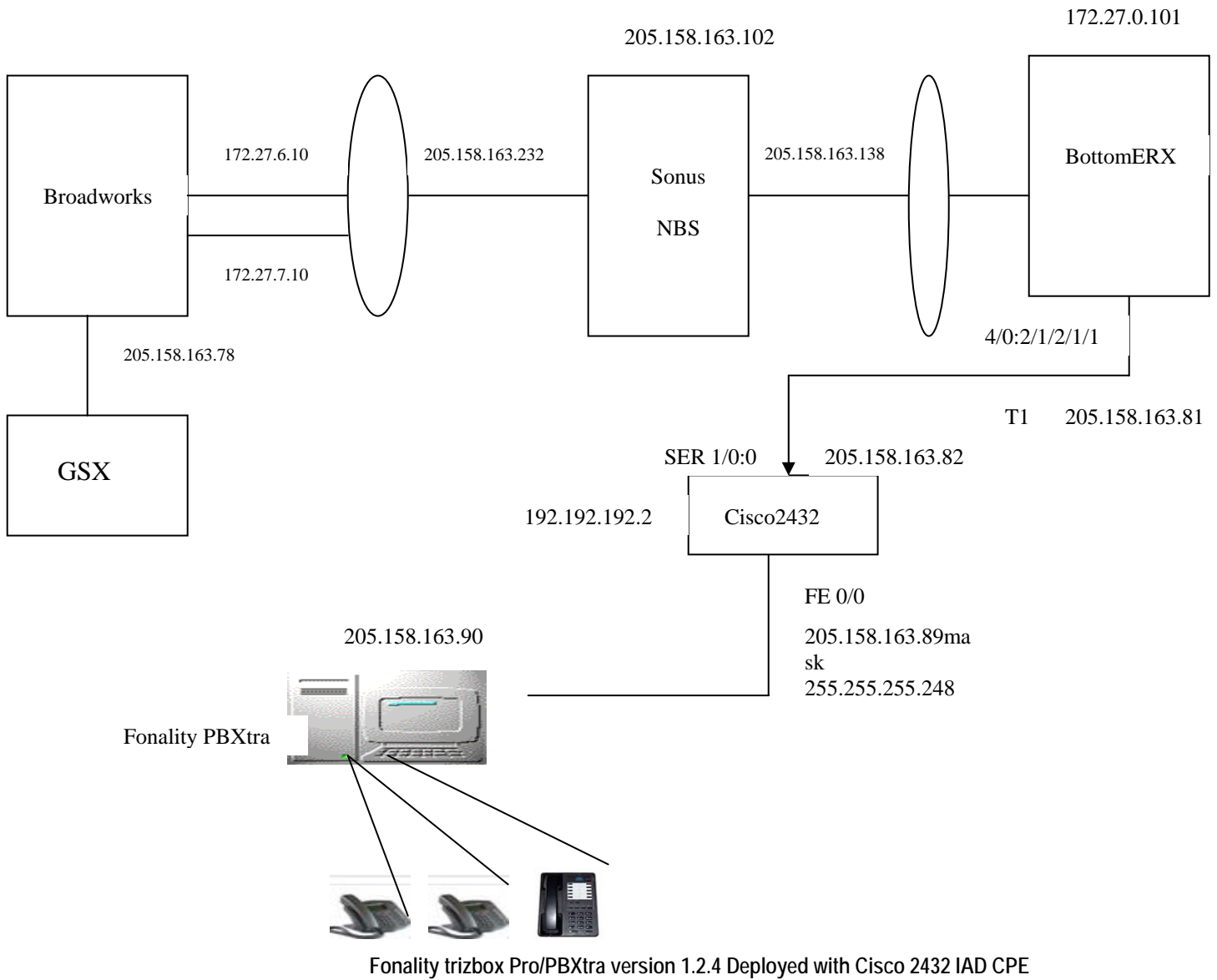
Sonus PSX and GSX
Release: V06.04.03R000

Fonality PBXtra-asterisk 1.2.17
Hardware No: 4651
Software: Asterisk Build: Asterisk 1.2.17

Polycom & Aastra SIP Phone & 1 Analog Phone
Polycom IP 430 release 2.2.1 revision C
Aastra 9112i IP phone

2.2 Diagram of Lab Test Set-Up for Fonality PBXtra version 4.0

XO IP Network The following diagram is the configuration used during XO lab testing for the Fonality PBXtra version 4.0 deployed with a Cisco 2432 IAD as CPE.



2.3 Fonality PBXtra Configuration for Version 4.0

In This Section This section contains GUI and dump configuration on phone provisioning, service provider, users, voicemail, and SIP configuration.

1. Extensions Configuration

The screenshot shows the Fidelity PBXtra web interface in a Mozilla Firefox browser. The page title is "PBXtra@ :: Server #4651 - Mozilla Firefox". The address bar shows the URL "http://cp.fonality.com/cpa.cgi?do=ext&list_ext=all". The page has a navigation menu with tabs for "AutoAnswer", "Extensions", "A.C.D.", "Reporting", "Status", and "Options". Below the menu, there are buttons for "add extension", "view extensions", "phone numbers", and "groups". The main content area shows a table titled "Showing 3 Extensions" with a search bar. The table has columns for "Del", "Ext", "Status", "User's Name", "Description", "Direct Dial", "Blast", "HUD", "Fwd.", and "Login".

Del	Ext	Status	User's Name	Description	Direct Dial	Blast	HUD	Fwd.	Login
X	7003	n/a	FX1, Port1	analog line port1	469-387-6603		✓		LOGIN
X	7000	●	Fonality, Pbxtra-1		469-387-6600		✓		LOGIN
X	7001	●	Fonality, Pbxtra-2	Test Polycom	469-387-6601		✓		LOGIN

2. Phone numbers configuration

Any phone number used by PBXtra must be added to the table below. These phone numbers are used in many places such as for Caller-ID and DIDs (Direct Inward Dials).

Add Phone Number

Number? Type? VoIP Description?

Note: To add a range of phone numbers, read [this](#)?

Showing 4 Phone Numbers

Del.	Phone Number	Type	Ext.	Name	Description	Edit
<input type="checkbox"/>	469-387-6600	VoIP	7000	Fonality, Pbxtra-1		
<input type="checkbox"/>	469-387-6601	VoIP	7001	Fonality, Pbxtra-2		
<input type="checkbox"/>	469-387-6603	VoIP	7003	FX1, Port1		
<input type="checkbox"/>	469-387-6606	VoIP			Hunt Group	
<input type="checkbox"/> Check All						

3. Hunt Group configuration

The screenshot shows the Fidelity PBXtra web interface in a Mozilla Firefox browser window. The browser address bar shows the URL: `http://cp.fonality.com/cpa.cgi?do=list_queues`. The page title is "PBXtra@ :: Server #4651 - Mozilla Firefox".

The interface has a navigation menu with tabs: AutoAnswer, Extensions, A.C.D., Reporting, Status, Options, and logout. Below the menu are links: view queues, add queue, acd reports, and recording. A note states: "Note: users must log-in to be marked as available! [Learn how]".

Below the note, there is a text block: "Below is a list of queues on your server. You may select a queue from the bottom of this page and click 'Clear' to zero out it's displayed stats (Holding, Hold Time, Completed, etc.) Note: this will only clear this page - historical stats will always be available under the ACD->reports link above."

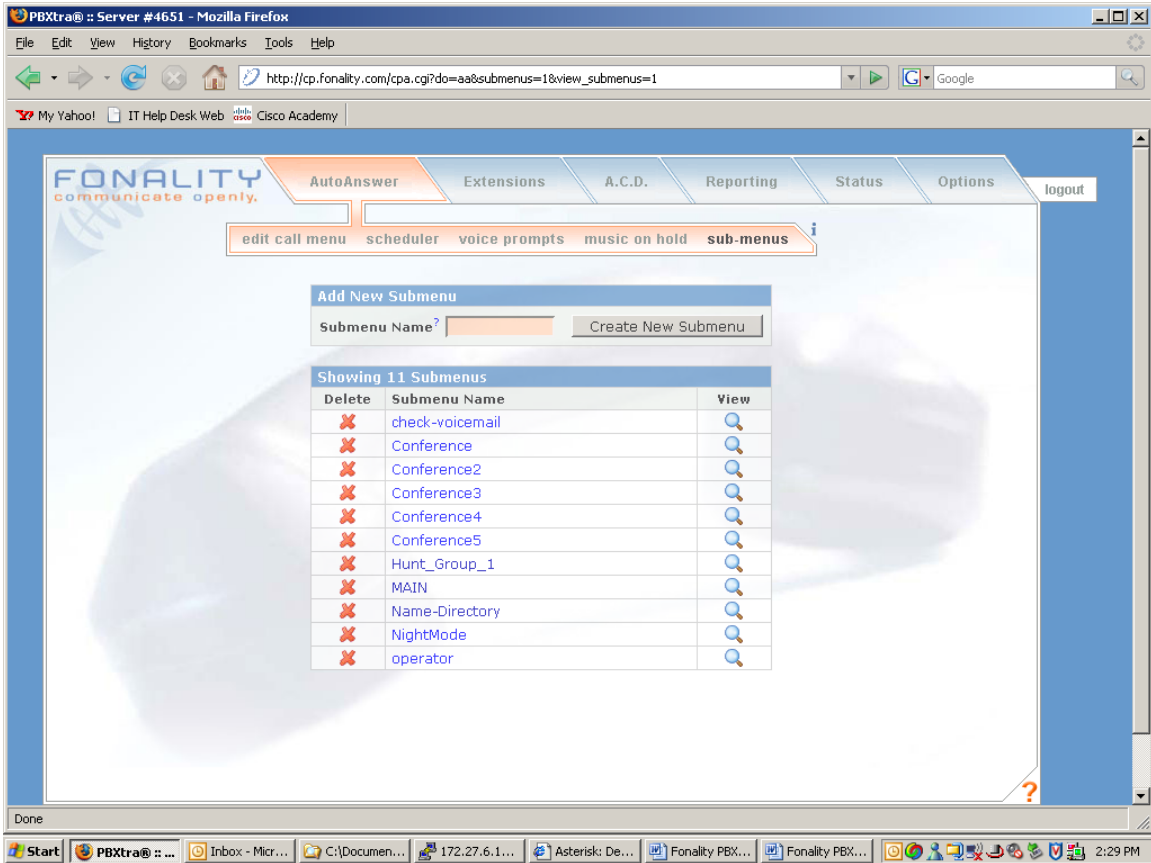
The main content area displays "Showing 1 Queue" with a search box. Below this is a table with columns: Del, Name, Hold, Hold Time?, Comp., Aband., and Queue Agents.

Del	Name	Hold	Hold Time?	Comp.	Aband.	Queue Agents
	HuntGroup1	0	0.2 min.	4	41	Pbxtra-1 Fidelity 7000 available logout Port1 FX1 7003 available logout Pbxtra-2 Fidelity 7001 available logout

Below the table, there is a "Clear stats for:" dropdown menu with "HuntGroup1" selected and a "Clear" button. To the right, it says: "All stats cleared 1 week, 3 days, 3 hours, and 56 minutes ago."

At the bottom of the page, it says: "This page will reload in 56 seconds."

The Windows taskbar at the bottom shows the Start button, several open applications including "PBXtra@ :: ...", "Inbox - Mic...", "C:\Documen...", "172.27.6.1...", "Asterisk: De...", "Fidelity PBX...", and "Fidelity PBX...", and the system tray with the time "2:28 PM".



Call Sequence (what caller hears)

Del	Step	Action
<input type="checkbox"/>	#1	Answer the incoming call
<input type="checkbox"/>	#2	Go to: <input type="text" value="Hunt_Group_1"/> if No.: <input type="text" value="469-387-6606"/>
<input type="checkbox"/>	#3	Go to: <input type="text" value="Check-voicemail"/> if CID begins with: <input type="text" value="4693876610"/>
<input type="checkbox"/>	#4	Play: <input type="text" value="hello"/> listen for Keypress
<input type="checkbox"/>	#5	Wait: <input type="text" value="5"/> second(s) - listen for Keypress
<input type="checkbox"/>	#6	Play: <input type="text" value="hello"/> listen for Keypress
<input type="checkbox"/>	#7	Wait: <input type="text" value="5"/> second(s) - listen for Keypress
<input type="checkbox"/>	#8	Play: <input type="text" value="goodbye"/> listen for Keypress
<input type="checkbox"/>	#9	Hang-up the call

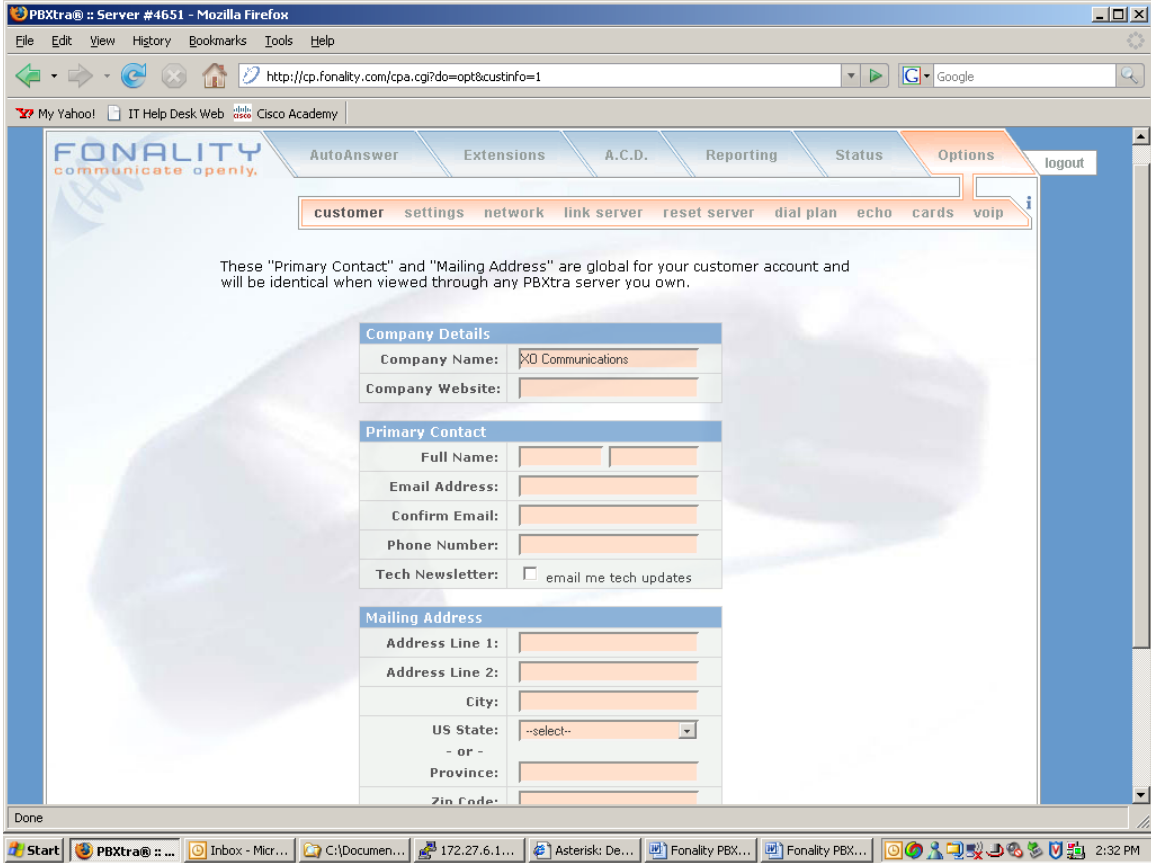
Keypress Options (what caller presses)

Del	If caller presses	Go to Submenu/ext.
<input type="checkbox"/>	<input type="text" value="0"/>	<input type="text" value="Operator"/>
<input type="checkbox"/>	<input type="text" value="6"/>	<input type="text" value="Check-voicemail"/>
<input type="checkbox"/>	<input type="text" value="9"/>	<input type="text" value="Name-Directory"/>

Create Submenu?

♦ Any extension may be dialed while Call Sequence is "listening" for Keypress

4. Service Carrier configuration



5. Network configuration

The screenshot shows the Fonality PBXtra web interface in a Mozilla Firefox browser window. The browser address bar shows the URL: `http://cp.fonality.com/cpa.cgi?do=opt&ip_config=1`. The interface has a navigation menu with tabs for Auto Answer, Extensions, A.C.D., Reporting, Status, and Options. The 'Options' tab is active, and a sub-menu is open showing 'customer', 'settings', 'network', 'link server', 'reset server', 'dial plan', 'echo', 'cards', and 'voip'. The 'network' option is selected.

The main content area displays two configuration sections:

IP Configuration Settings

Revert If Error?	Yes
IP Address Mode?	Static
IP Address?	205.158.163.90
Subnet Mask?	255.255.255.248
Gateway?	205.158.163.89
Network?	205.158.163.88
Broadcast?	205.158.163.95

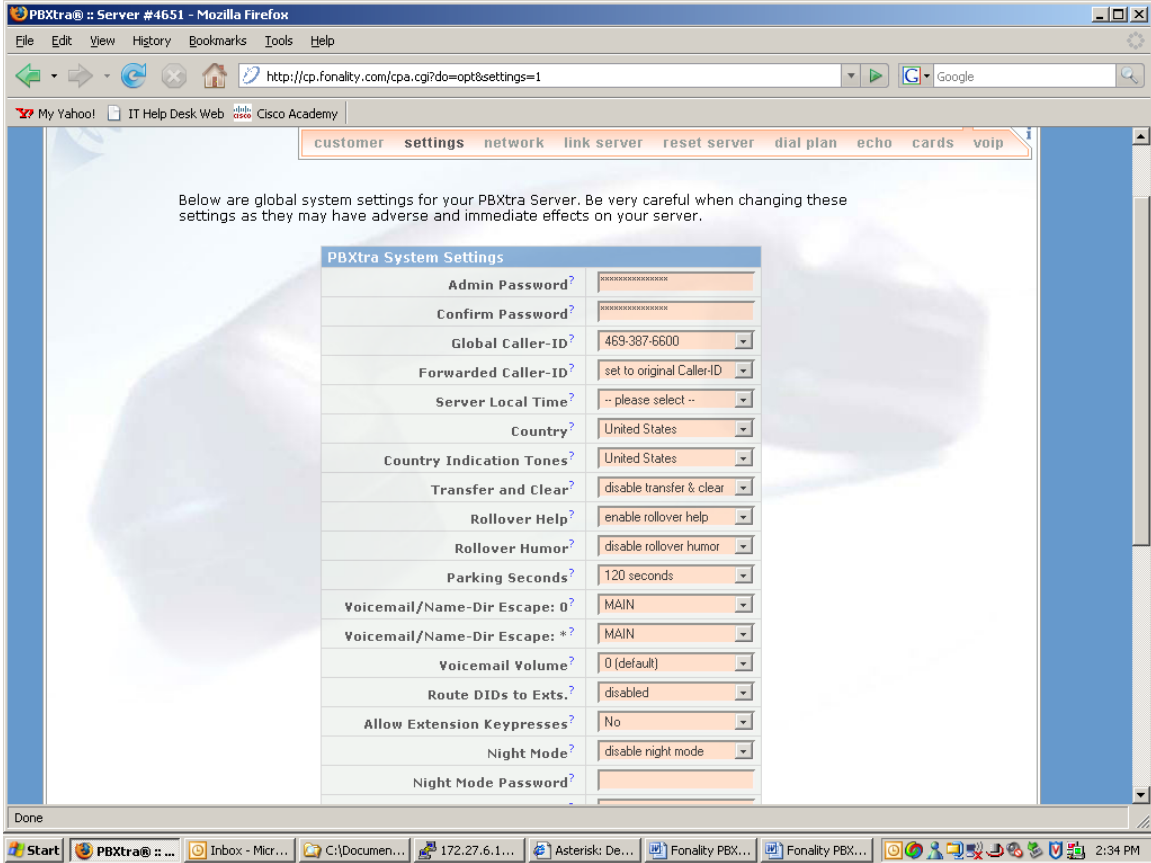
DNS Settings

Dynamic DNS?	Static
Primary DNS?	172.27.0.208
Secondary DNS?	10.76.107.8

Below the settings is an 'Apply All Changes' button.

The Windows taskbar at the bottom shows the Start button, several open applications including 'PBXtra@ :: ...', 'Inbox - Micr...', 'C:\Documen...', '172.27.6.1...', 'Asterisk: De...', 'Fonality PBX...', and 'Fonality PBX...', and the system tray with the time 2:33 PM.

6. Global System settings



7. Dial Plan configuration

[AutoAnswer](#) | [Extensions](#) | [A.C.D.](#) | [Reporting](#) | [Status](#) | **[Options](#)** | [logout](#)

[customer](#) | [settings](#) | [network](#) | [link server](#) | [reset server](#) | **[dial plan](#)** | [echo](#) | [cards](#) | [voip](#)

Add a new Dial Plan Entry below. You may have up to three routes per Dial Plan Entry.

Add Dial Plan Entry

Dial String?	Description?	Type?	Route?	Strip Digits?	Prepend?
9? +		toll free	1st: VoIP: xo	1 (default)	
			2nd: -- none (optional) --	1 (default)	
			3rd: -- none (optional) --	1 (default)	

Legend: n=any single digit other than 1 or 0
 x=any single digit, including 1 & 0
 .=0 or more digits, up to any length

Showing 10 Dial Plan Entries

Delete	Dial String	Description	Type	Primary Route	Strip	Prepend
	9 + nxxxxxx		local call	VoIP: xo	1	1972
	9 + nxxxxxxx	Standard long di...	local call	VoIP: xo	1	
	9 + 11	Emergency 911	local call	PRI T1 1-23 (23)	0	
	9 + 411		local call	VoIP: xo	1	
	9 + 611		local call	VoIP: xo	1	
	9 + 0	Operator call	local call	VoIP: xo	1	
	9 + 00	Operator call	local call	VoIP: xo	1	
	9 + 0nxxxxxxx	Operator call	local call	VoIP: xo	1	
	9 + 1nxxxxxxx	Standard long di...	long distance	VoIP: xo	1	
	9 + 011.		international	VoIP: xo	1	

8. VOIP configuration

Welcome to your VoIP Accounts area. From this screen you will be able to see your existing VoIP Accounts as well as add new VoIP Accounts - now of type SIP or IAX2!

Add VoIP Account

Route Name? Provider? SIP

Username? Register?

Password? Server?

Advanced

From User? Direction?

From Domain? TOS?

Allowed IPs? Authentication?

Disallowed IPs? Transfer?

Qualify? Reinvite?

Codec 1? NAT?

Codec 2? Insecure?

Codec 3? DTMF Mode?

Advanced :: Register String / RSA Key

Register String?

RSA Key?

Showing 1 VoIP Account

Delete	Name	Provider	Direction	Type	Host	Status
<input type="button" value="X"/>	xo	other	both	SIP	205.158.163.138	N/A