PRACTICAL EXPERIENCE WITH TRIXBOX

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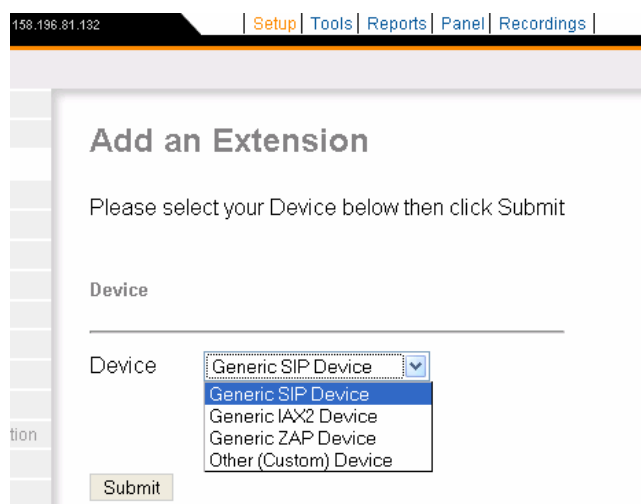miroslav.voznak@vsb.cz , brastmanghandi@hotmail.com , a30304@alunos.det.ua.pt

Briefing: *Trixbox is a business class IP PBX system based on Digum's Asterisk Open Source PBX Software. The trixbox package makes it fast and simple to install and configure any PBX system. With unlimited extensions, voicemail, music on hold, call parking, and many other features, trixbox brings big business PBX features to small and medium businesses. Whether there's the need for a few analog lines or high density T1/E1 PRI circuits, trixbox provides all the functionality a business needs.*

Devices Configuration

The first step was to install Trixbox in one computer; turning it into the server of our network. The next step was the configuration of the files **zaptel.conf** and **zapata.conf**. It is now possible to access the server and configure our terminals by IP address. For this we followed the instructions in trixbox.org, quick install guide;

Connecting by internet to the server IP address, in this case 158.196.81.132, entering the right password, we could go to Asterisk options FreePBX, and then to setup mode. Here we can create or remove our extensions and then connect some VoIP device to our network.

The next step is to choose and submit the device. Here we are able to configure it (Display name, password, mailbox and fax configurations).

Now we only need to connect our device to internet, change the domain and number of it (IP address of the server) and restart. These are the minimum configuration that we need to change. After we restart the device we should be able to phone to any device connected to our network. As hard phones we used Grandstream and Siemens IP phones.

We can also work with other kind of devices, programs that replaced them, called soft phones. We only used free soft phones, respectively:
-   X-lite (working only with SIP protocol)
-   Idefisk (IAX2 protocol)
-   Firefly (works with both protocols).

Like the other kind of devices we need to configure name and domain of the device. If it's working correctly it should appear something very similar to that in the images below.

In the example we are phoning from Firefly device to X-lite device. We can see that it is working by the small box "incoming call". It appears the extension "fireflyIAX2_Luis".

Now we are doing the same but with Idefisk (IAX2 protocol) and X-lite (SIP protocol). We can also see that they are working well.

We only noticed one problem, when we tried to connect Idefisk and Firefly with IAX2 protocol it didn't work. We can't use for the same IP address (same computer) two equal protocols.

Asterisk Functionalities

First we log in at https://158.196.81.132/maint/ introducing the administrator password. To configure the server, we access Asterisk → FreePBX → Setup. The most important options are:

1) *Extensions* – for adding and manipulating extensions
The relevant settings are the Display Name - the caller's ID name -, the Secret - our extension's password - and the Context - this should be set to **from-internal** so that we are able to call outside the internal network.
Other interesting features we can configure are the Recording Incoming and Outgoing - we set them to **always** on this case, so every call is recorded -, and the Voicemail - we set its status to enable, including the mail address to which the voicemails are sent too. For this there are some sub-options:´

- **PLAY CID** – allows to read back caller's telephone number prior to playing the incoming message, after announcing the date and time the message was left.

- PLAY ENVELOPE - Envelope controls whether or not the voicemail system will play the message envelope (date/time) before playing the voicemail message. This settng does not affect the operation of the envelope option in the advanced voicemail menu.

- DELETE VMAIL -. If set to "yes" the message will be deleted from the voicemailbox (after having been emailed). Allows a user to receive his voicemail via email alone, rather than having the voicemail able to be retrieved from the Webinterface or the Extension handset. The attach voicemail to email must be set to yes, otherwise the messages are lost forever.
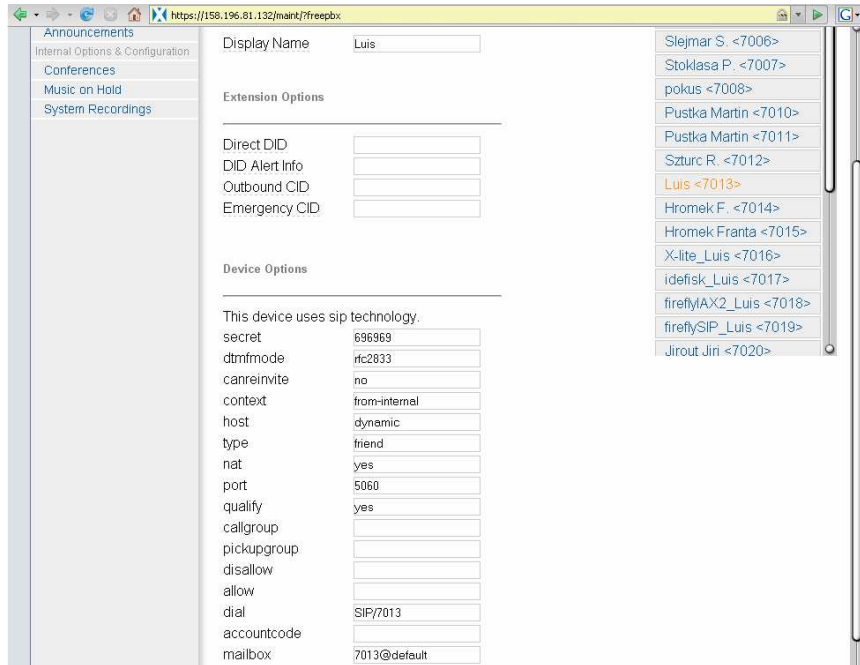
2) *Feature Codes* – for call forwarding, call waiting, recordings, voicemail and other codes.
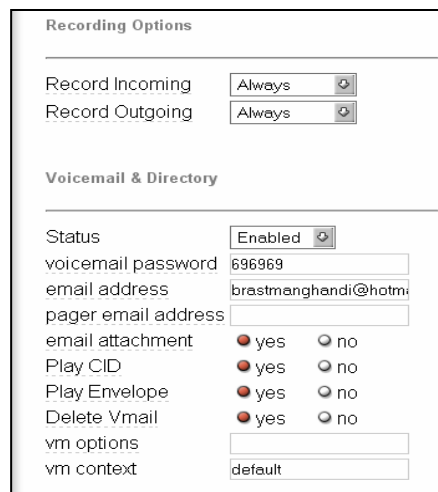
3) *Outbound Routes* – here we can specificate the type of number we type for local calls, long distance calls, etc.

For example, configuring our university PBX route, we choose the Dial Pattern and Trunk Sequence as follows:

Dial Pattern - A Dial Pattern is a unique set of digits that will select the respective trunk. Enter one dial pattern per line.
Rules:
X matches any digit from 0-9

Z   matches any digit from 1-9
N   matches any digit from 2-9
[1237-9] matches any digit or letter in the brackets (in this example, 1,2,3,7,8,9)
.   wildcard, matches one or more characters
| separates a dialing prefix from the number (for example, 9|NXXXXXX would match when we dialed "95551234" but would only pass "5551234" to the trunks)

Trunk Sequence - The Trunk Sequence controls the order of trunks that will be used when the above Dial Patterns are matched.

For Dial Patterns that match long distance numbers, for example, you'd want to pick the cheapest routes for long distance (ie, VoIP trunks first) followed by more expensive routes (POTS lines).

    4)   Trunks

Note: We only tested outgoing calls, so no incoming trunk settings are modified. The following setting was altered:

Peer Details - Modify the default PEER connection parameters for your VoIP provider.

Next we show the SIP trunk "cesnet" specifications:

    -   Outbound Caller ID = VSB-TUO

5) Conferences

In this option it is possible to make multi-users conversations by calling the conference number and inserting the corresponding PIN number.

It's a very powerful tool as it permits to join an unlimited number of users together.

Recordings

In FreePBX → Recordings, it is possible to consult the recorded calls and its details (duration, time of call, destination, source), and also listen to them, by entering the extension number and corresponding password.

Voice Mail
Call Monitor
Call Forwarding

Call Monitor for Luis (7013)

[Search]

| | delete | duration | ignore | | | Results 1 - 15 of 149 |

Voicemail
Call Monitor
Help

Configurações
Logout

select: all none

	Date ▼	Time	Caller ID	Source	Destination	Context	Duration	Monitor
☐	2007-03-19	10:35:27	"Luis" <7013>	7013	00420775097349	from-internal	44 sec	play
☐	2007-03-19	10:34:03	"Luis" <7013>	7013	7027	from-internal	7 sec	
☐	2007-03-19	09:08:18	0737968287	0737968287	7013	default	11 sec	play
☐	2007-03-19	04:57:27	"Luis" <7013>	7013	351233910439	from-internal	16 sec	play
☐	2007-03-19	04:39:01	"Luis" <7013>	7013	7027	from-internal	8 sec	
☐	2007-03-19	04:15:33	"Luis" <7013>	7013	351210329168	from-internal	10 sec	play
☐	2007-03-19	04:14:50	"Luis" <7013>	7013	00420775097342	from-internal	10 sec	play
☐	2007-03-19	04:02:38	"Luis" <7013>	7013	7027	from-internal	9 sec	
☐	2007-03-19	04:00:32	"device" <7013>	7013	s	from-internal	0 sec	
☐	2007-03-19	03:55:43	"Luis" <7013>	7013	351210329168	from-internal	10 sec	play
☐	2007-03-19	03:42:22	"Luis" <7013>	7013	7027	from-internal	9 sec	
☐	2007-03-19	03:41:05	"device" <7013>	7013	s	from-internal	0 sec	
☐	2007-03-19	03:39:15	"SjLuis" <7027>	7027	7013	from-internal	26 sec	play

Different Domains Test

We did also a call test between our domain and a proxy server from iptel. We needed to create an account with a number starting with 8 as log in. This fact would be important for the dial pattern configuration of the iptel outgoing route too.

PDA Call

It was necessary to download a software to allow the of a proxy server. We did the download of a free version in http://www.sjlabs.com. The PDA in use was Nokia E61.

Configuration:
Main Menu → Connection → SIP Settings
When creating a new profile, we need to fill Profile name (Trixboxprace), service profile (IETF), default access point Vsb-tuo and public user name (sip:7000@158.196.81.132). We also set Use Compression to **No**, Registration to **Always On** and Use Security to **No**.

Next step was Server Registration, as follows:

To finish, the call. We called from PDA (extension 7000) to one VSB extension.

References:

[1]   Trixbox Without Tears, The Dumb-Me guide, online, http://dumbme.voipeye.com.au/trixbox/

[2]   Meggelen,J.-Smith,J.:Asterisk™ - The Future of Telephony, O'Reilly Media, Inc. ISBN: 0-596-00962-3

[3]   Baroňák,I., Halás,M.: SIP Protocol – the Future of IP Telephony. In: 27th International Conference „TSP 2004". Brno, September 2-3, 2004, ISBN 80-214-2684-5.