ElastixEasy

By Haamed Kouhfallah

PDF Guide by Haamed, 23 Jan 2012

ELASTIX

EASY

[Haamed Kouhfallah]

ElastixEasy is a reference book for who interested in Asterisk and Elastix and typically the voip technology. I'd like to dedicate this book to all children suffering from pediatric cancer. This book is free, though, if you find it useful; you can donate to your local pediatric cancer foundations. Thanks in advanced.

ELastix Easy

For ELastix 2.x and Freepbx 2.x

http://www.voip-iran.com

by

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Last Update: **18 Jan 2012**

About the author [**Haamed** Kouhfallah]:

His experience in the field of VoIP includes:

- Elastix Engineering (ECE) from Palosanto Group;
- Community member of Elastix website on June 2009 and among the top ten members of Elastix.org in 2009;
- Community member of Elastix website on December 2010 and among the top ten members of Elastix.org in 2010;
- Produce Vaak telephone system that is the translation of Elastix system into Persian, he also provide package of Persian converter of Elastix per each copy provided from Elastix website;
- Author of Elastix in Persian as the only Persian reference and Free PBX which is introduced as reference book in www.elastix.org ;
- conducting training courses of Elastix and VOIP in Iran;
- Official trainer of Elastix Engineering (ECE & ECT) in Iran.

1. Introduction

1.1 About Asterisk

Asterisk is open source software used for unified communications which was first created in 1999 by Mark Spencer, the managing director of Digium Company, based on GNU/GPL license. Asterisk is a step towards generalizing various communication methods based on computer & local networks for audio-visual telecommunications & related applications such as IM, Call/Contact Centre & etc.

As IP based computer network communications grows, Asterisk faces ever increasing success. Being free for all when compared to the enormous prices of the current brands in the market, in addition to having various potential capabilities with adequate quality, standard protocols, not being limited/depended to a particular brand of software or hardware, easy to install & operate, the sheer size of its third party developer community & most importantly, unified voice (whether voice or telephone), visual & data services have made Asterisk as a soft switch to become one the effective & dynamic components of the next generation of communication.

Asterisk is based on C programming language and is loaded in various operating systems such as Linux NetBSD, UNIX, Solaris, Mac OSX, FreeBSD, and OpenBSD. In addition, other versions of Asterisk can be installed in windows platform. Although by using computers, common servers and calculating the power of system (CPU/RAM) based on the number of users, Asterisk services can become operational, but the popularity and variety of its services prompts many manufacturers to use combined platform of Linux and Asterisk to make unified communications equipment in different scales. From very simple, efficient and cheap equipment in SOHO & SMB scale to complex designs with large number of users in Enterprise environments, production of such tools is very easy and simple because their software is available and it is enough to facilitate the operation of system by designing appropriate interface and web based. In more complex samples, because of open source of Linux and Asterisk, changing the source of software can be possible for better performance.

As multipurpose software which is based on information networks the best thing to do is to designing a network (QoS, Redundancy, Traffic Management& planning) and using its hardware appropriately in SMB & Enterprise environments. Thus, Asterisk should have these requirements whether it is used in simple application like phone center (IPBX), more complex like video conference and call/contact center or in unification with software such as office automation, ERP and etc.

Contrary to many, not only Asterisk and basically soft switch idea, audio and video communications based on network application is not in conflict with traditional view of telecommunications but also it has complement and developer role. Although Asterisk is popular, its communication based on computer network (Video Conferencing, IP Technology, VoIP, etc...) is cheap and extended, but justification with traditional structure, generally TDM, is not forgotten in Asterisk and more importantly communication media has no effect in its operation. Set up Asterisk based on IP Based equipment is easier and cheaper but justification with older technologies should be considered as well. Security and reliability of operation in soft switch systems and Asterisk in compare with traditional communicative systems is a reason of conflict between soft switch and traditional ideas. These two articles should be discussed separately but at the end solution of an Asterisk system for security and operation reliability is shortly expressed.

Encryption of communication is the best way which line tapping and having fast computers cannot decode it. In addition to common ways, proprietary protocols can be made in encryption of communication. This is possible because of capability of Linux operating system and for providing the communications security between systems components based on Asterisk both common encryption and propriety protocols can be used. Beside this Linux is an appropriate firewall that can guarantee the security coefficient of accessing the Asterisk services to the high level beside other firewalls.

Most of the typical features of Asterisk system, which is installed in Linux platform, are actually taken from Linux operating systems. High power capabilities such as Clustering & HA (High Availability) of Linux guarantee the operation reliability of soft switch system based on Asterisk. Besides, hardware redundancy like power supply with redundancy of computer network in links, equipment, protocols and etc… cause that Asterisk be in the same level with TDM Based systems. So Asterisk is a way toward presenting next generation services in divers scales (Enterprise,SMB,SOHO). A way which leads to unified communications, innovation and simply providing extensive range of audio and video services and also Fixed Mobile Convergence Enterprise in organizational interactions.

1.2 About Elastix

Elastix is a collection of best open source programs and tools which are combined together and finally create a comprehensive IP PBX. It is designed properly and gives you a PBX system that can compete with others, not only because of PBX part but also because it is capable of creating a powerful system with other products and programs.

The most important parts of Elastix:

- Asterisk: as the core PBX (Digium’s most well-known product)

- VTigerCRM and SugarCRM: as a communication system with customers
- A2Billing: program to pay bills of Asterisk
- Flash operator panel: operator console which is like monitor display.
- Hylafax: a software fax system
- Openfire: a server with dialogue system, sending text and telephone network
- Conferencing: is an controlling devise
- freeBPX:an application tool for Elastix
- A report system: part of Elastix that provide CD report
- OSLEC: it is a software that remove echo sound
- Postfix: a popular mail server
- Round cube webmail: an interface for using web based mail services
- CentOS: it is a version of Linux, Redhat with free support, and one copy of Centos will be released by each copy of Redhat. They both were supported and produced by different companies and in many cases big and small companies uses these for manufacturing their products. Elastix producers compile a web interface to access the programs which seems to be complete. Also Elastix company provide a software for reporting, diagnosing the hardware, network setting, module of updating software, backup module, managing users and other modules.

1.3 About freepbx

When Asterisk suddenly became popular, many company started to invest on it. Asterisk does not have graphical environment for easy set up so companies started to design a graphical environment for that. Some of them were free and open and others commercial. One of the graphical environments is freepbx which is popular because of providing many applications for users. Some companies that produce ISO for Asterisk, use this program for managing and setting up the Asterisk such as Trixbox, Elastix, Asterisk now…

The official website of this program is:

http://www.freepbx.org

1.4 Common terms/phrases in VoIP

There are some common phrases in VoIP technology

Definition		abbreviation
adaptor of analogue telephone to IP based telephone	Analogue Telephone Adapter	ATA
Port of RJ11 is like a telephone and equipment attached to it should be capable of receiving alarm signal answering and ending a call	Foreign Exchange Office	FXO
Open line which is attached to RJ11 can provide an open line and power of FXS in an analogue line to a telephone	Foreign Exchange Station	FXS
Network based on honeycomb model	Global System for Mobile	GSM

which is the most common standard for mobile equipment	Communication	
Proprietary protocols of Asterisk with RFC5456	Inter Asterisk eXchange protocols(version 2)	LAX (IAX2)
Internet standardization committee	Internet Engineering Task Force	IETF
Committee of the international telecommunication	International Telecommunications union	ITU
It refers to automated telephone answering system	Interactive Voice Response	IVR
It is an interface between local network and an internet. NAT allows a device in internet to work like a router	Network Address Translator	NAT
It refers to local telephone centers. The main task of PBX is exposure between one or some telephone line and some of users and also dividing bilateral contacts	Private (Automated) Branch Exchange	PBX (PABX)

Digital display of an analogue signals is used as an interval 0 & 1 and standard in digital audio and video	Pulse Code Modulation	PCM
Telecommunication public network which also is called fixed telephone network	Public Switched Telephone Network	PSTN
It refers to networks based on (packet-switched) depends on mechanism of controlling powers to access to appropriate services	Quality of Services	QoS
Published note about IEFTF which explained methods, approved researches and innovations about internet and systems attached to it, has unique number	Request For Comment	RFC
Standard of audio and video formats in internet RFC3550	Real-time Transport Protocol	RTP
A dedicated protocol of Cisco is used to control the network	Skinny Call Control Protocol	SCCP

between work stations and Call Manager of Cisco. At first it was developed by Cilsios company and now its owner is Cisco		
It is protocol of an audio signaling based on VoIP	Session Initiation Protocol	SIP
It is collection of signaling phony protocols which is used in set up most PSTN	Signaling System7	SS7

2. VoIP Hardwares

For installing Elastix telephony system, you may just need a computer but for communicating with other telecommunication systems or more comfortable working with Elastix some hardware may be needed. These hardwares are the most useful ones and can be divided into 3 categories as follows:

2.1 Access methods and using the services of VoIP

In order to use extension registered on Elastix, some equipment is needed to register your extension number on it and use it. Generally communicative ways to VoIP is as follows:

- Analogue telephone adapter(ATA)

This adaptor is known to Gateway. They have network port and placed on the network with IP. On the other hand they have FXS port which can be used by connecting analogue phone and extension number registered on it. These Gateways can have several ports and it is possible to register an extension number on each port and use it. In the other word it can be stated that the main task of Gateways is turning an analogue phone to an IP phone.

Adaptor of a phone to IPPhone

- IPPhone :

These are similar to ordinary phones but ordinary phone is connected to RJ11 and in IPPhone it is connected to RJ45 (like network connection) and all the required softwares and hardwares are Built-in. nowadays this type of devise is one of the affordable and user-friendly. An SNOM phone is shown below:

An IPPhone

- Softphone:

It is software but included in this category. For registering an extension number you can use a softphone. The biggest advantage is that it does not cost to you (if you use free versions) and it is easy to use. Eyebeam softphone is shown below which can be communicated visually.

Eyebeam softphone

2.2 Computer

In order to install each software you need a computer, Elastix is collection of the software so we need a computer either. The main question for bigginers and profesionals is that what properties should we consider for the sysytem we want to set up? The answer is different becouse in VoIP, set up and used applications is different. So generally it can be said that you can chose your system by expiriance and sometimes you need to put your system in operational enviroment to see the reproductive rate of system. Some system and reproductive rate of them is shown bellow:

Name	City line	facilities	features
Small	City card(4line), about 20 extension number	Voicemail,without recording conversation	1GB RAM, Dual Core 2.6 CPU, 128 GB HDD
Average	E1 line(30 line), about 50 extension number	Voicemail, without recording conversation	1GB RAM, Core2Duo 2.8 CPU, 128GB HDD
Average	2line E1 (60 line), about 100 extension number	Voicemail, without recording conversation	2GB RAM, Core2Quad 2.8 CPU, 256 GB HDD
Larg	4line E1(120 Line), about 300 extension number	Voicemail, without recording conversation	4GB RAM,2*Xeon 2.8 CPU, 512 GB HDD

There are some tips about chossing a hardware which is totally experimental:

- Recording will put extra pressure on CPU. If you want many extension numbers or to record all of your lines ,you may need the second server.
- be carefull in using voicemail! This feature will be heavy for your system either so don't active voicemail for the extension numbers which don't need.
- If you want to provide a system, Gigabyte motherboard are better options especially for installing Linux on them.

- If you use E1 cards,especially for 2 and more E1 cards, use Echo Canceller cards with them. They are effective in voice quality and in reducing extra pressure on CPU.

2.3 Telephony card

Telephone cards usually used for Elastix link with PSTN city lines. Any telephone card cannot be used for this purpose. It should be Asterisk compatible.

3. Installation

3.1 Prerequisites

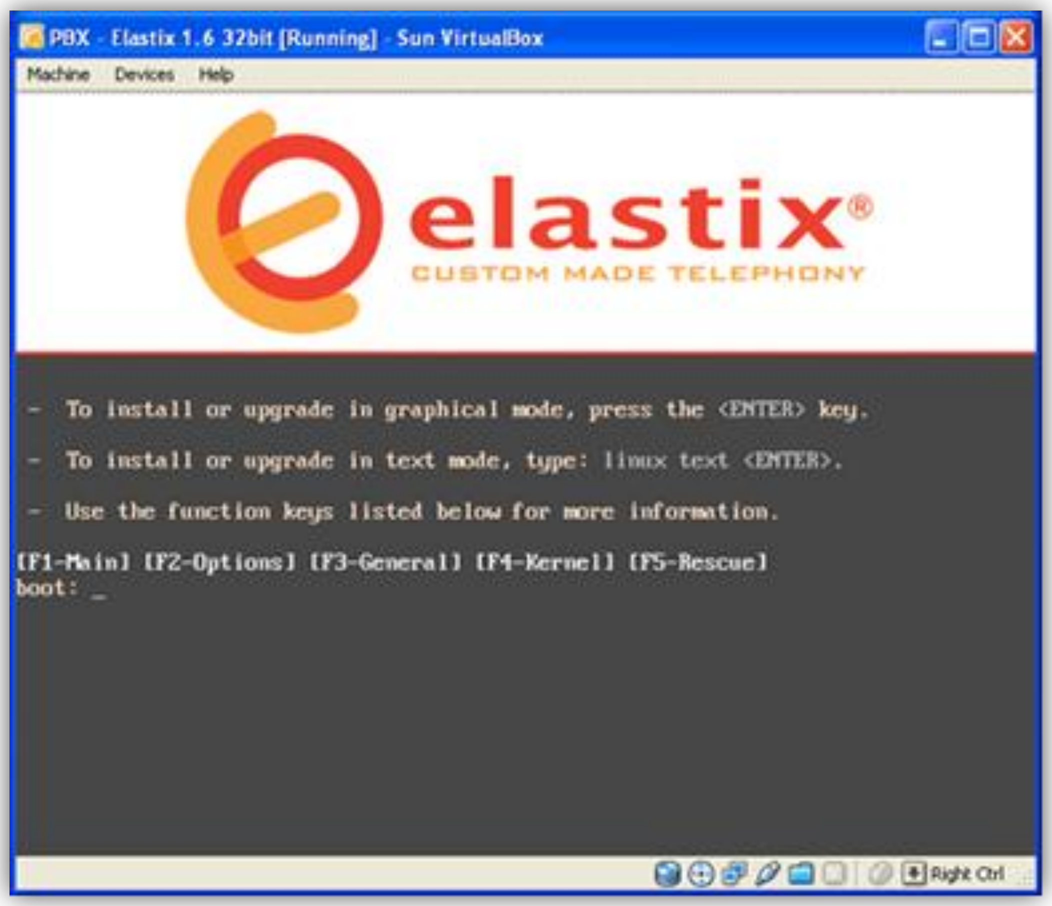
As it was mentioned for installing Elastix and dialing extention just a computer is needed. But bear in mind that installing Elastix will format your computer hard. That computer dedicate to this purpose and for communicating you need an IPphone which softphones can be used either. In this article instalation of Eyebeam as a IPphone is explained.

3.2 Installing ISO

For receiving the latest version of Elastix ISO go to the following address:
http://sourceforge.net/projects/Elastix/files

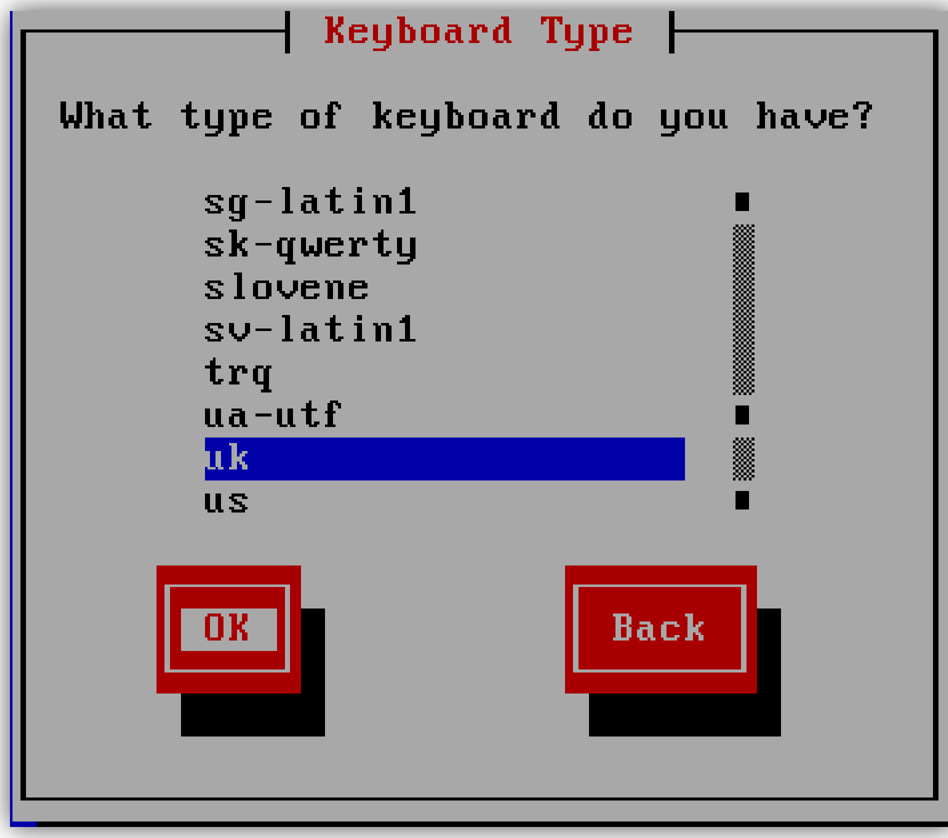
for receiving persian Elastix see following address:
http://www.voip-iran.com

installing ISO Elastix is like Trixbox and other distributions of Linux. For installing, download an ISO file from the site and then with nero or MagicISO or etc write like a Burn Image on CD and install it.
Attention: this will format your disk,so be sure you don't have any important information before instalation.
First put Elastix cd on cdrom and boot your system. The first image which shows installation displayed abit later.

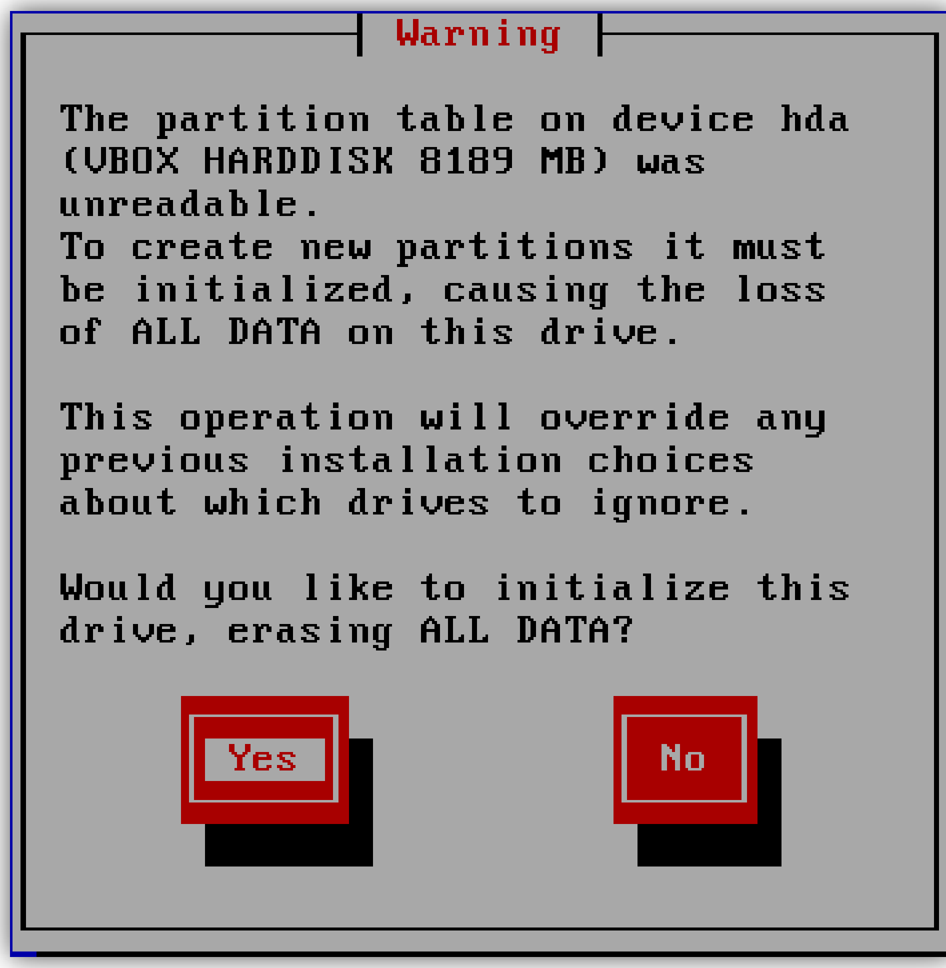


Installation start with pressing enter, wait for loading files to be complete, when installer start image of choosing language will appear.

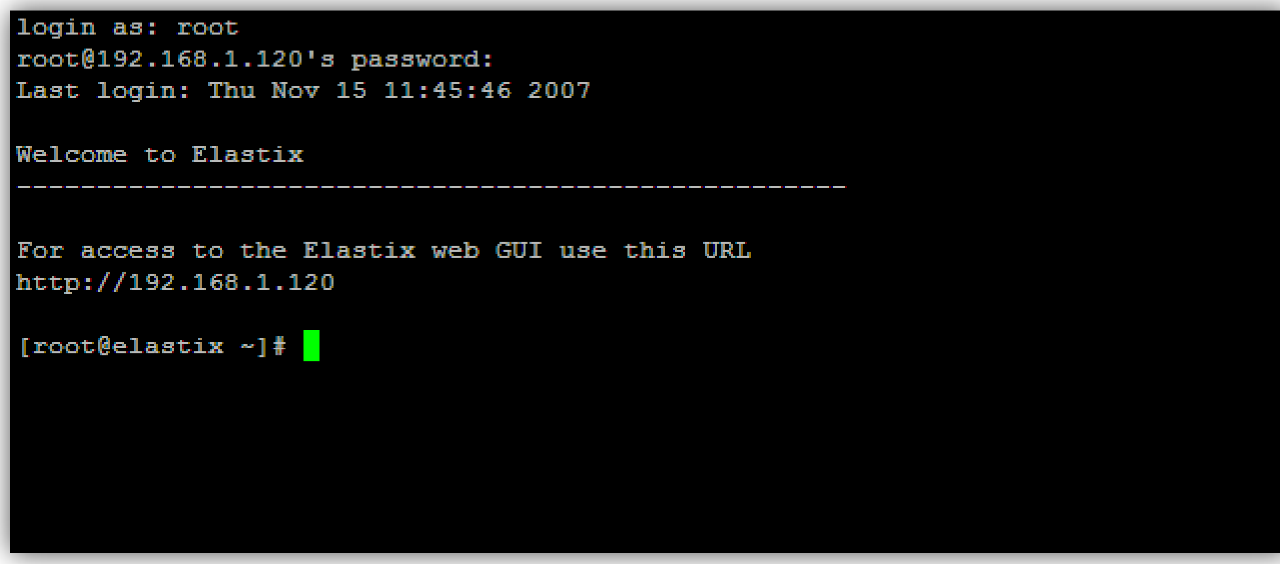
Choose your language and keyboard language.



You choosed the language of your system. Then you are welcome to instalation process. If it was new installation, and there was nothing on your drivers, you don’t receive this message. Only when there is something on the hard and system cannot recognize it,this image will be shown.



The next window is driver setting. If you want to install a new system, it is better to click on remove all partitions, then yes and move to the next level.

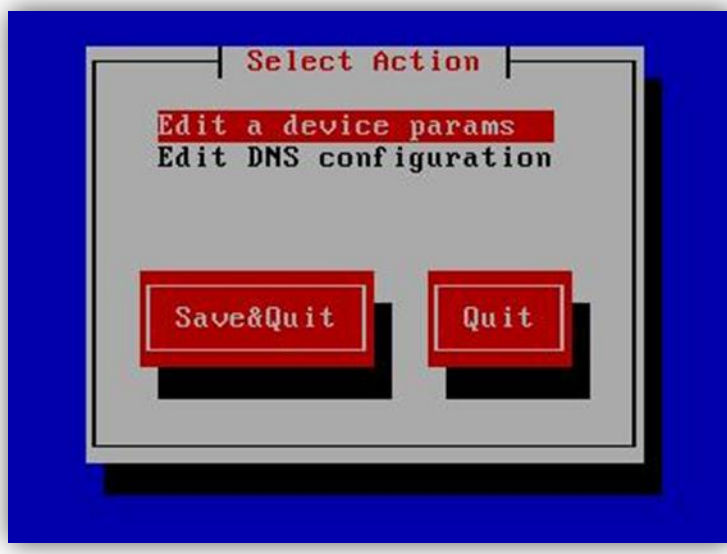


Attention: if you want to install Elastix just for testing, it is better to use Sun Virtual Box or something like that. The only restriction is that you can not use TDM400 card in Virtual Machine otherwise you can have all the efficincy of VoIP. Bear in mind that network interface should be bridge.

3.3 Network configuration

After installing Elastix, server get IP from DHCP as a difult. The IP is shown after entering to the Linux enviroment or with ifconfig eth0, you can see the IP. Now you can access to the login page of Elastix with writting IP in addressbar, but if you want to give IP manually to Linux, there are different ways which the easiest way for begginers is shown as follow:

```
system-config-networking
```



Chosse the network card for giving IP.

Save the changes and exit. After that you can conneted to UI from any url with giving IP server. By entering

Username: admin

Password: "Your entered pass on Istallation"

You can log in. (this username and password is for Elastix version 1.6 and before that. For Elastix version 2 or after that you should enter admin and your password for installation).

3.4 The default usernames

Elastix version 2 receive passwords of Freepbx, Database, Vtiger, a2billing, Elastix web during installation while passwords of Elastix version 1.6 or other programs is as follow:

Web graphical enviroment

Username: admin

Password: palosanto

```
Freepbx:

User: admin

Password: admin


relationship with customer Sugar CRM:

username: admin

password: password


calculating telephone A2billing:

username: admin

password: mypassword


Vitger:

Username: admin
```

Password: admin

Attention: change psswords after installing Elastix as follow:

For changing admin password of Freepbx first respectively go to the graphical enviroment of Elastix, menue of call center(pbx), pbx configuration and unembedded freepbx. In this way you will enter to the freepbx enviroment. Go to the set up, basic, administrators and change the admin password of freepbx. For other 2 program you need to change from internal menue of the program.

3.5 Accessing to the graphical enviroment

for seeing graphical enviroment (web) if you set the ip server correctly, you just need to enter the ip in browser. You can do it with any computer which is cennected to server through network. It is good to use firefox for seeing the web.

3.6 changing the admin password

in order to change the admin passwod of graphical enviroment of web, go to the System and User Management.

4. Useful programs

4.1 Webmin

Webmin sofware is Swiss knife of Linux!! It means by installing this program you can easily configure Linux but installation of this program on Redhat (like operating systems of Elastix and Trixbox) is as follow:

1- First download the file.
 http://www.webmin.com/rpm.html

Attention: those who live abroad don't have any problem with downloading from Sourceforge servers and can directly receive it from Linux.

```
wget
http://internap.dl.sourceforge.net/sourceforge/webadmin/webmin-1.710-1.noarch.rpm
```

2- Copy the file on /tmp
3- Go to tmp from console:
 Cd /tmp
4- Enter the following command:
 Rpm –i webmin-1.710.-1.noarch.rpm

The last part of the file is copy so if the version of webmin was different, the name entered would be different either. After that process the webmin is installed and for accessing you should enter the following on your browser:

Http://YOUR-SERVERIPADDRESS:10000

4.2 WinSCP

Maybe you are unfamiliar with Linux or maybe you are familiar but you are not patient enough to work with that. By the way WinSCP is one of the best projects of Sourceforge and its task is accessing to the Linux files. It will be installed on operating system of your windows and Linux environment can be easily seen, especially it is good for copying the file on Linux or changing the files. Professionally it can be said that it is a FTP & SFTP client for windows.

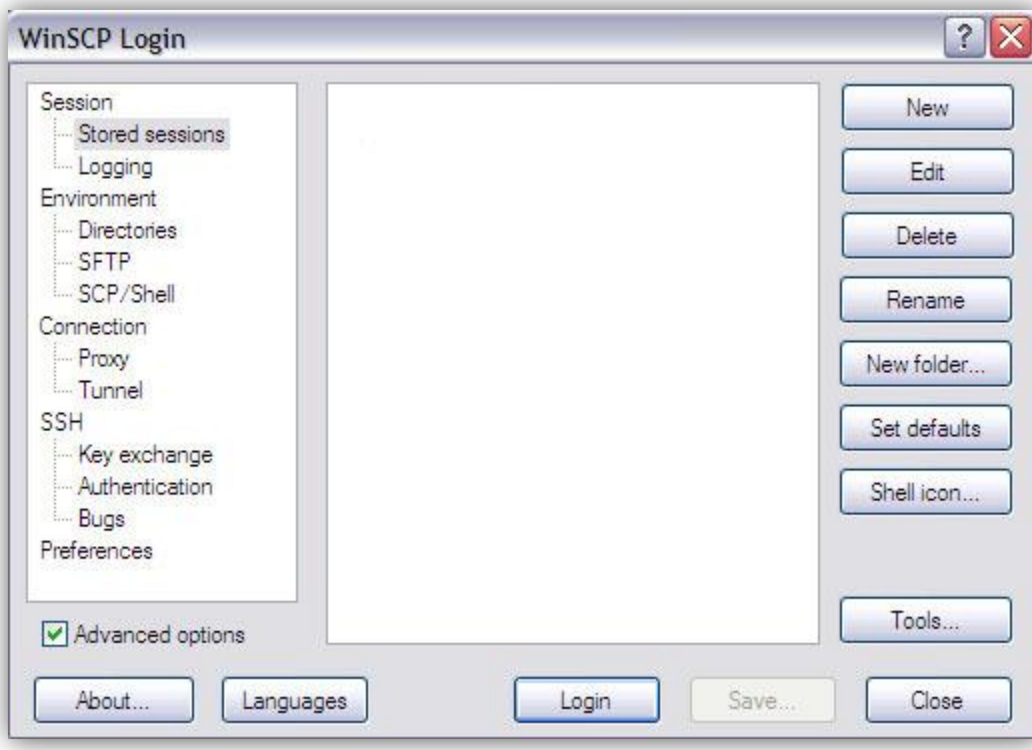
Those who live abroad can receive the file from Sourceforge.

http://sourceforge.net/projects/winscp/files/winSCP/4.2.8/winscp428setup.exe/download

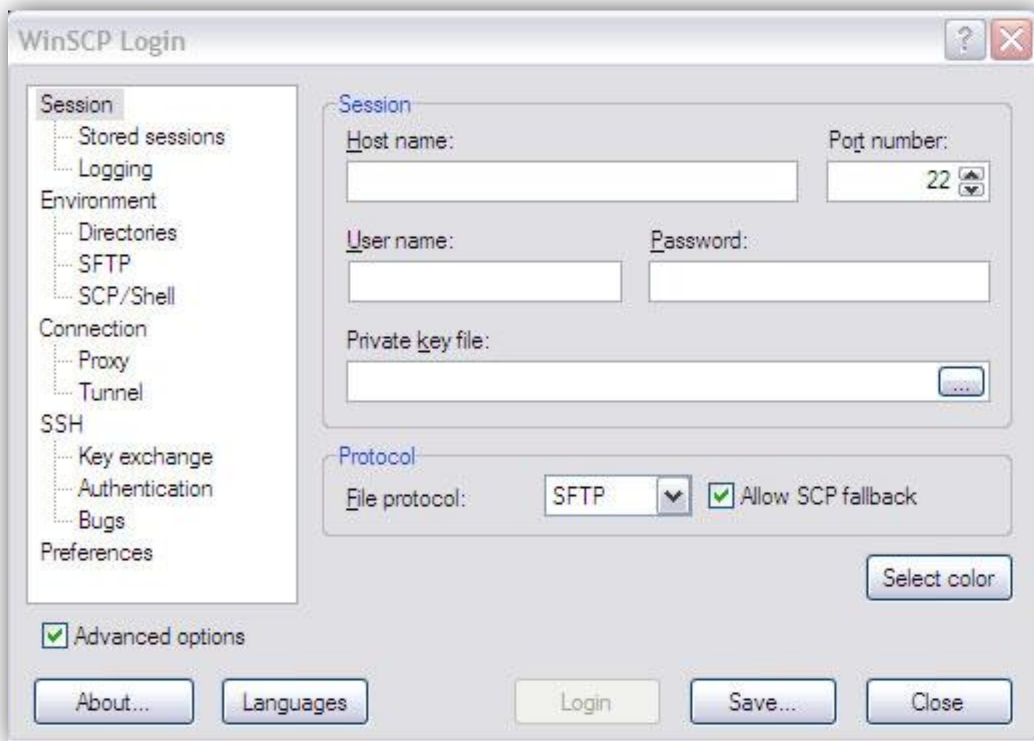
Those who live in Iran can see the Utilities of this link.

https://sourceforge.net/projects/vaak/files

You won't have any problem in installing it for sure. After that you should make a new Host for any Linux you want to connect. The page below is shown after installation:



Now by clicking on NEW the page below will be open. Linux features should be written in this paged.



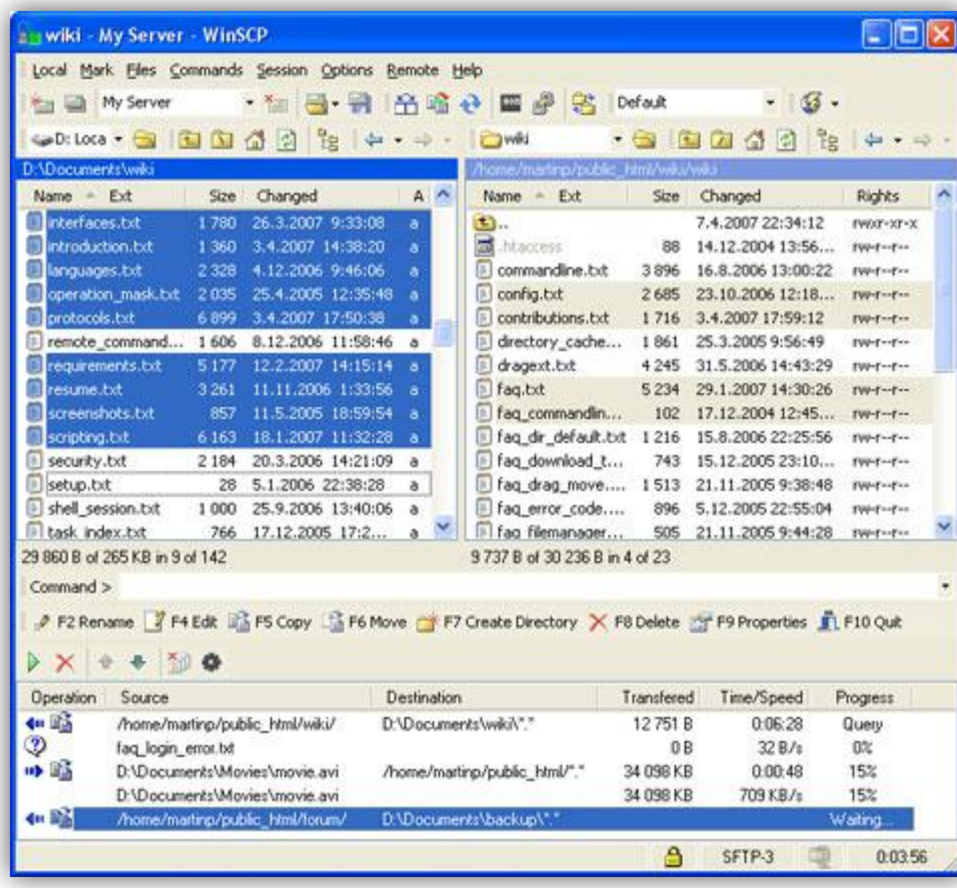
In this page you should fill 3parts:

Host name: IP address of your Linux server

User name: user code for Linux is usually root

Password: password of user

When you make a Host, by clicking on it you will be connected to Linux.


wiki - My Server - WinSCP
Local Mark Files Commands Session Options Remote Help
My Server
Default
D: Loca
wiki
D:\Documents\wiki
/home/martinp/public_html/wiki/wiki
Name Ext Size Changed A
interfaces.txt 1 780 26.3.2007 9:33:08 a
introduction.txt 1 360 3.4.2007 14:38:20 a
languages.txt 2 328 4.12.2006 9:46:06 a
operation_mask.txt 2 035 25.4.2005 12:35:48 a
protocols.txt 6 899 3.4.2007 17:50:38 a
remote_command... 1 606 8.12.2006 11:58:46 a
requirements.txt 5 177 12.2.2007 14:15:14 a
resume.txt 3 261 11.11.2006 1:33:56 a
screenshots.txt 857 11.5.2005 18:59:54 a
scripting.txt 6 163 18.1.2007 11:32:28 a
security.txt 2 184 20.3.2006 14:21:09 a
setup.txt 28 5.1.2006 22:38:28 a
shell_session.txt 1 000 25.9.2006 13:40:06 a
task_index.txt 766 17.12.2005 17:2... a
Name Ext Size Changed Rights
.. 7.4.2007 22:34:12 rwxr-xr-x
.htaccess 88 14.12.2004 13:56... rw-r--r--
commandline.txt 3 896 16.8.2006 13:00:22 rw-r--r--
config.txt 2 685 23.10.2006 12:18... rw-r--r--
contributions.txt 1 716 3.4.2007 17:59:12 rw-r--r--
directory_cache... 1 861 25.3.2005 9:56:49 rw-r--r--
dragext.txt 4 245 31.5.2006 14:43:29 rw-r--r--
faq.txt 5 234 29.1.2007 14:30:26 rw-r--r--
faq_commandlin... 102 17.12.2004 12:45... rw-r--r--
faq_dir_default.txt 1 216 15.8.2006 22:25:56 rw-r--r--
faq_download_t... 743 15.12.2005 23:10... rw-r--r--
faq_drag_move.... 1 513 21.11.2005 9:38:48 rw-r--r--
faq_error_code.... 896 5.12.2005 22:55:04 rw-r--r--
faq_filemanager... 505 21.11.2005 9:44:28 rw-r--r--
29 860 B of 265 KB in 9 of 142
9 737 B of 30 236 B in 4 of 23
Command >
F2 Rename F4 Edit F5 Copy F6 Move F7 Create Directory F8 Delete F9 Properties F10 Quit
Operation Source Destination Transfered Time/Speed Progress
/home/martinp/public_html/wiki/ D:\Documents\wiki*.* 12 751 B 0:06:28 Query
faq_login_error.txt 0 B 32 B/s 0%
D:\Documents\Movies\movie.avi /home/martinp/public_html/*.* 34 098 KB 0:00:48 15%
D:\Documents\Movies\movie.avi 34 098 KB 709 KB/s 15%
/home/martinp/public_html/forum/ D:\Documents\backup*.* Waiting...
SFTP-3
0:03:56

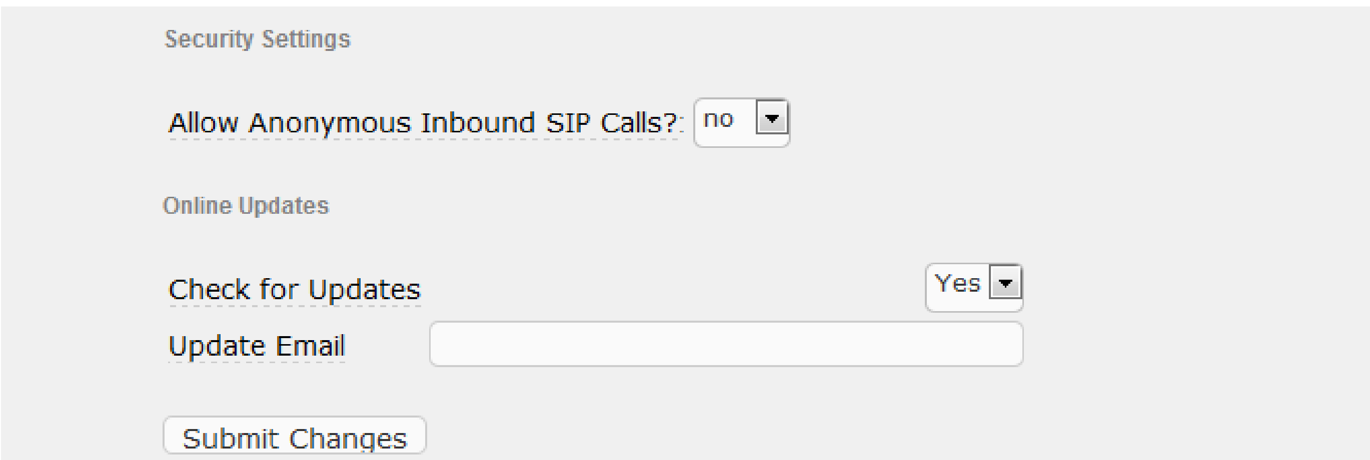
5. Telephony configuration (PBX)

5.1 General setting:

In this part you can change some of the general and main setting of Asterisk. Those who are beginners and they just want a simple and small telephony system, there is no need to change but it is good to know these features:

- Allow Anonymous Inbound SIP Calls:

The main part of this page is receiving calls without permission of SIP, which is good to change:

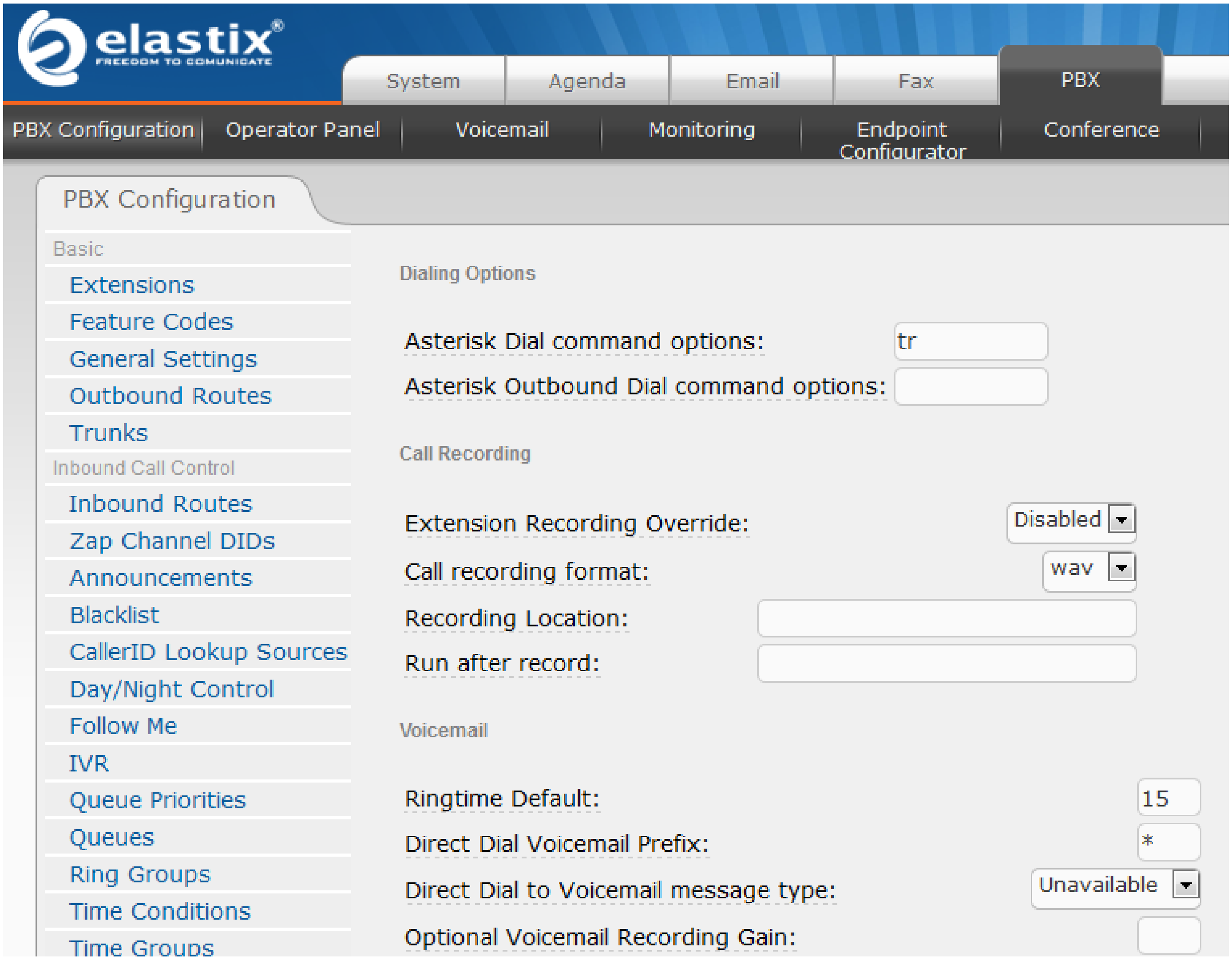


Allow Anonymous Inbound SIP Calls: yes

This item is “no” as the default and SIP inbound calls which are unknown are not accepted. To enhance the security, it is better to change it to “no” after testing the system.

- Dial Command Option:

In “Asterisk Dial Command Options” you can configure the Asterisk in a way to show desired performance, for instance if you want the caller hear the desired song instead of beep sound, change the option –r with –m (it is not recommended).



You can use these options if you want:

A(X)	Play an announcement for receiver and x is the played file
C	CDR reset the call
D	It allows the caller to dial one-digit extension number while he is waiting for answer. So if that number registered on context or EXITCONTEXT, it will be dialed.

D([called][:calling])	Certain DTMF will be sent to the receiver after answering the call and before the calls become bridged. “Called” of DTFM is sent to receiver and “calling” to the caller. Each 2 parameters can be used separately.
f	It forced the channel of caller to equal its caller ID with relative extension of this channel and for that uses Dialplan hint. For instance, some PSTN network doesn’t allow caller ID to have different numbers to the caller.
g	When destination channel cut, in order to Dialplan it continued for the present extension.
G(context^extent^pri)	If the call is answered, both side of the call will be transferred by identified priority. Registration of extension or context is optional and available extension will be used if they don’t register.
h	This allows the receiver to hang up with sending ‘*’DTMF.
H	This allows the caller to hang up with sending ‘*’DTMF.
i	If all the required channels are busy, it will jump by priority of n+101
L((x)[:y][:z])	It limit the call time to x millisecond. It warns when y millisecond remains and it repeated any z millisecond. Following

	variables can be used by this item LIMIT_PLAYAUDIO_CALLER yes/no (which is used as the default, sound will be played for callers. LIMIT_PLAYAUDIO_CALEE yes/no sound will be played for receiver. LIMIT_TIMEOUT_FILE it plays after finishing the time. LIMIT_CONNECT_FILE it plays at the beginning of the call. LIMIT_WARNNING_FILE it plays as a warning when y was defined and remain time will be announced as the default.
M([class])	Play hold music for caller until one of the channels get ready to answer. MusicOnHold can be defined.
M((x)[^arg])	Perform the related Macro with the channel of receiver before connecting with that channel. Args of Macro can be sent for putting space between them by "^".Macro can determine the amount of MACRO-RESULT variable and according to the amount placed in it, the following operations performed after finishing Macro. ABORT: cut the connection between 2 callers

	CONGESTION: it seems the line has congestion. BUSY: when the line is busy. It causes that the program jump by the priority of n+101(if the j was determined). CONTINUE: cut a call of receiver and allows the caller to continue Dialplan performance and go to the next priority. GOTO:<context>^<extent>^<priority>: transfer the call with identified priority and can clear the extension and context.
n	This option changes the Screen/Privacy and clear that there shouldn't be any introduction in directory of priv-callerintros.
N	This option changes the Screen/Privacy and identified that caller ID is available and there is no need to screen the call.
O	It changes the caller ID of channel of caller to the caller ID of channel of receiver. Asterisk version 1.0 and before that were capable of doing this.
p	It activates the screen mode option which doesn't have protection like Privacy mode

P([x])	It activates the privacy mode and if x was defined, it was used for Family/Privacy key. If it wasn't defined, present extension will be used.
r	It plays the beep sound for the caller and there is no sound before answering.
S(x)	Cut a call x second after answering the receiver.
t	It allows the destination to transfer the call by the process of DTMF defined in features.conf
T	It allows the caller to transfer the call by process of DTMF defined in feature.conf
w	It allows the destination to record the conversation by DTMF process which is defined for one-touch recording in features.conf file.
W	It allows the caller to record the conversation by DTMF process which is defined for one-touch recording in features.conf file.

5.2 Extensions

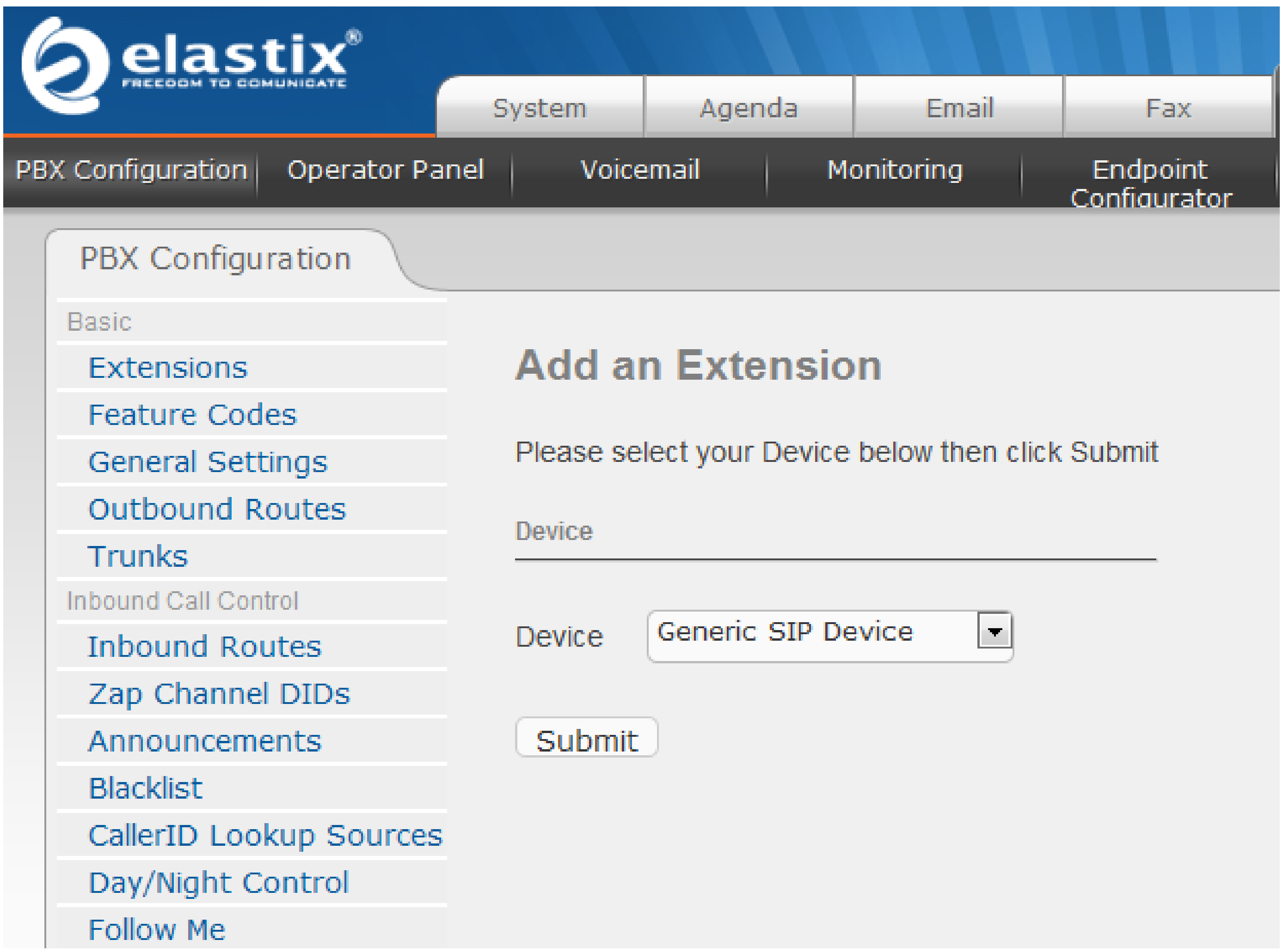
You can dedicate each number you wanted to your extension. Just don't use the following:

79-70	They have been booked for holding the calls
799-700	They have been booked for

	holding the calls
7777	It has been booked for simulation of the incoming calls
911	Emergency call (in Iran is 110)
999	Emergency calls of other countries

5.2.1 *Create extension*

For making extension the following menu is used:



You can choose a Protocol and create the extensions:

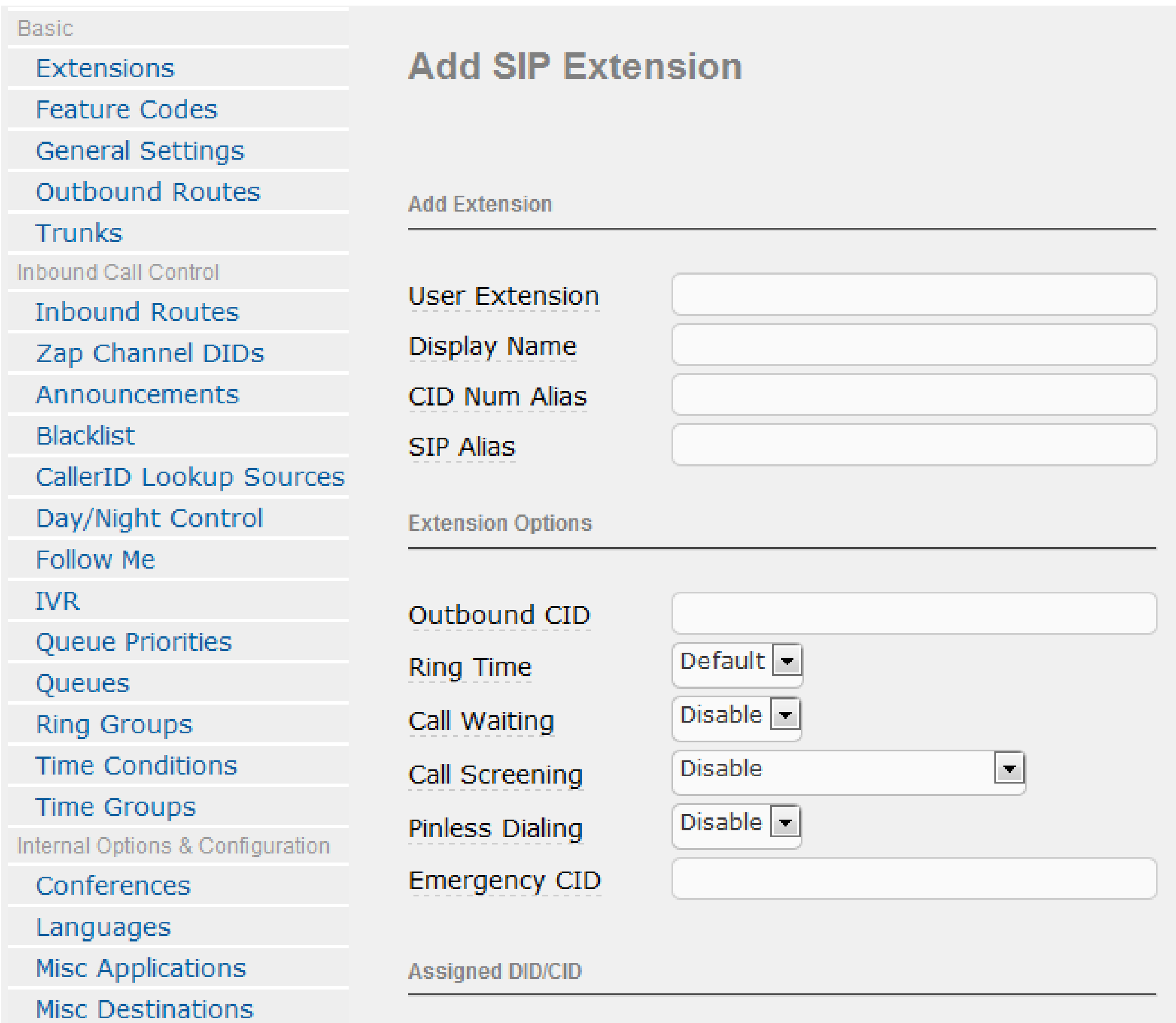
Generic SIP Device: SIP is a standard protocol for ATA & VoIP phones.

Generic LAX2 Device: LAX is used for connection between two servers of Asterisk or you may want to have IAX extension.

Generic ZAP Device: ZAP is a hardware which is connected to your Asterisk server and they are used for defining the FXS modules.

Other (Custom) Device: this term is for non-standard devices like H.323. It also can be used for mapping of an extension to another number.

For creating an SIP extension, Genetic SIP Device should be chosen and then we click on submit. Enter your information and click on submit:



User extension: dedicated extension of the user

Display name: (optional) name of user

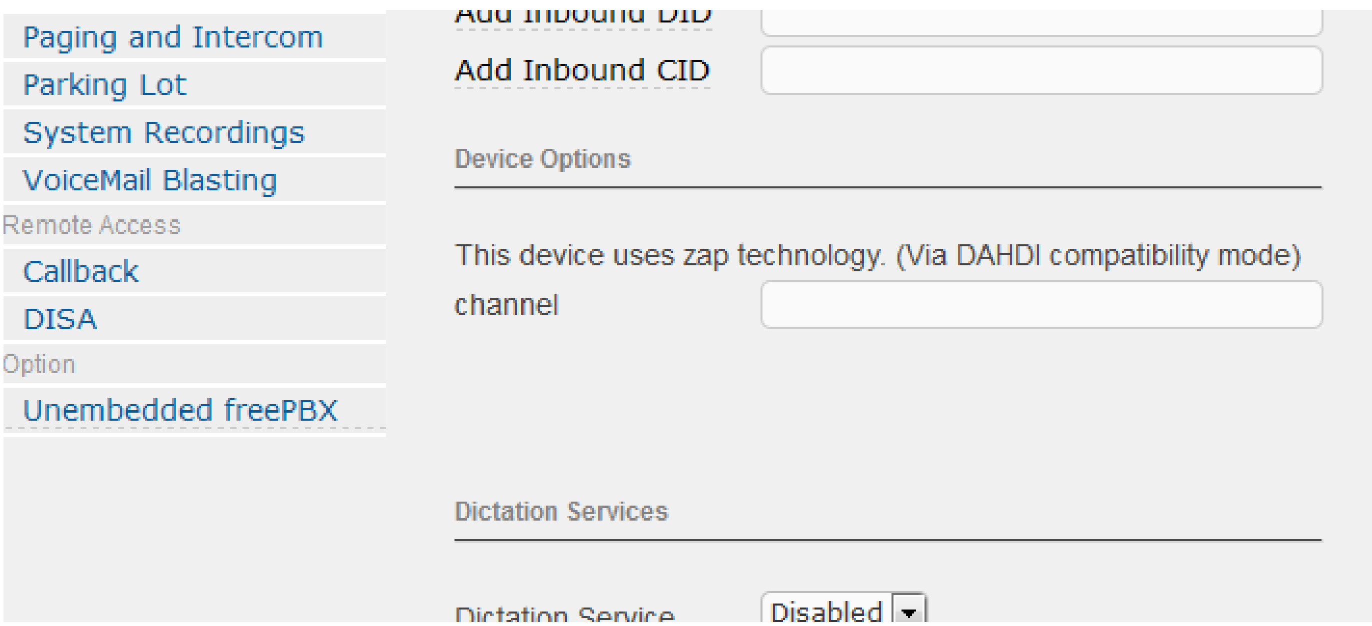
Secret: password with extension number used for authentication of the user.

Don’t forget to click on Apply Configuration Changes.

5.2.2 ***ZAP extension***

Creating ZAP extension (for module of FXS) doesn’t have any difference with creating SIP or LAX extensions. You should just use ZAP Genetic Device instead of SIP Genetic Devise (bear in mind that newer versions of Asterisk and Elastix for recognizing hardware use DAHDI instead of ZAP but the word ZAP is used for recognizing the hardware).

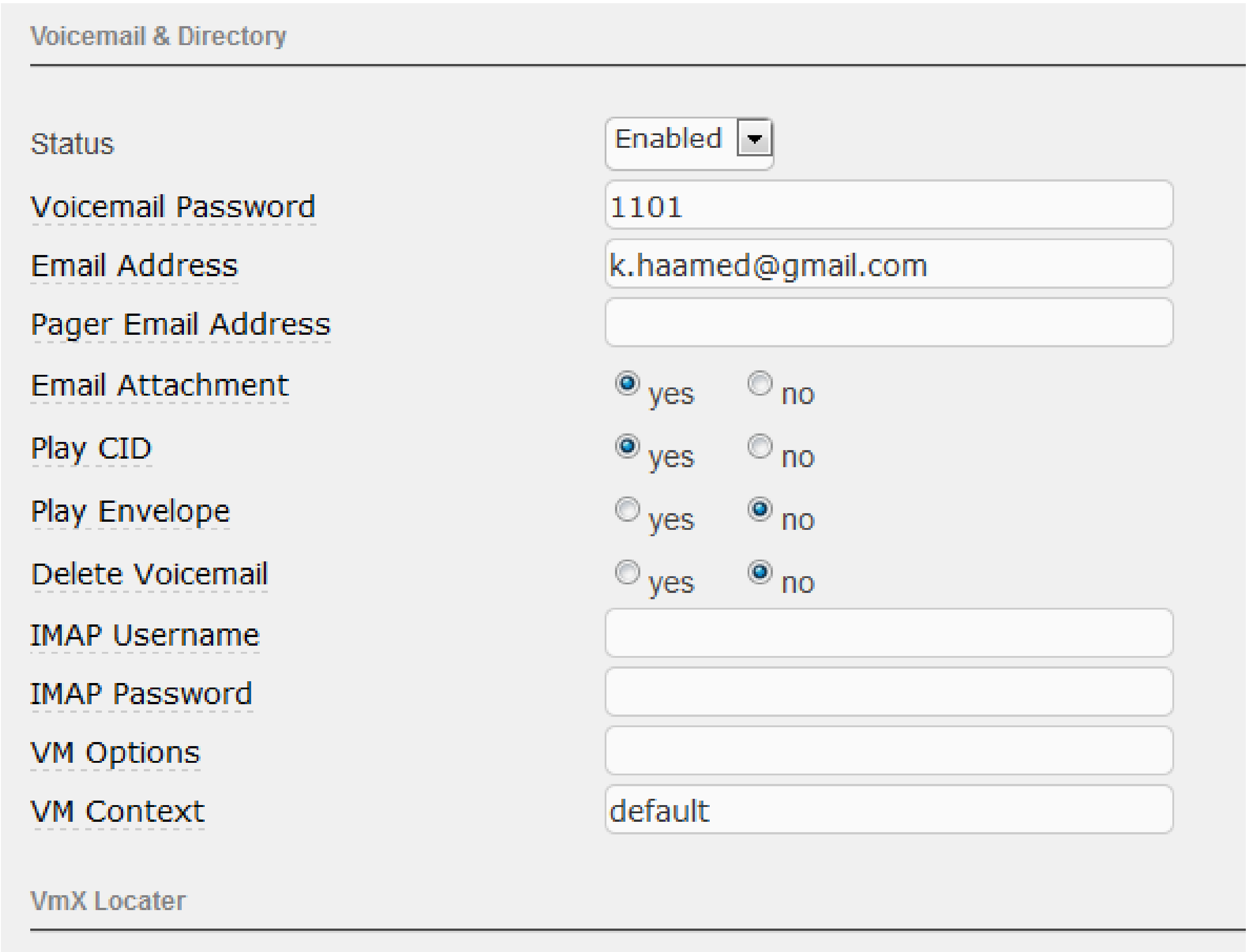
Attention: it is important that enter the channel number of your module properly otherwise the sound will be sent one sided or fax devices attached to the FXS module will have trouble in sending fax.



Attention: consider that if you send sound on SIP, you should use ulaw and alaw codec because the fax which is sent by other codec will be failed.

5.2.3 *Enable voicemail*

During registering extension you can enable your voicemail otherwise whenever you need to have an extension and voicemail you can enable it. For that you can go to the phone setting of extension and enable voicemail on any extension and click on it (create a new extension).



Voicemail password: the password of your voicemail

Email Address: email of a person who has that extension (it is recommended if you want to be informed through emails)

Email Attachment: yes (attaché your voice mail in email)

Play CID: yes

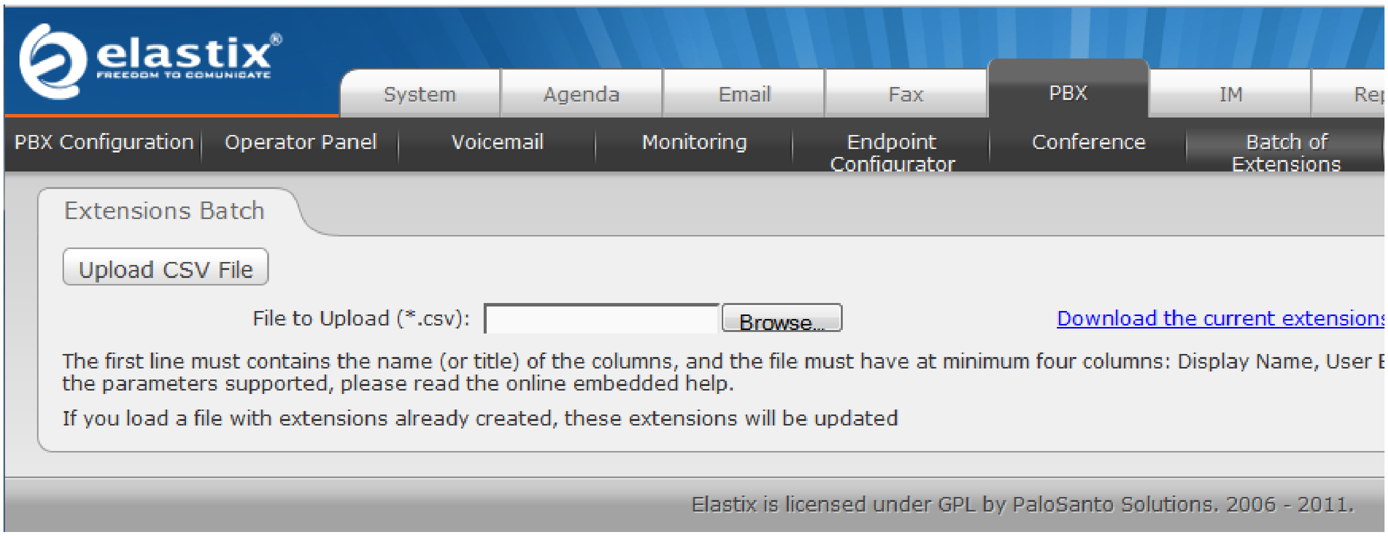
Enable Envelop: yes

VM Context: default

After changing these, click on submit and Apply Configuration Changes.

5.2.4 *Definition of Extensions Batch*

In this section you can enter extensions batch to the system by excel & …. In order to work easily with this part, it is recommended to create one or two extension as it was said.



Then click on file of extensions batch and choose to download extensions in CSV format. You can download the extensions file on your local hard. After opening the file with your favorite program, you can see the name of columns and extensions you have created. By that you have a guide to fill your file cells. Most fields are repeated in all users and you can copy them and fill the sections you need.

After entering all the extensions, save the CVS file with your desired name. Now you need to upload your file in server. You should enter to the extensions batch files and click on brows. Choose the file, where it is saved, and click on CVS upload file and that's it! For checking extensions go to the phone setting and see your extensions.

5.3 **setup softphones:**

- Setting the CounterPath Xlite:

First download this softphone from CounterPath or VoIP-IRAN:

http://www.counterpath.com/xlitedownload.html

eybeam software is one of the best softphones of the world which is used for SIP calls. This is made by Counterpath Company and its free version is X-Lite.

This software gives the following option to the users:

- The application provides a graphical interface which is made by phone standards.
- Feasibility of using 10 account of SIP simultaneously.
- Having video calls.
- Providing call transfer
- Recording the conversations

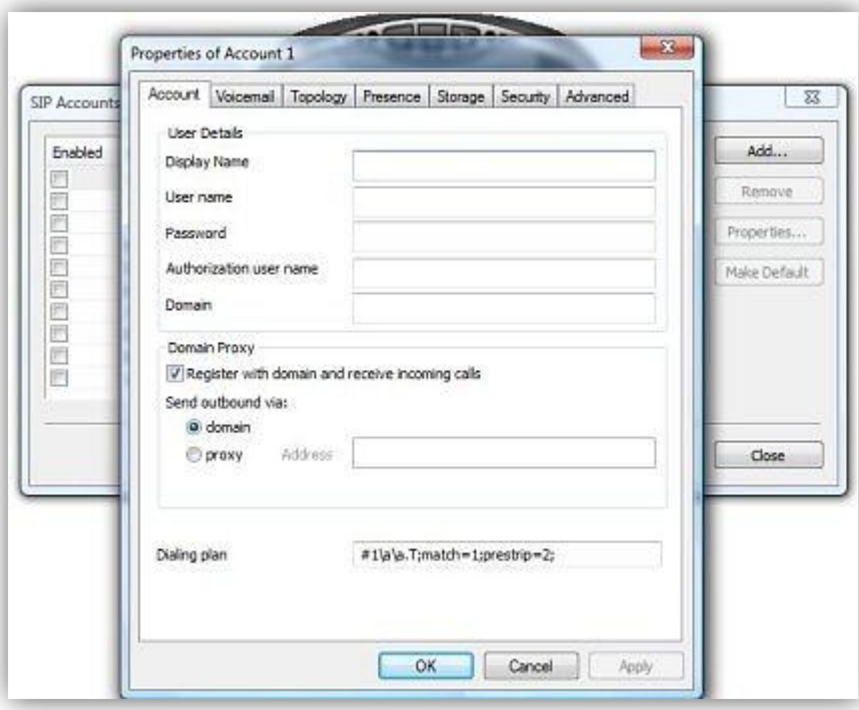
  And so on.

After installing the program and entering the serial number, you should create an SIP account. (Bear in mind that this account should be made in Asterisk, Elastix or any other server which you used). For that you should click on show menu above the eyebeam.

After clicking this window will be open:

Choose the SIP Account Setting. The window will open as follow:

Click on add key and the windows of properties of account will be open:



As you can see there is different part in this window that should be filled with information given to you by network manager.

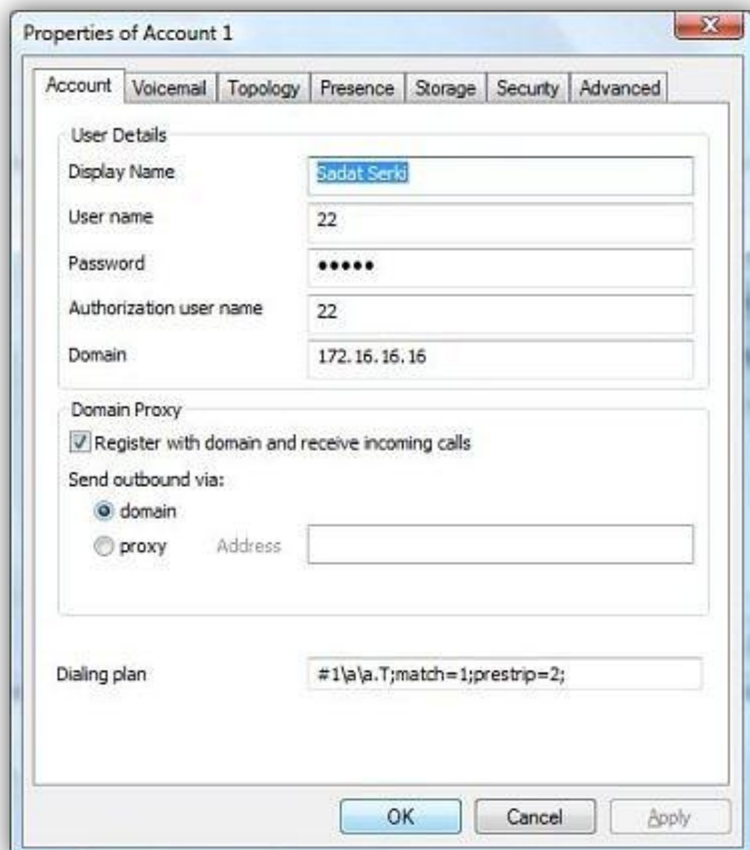
Display name: when you call someone, receiver sees your name and number. Enter the name you want to see.

Username: enter your username. Generally the user name is the same with your extension number.

Password: enter your password.

Authorization user name: again enter your username (this part is used for connecting to Asterisk by Sip Proxy and if you are connected directly, it is not necessary to enter that).

Domain: enter your name or ip server. (Just enter your name when you have DNS Server in network)



For other parts don't suffice to default setting and click on ok. The following window will be open and your account will be shown there.

Click on close. Eyebeam tries to connect with server (be careful in some systems firewall ask you if you allow eyebeam to access or not. You should let eyebeam otherwise you will have problem in receiving or creating calls). After connecting eyebeam with server and registering your account in server, eyebeam shows the ready message.

Some of the common errors of eyebeam:

Error 403: trouble in registering: it occurs when user make a mistake in creating SIP account and by checking username and password and… and reentering them it will be solved.

Error 408: request timeout: it occurs when eyebeam face one of these problems:

- Problem in network that cause not to find the server.
- Mistaken ip or name of the server which entered in SIP account section.
- Closing the port of SIP by firewall on the server or system of the user.

Not hearing the other voices: it can appear when:

- Defected headset or not being connected to device (it is obvious that if the headset wasn't connected, no voice can be heard. You should choose the easiest way to solve the problem if you have the network management)
- If the speaker's voice was closed in Eyebeam.
- Using the Codec that is not activated in Eyebeam. For managing the Codec in Eyebeam you should go to the Show menu--> Options --> advanced--> Audio codec and chose the activated codec in server.

Not being able to hear your voice:

- Defected headset or not being connected to device (it is obvious that if the headset wasn't connected, no voice can be heard. You should choose the easiest way to solve the problem if you have the network management)
- If the speaker's voice was closed in Eyebeam.
- Defected Microphone or not being connected to the device.
- If the key of microphone was inactive on headset. (The most common problem happened in our company). When the microphone being muted on Eyebeam.
- Lack of coordination of Codecs

Eyebeam has a secret part for hiding the advanced setting which is possibly good for professional users. Dial ***7469 for entering to this part.

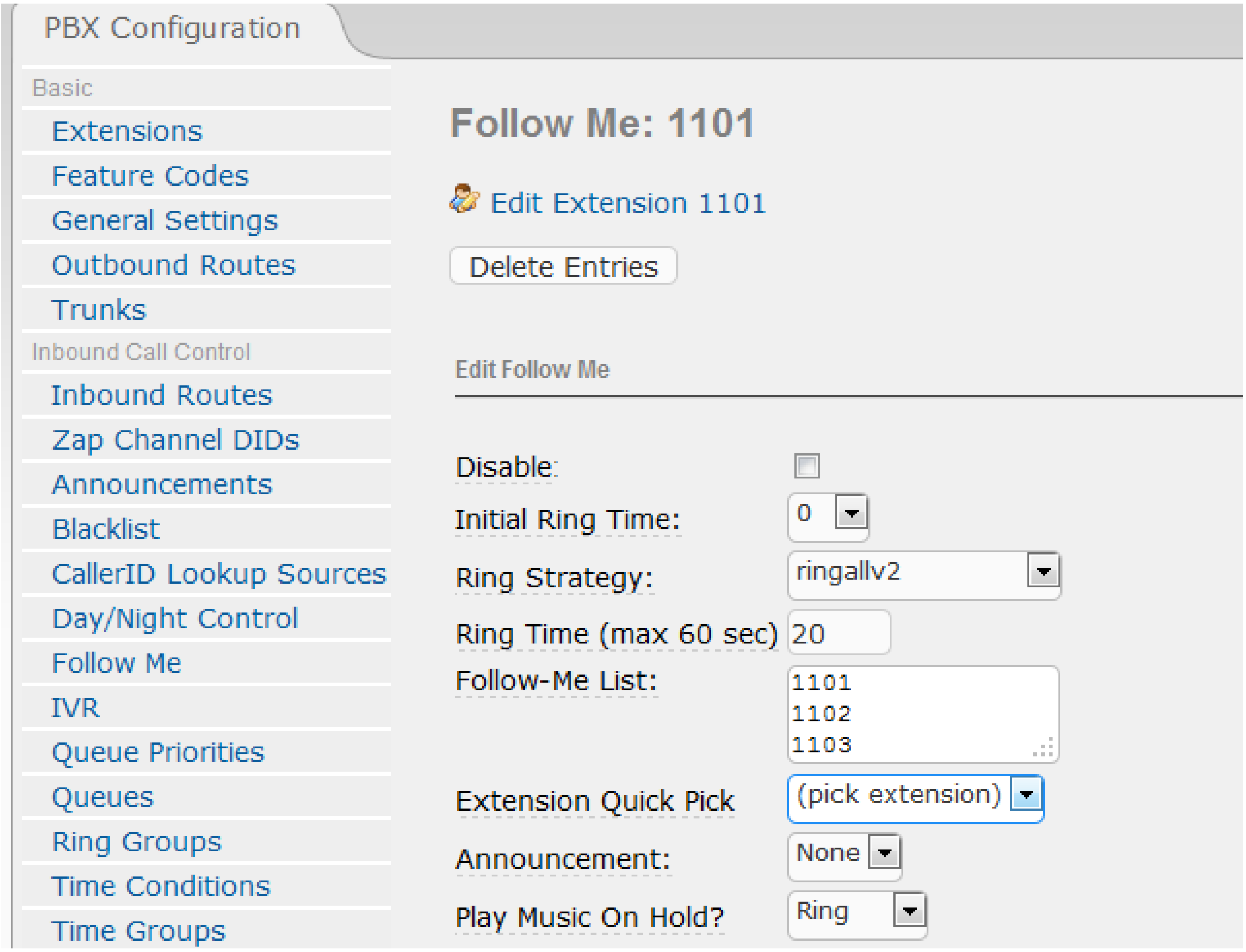
5.4 follow me

When we registered extensions, Elastix can be set to meet our needs. It is possible that we want the system automatically connected to the extension we already defined if our extension didn't reply. And we should do as following:

Call center, configuration of telephony system, follow me

We faced with this window:

Choose the extension you want to define this features.



Ring Strategy: ringallva2 dial the main number first and then the others.

Extension list: 1102 is the deputy director and 1103 is the office assistant.

Ring Time: 20 second

Destination If no answer: terminated call-hang up

Whenever dialed to the manager, dials the Asterisk extension number of the manager. If no one replied, the contact is with the extension of 11 and 22. And if no one answered again, Asterisk terminates the call. After finishing, choose the submit changes key and Apply Configuration changes here.

Attention: as we saw in follow me, several extensions can be determined as the next destination and it is the main difference between follow me and call forward.

5.5 What is dial pattern?

Dial pattern is a pattern for management and supervision of Asterisk for sending and receiving the dialed number or entered on telephony system. For instance a number is dialed on the system by a user; you can manage the number with a dial pattern, add or remove something to it. It is used both in Outbound & inbound route and in Trunk. Basically you will face with dial pattern in any part which deals with call management. There are some rules for definition of a pattern as follow:

X: represent a number between 0 to 9

Z: represent a number between 1 to 9

N: represent a number between 2 to 9

. : When there is a point in the pattern it means that there can be more number after that.

|: it means the number before that should be removed.

[ ]: if you want to choose your number, for example [1356] it means one number between 1, 3, 5, and 6. Or [3-7] means a number between 3 to 7

For example:

9|X.| it is a popular pattern. It means choose the numbers start with 9, take the 9 and send the rest to the destination.

ZX this pattern means the dialed numbers which are 2-digit, and their first number is between 1 to 9 and their second number between 0 to 9.

NX. It means 3-digit number and more that their first number is between 2 to 9 and rest can be anything.

5.6 What is trunk?

Trunk is used for connecting Elastix with outside and other systems. For making trunk required protocols for connecting with outside and other system should be determined and define trunk according to it. The current protocols are:

Zap: it is used for installing the Asterisk cards (Dahdi support) on telephony system. After putting the card in server, a Zap trunk should be defined for sending and receiving the calls. As a default a Zap trunk is defined in Freepbx known to ZAP/g0 which there is no need to install a card on system. But if you want to install more than one card, you should another ZAP trunk.

LAX: it is used for connecting to Gateways and VoIP providers with LAX protocol. One of the important uses of trunk is connecting 2 Asterisk servers together which are discussed later.

SIP: it is used for connecting to Gateways and VoIP providers with SIP protocol.

ENDUM: its full statement is E.164 Number to URL Mapping. And translate the numbers to the internet addresses.

DUNDI: its full statement is Distributed Universal Number Discovery. It is a mapping system on VoIP network which any group knows the channel of phone group around on the network (at least one number) and it is created by Mark Spenser creator of Asterisk.

Custom: for the other Trunks that Asterisk support or protocols such as h323 or SCCP that added you may define several trunks. For example you define a VoIP trunk for international calls for a service provider and a PSTN ZAP trunk for connecting to city line through card. For defining any trunk, especially VoIP trunks for connecting to a Getaways or VoIP provider, you should request the configuration and definition of the trunk from the provider or manufacturer. They could be different. Here we have a case for example. The case below is for connecting to the Pennytel provider.

Click on SIP creation of the trunk; consider an outbound caller ID, of course you should follow the general format; it should have the name and number as follow:

“Your name”<your number>

Maximum channels: use it for the emptiest capacity unless you want to limit. For instance for creating an urban ZAP trunk, you have a 4 port urban card but it create a trunk with 2 channel that only two call can be mapped to the trunk simultaneously.

Dial Pattern: it was discussed earlier that before manage the numbers before sending them to the trunk. For example adding or subtracting 9 to them. But it is popular in outbound trunk.

Peer Details: is some setting for sending call. The following setting is shown for sending a call to Pennytel provider.

```
Username: 8880XXXX

Type: peer

Secret: password

Insecure: very

Host: sip.pennytle.com

Dtmf mode: rfc2833

Disallow: all

Allow: allow&ulaw&gsm

Canredirect: no

Canreinvite: no

User Context: it is the user number for connecting. It usually don't use for connecting to Gateways.

User Details: required setting for receiving the calls

Canreinvite:no
```

```
Context:from-trunk

Fromuser: 888XXXXX

Qualify: no

Secret: password

Type: user

Username: 888XXXXX

Registration: some service providers want this field:

888XXXXX: password@sip.pennytle.com/888XXXXX
```

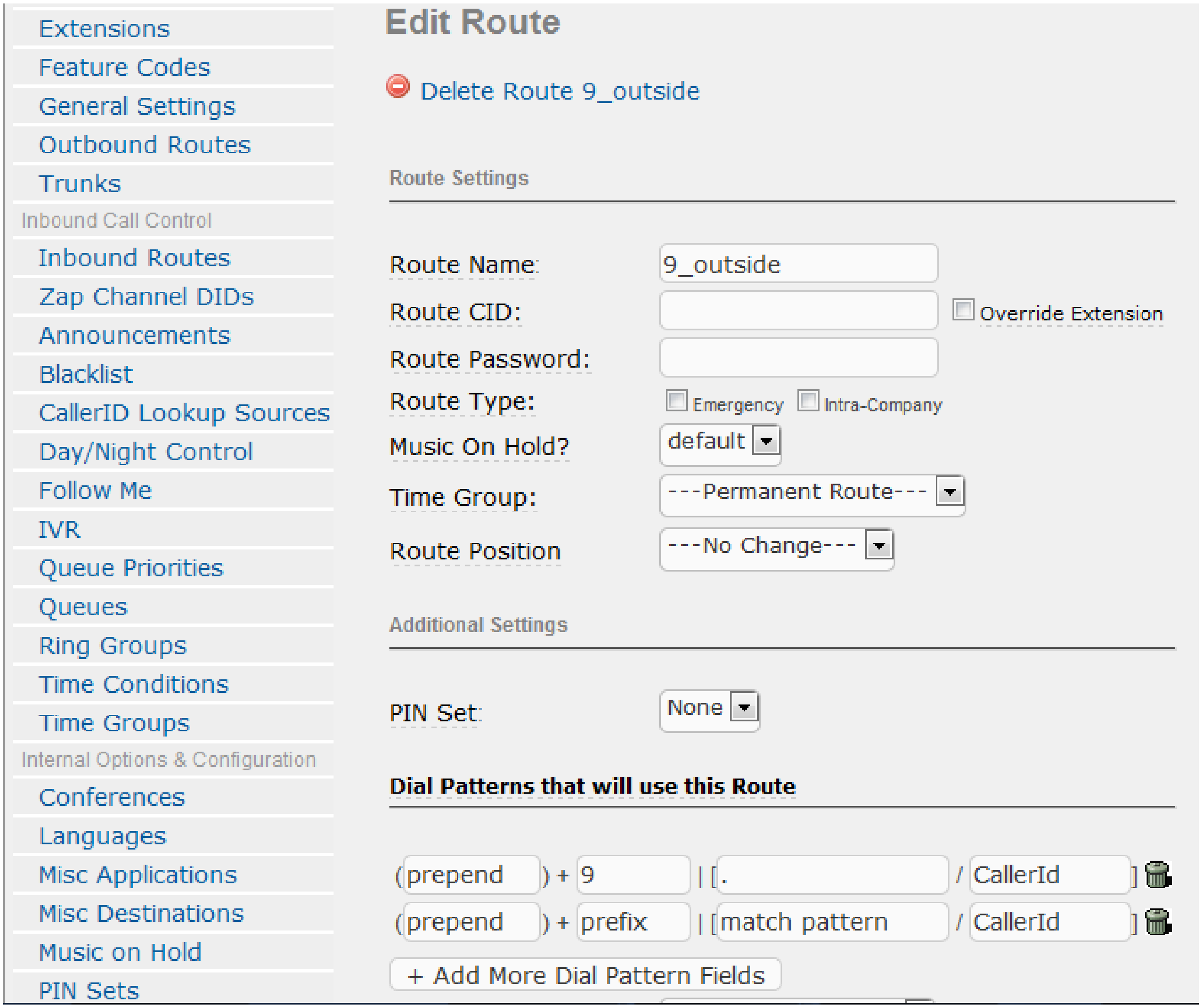
For more information, see the links below:

http://www.voip-info.org/wiki/view/Asterisk+config+sip.conf
http://www.freepbx.org/support/documentation/module-documentation/trunks

5.7 Outbound Routers:

Basically routes are used for mapping in Elastix. The purpose of Outbound routers is that a destination determined for each number gotten through ATA,IPPhone and Softphone. As a default of Asterisk if the number dialed, it goes to the extension. If there is no number with that extension, outbound route find the destination.

The most widely used of outbound routes is removing the number 9 and sending the rest to the urban trunk which is defined as a default due to the high usage. For defining the outbound route go to the call center menu, telephony system setting and click on outbound route.



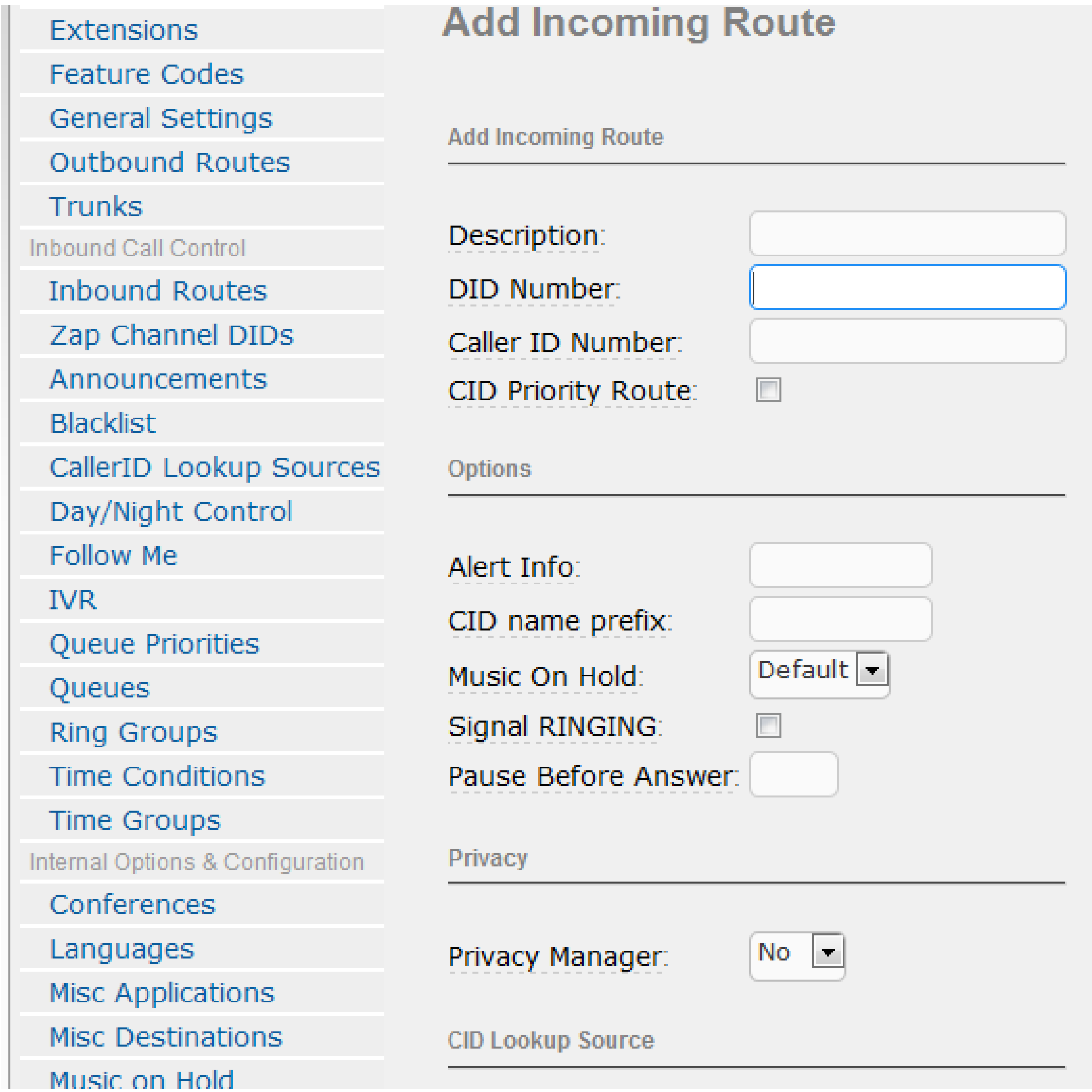
There are 2important setting and definition of route:

- Dial pattern which was mentioned fully in previous chapters. In this example you can see that this route is for calls that start with 9 and Asterisk took the 9 according to the dial pattern and sent to the determined trunk.
- Trunk sequence determine the trunk for this route and you can relate several trunk to it, so trunk are in the rank by priority and if the higher trunk is wrong and Asterisk cannot guide the calls, it moves to the next trunk.

  For example you defined an outbound route for 00 and international calls, in this case you may use different service providers of international phone. Trunks put them in priority in case to failure to communicate, use the alternative trunk.

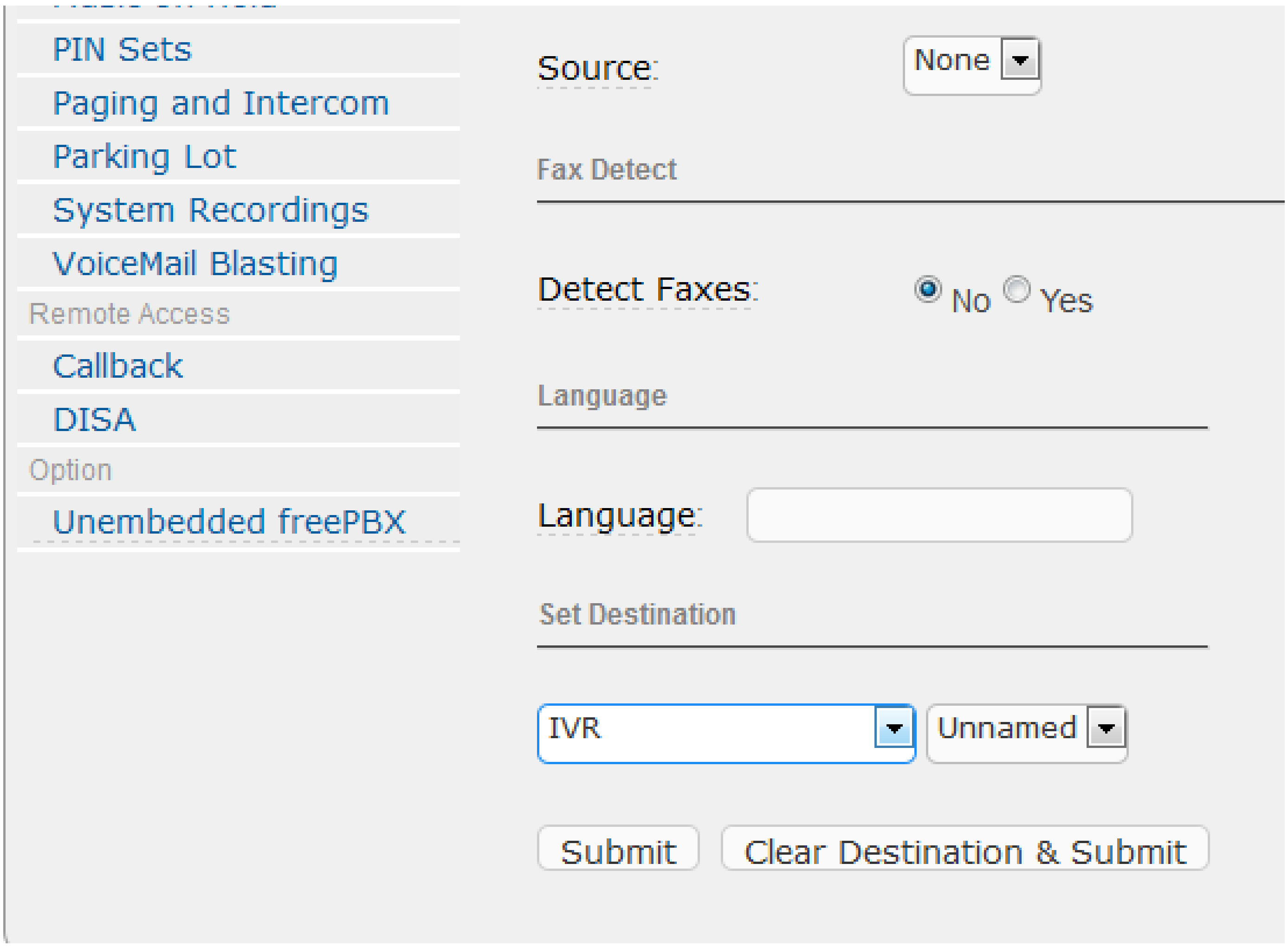
5.8 inbound routs:

The outbound calls to the Elastix should have an inbound route in the system otherwise Elastix wouldn't accept that. So choose an inbound route from call center and telephony system setting.



In the beginning of configuration, you can determine that this incoming router goes to lines (DID Number) or telephone number (Caller ID Number). For example if you use E1 for urban lines, you can determine an inbound router for main DID or define an incoming router for especial numbers (Caller ID) like the calls with pre number4465x connected to the announcement (especial message play for them).

Attention: you can use the dial pattern for the DID and Caller ID. When you write 4465X in all Caller ID, it include all caller IDs with pre number of 4465.



At last a destination should be determined for identified calls. You should set your destination in that field. Bear in mind that defined items are shown here. For example if you don't define any announcement, there is no option for that.

Attention: if you don’t need to separate the incoming calls or you just have one urban trunk and want to direct them to the specified destination, you don’t need to define DID or Caller ID and keep the fields empty. Just determine one destination and click on submit. In this case system creates an outbound route, Any DID/ Any CID, which include all the incoming calls to the Elastix.

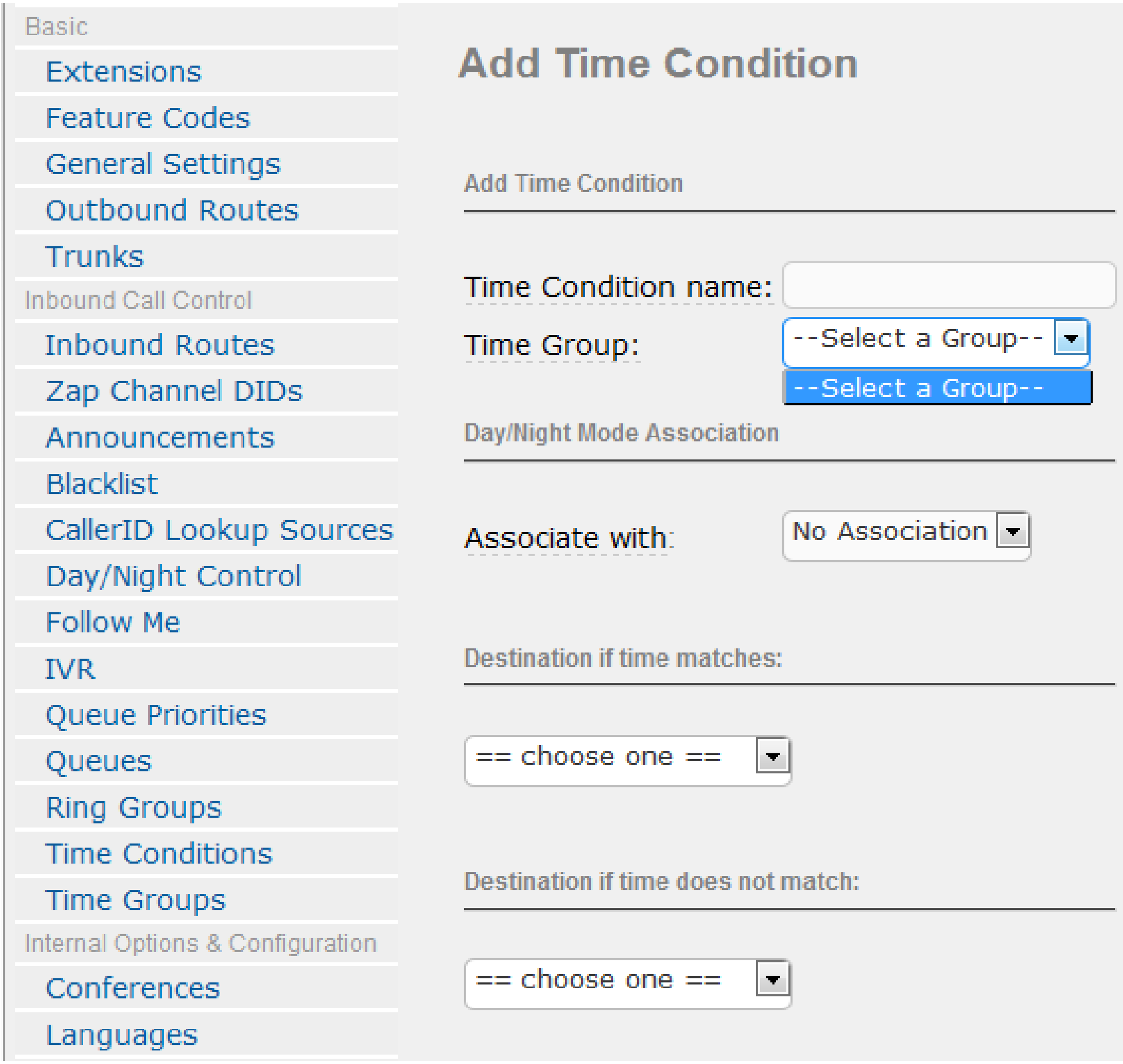
5.9 Time Conditions

This feature is used for this purpose that you could use several pattern in different time for inbound routes of system. For example you want IVR numbers 1 in working hours and IVR number 2 in closure of the company respond with 2diffrent algorithm.

To use this system in two parts the following settings must be applied:

Definition of response time:

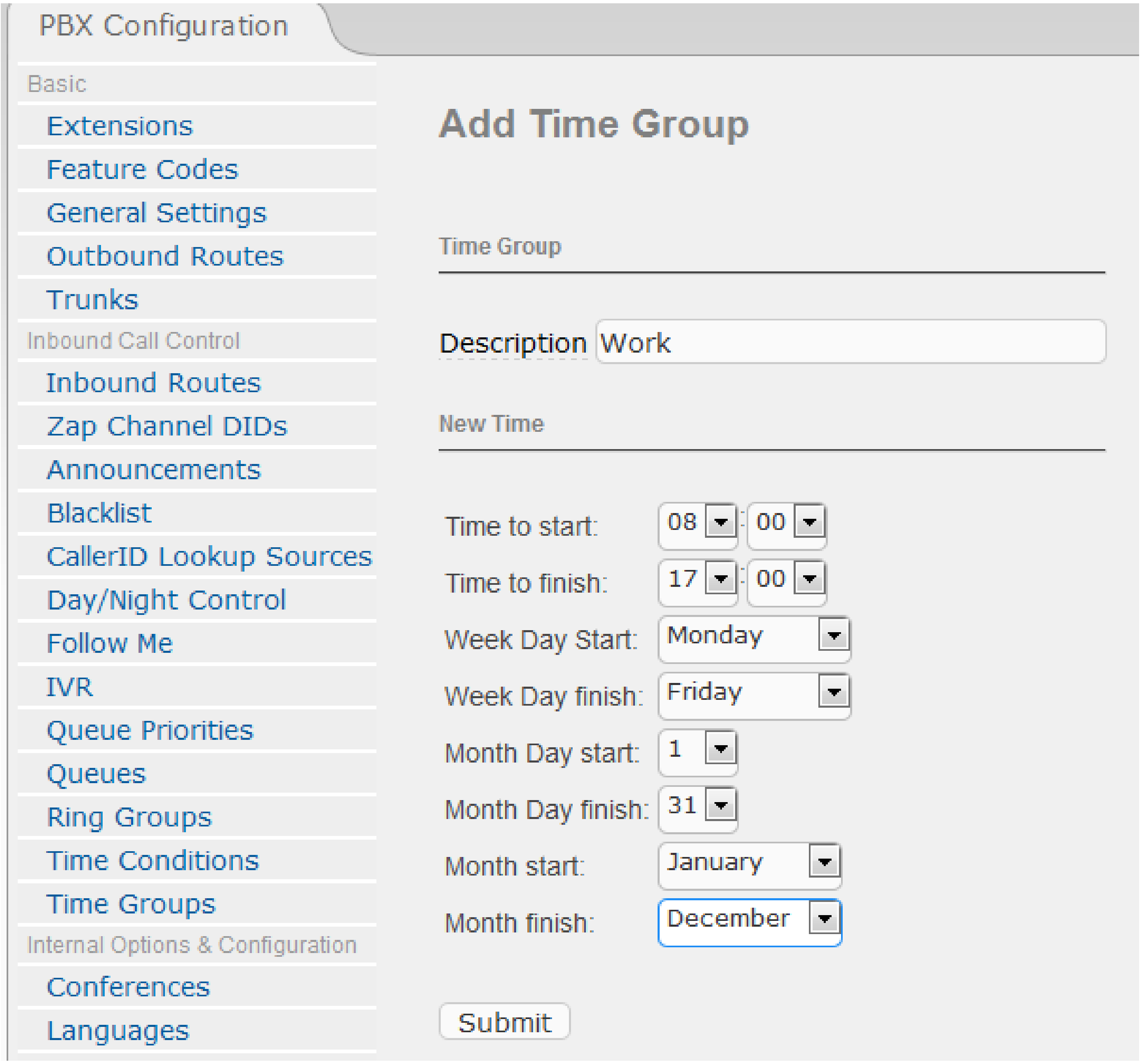
In order to define in “call center”, telephony system configuration of menu, choose the option of response time conditioning and click on Add Time Condition.



First, specify a name for it, in second field you should choose the determined time group. Time group is a specified interval such as month, weeks, days and hours and by choosing a group you determined that this situation belong to which interval.

Attention: for specifying the interval of time group, there is no menu in Elastix. It should be done through Freepbx. So from the menu go to call center, telephony system setting and choose Unembeded PBX. For entering to freepbx, use your admin user and password. (if you use Elastix version 1.6, the password is admin)

In freepbx, choose the time group.

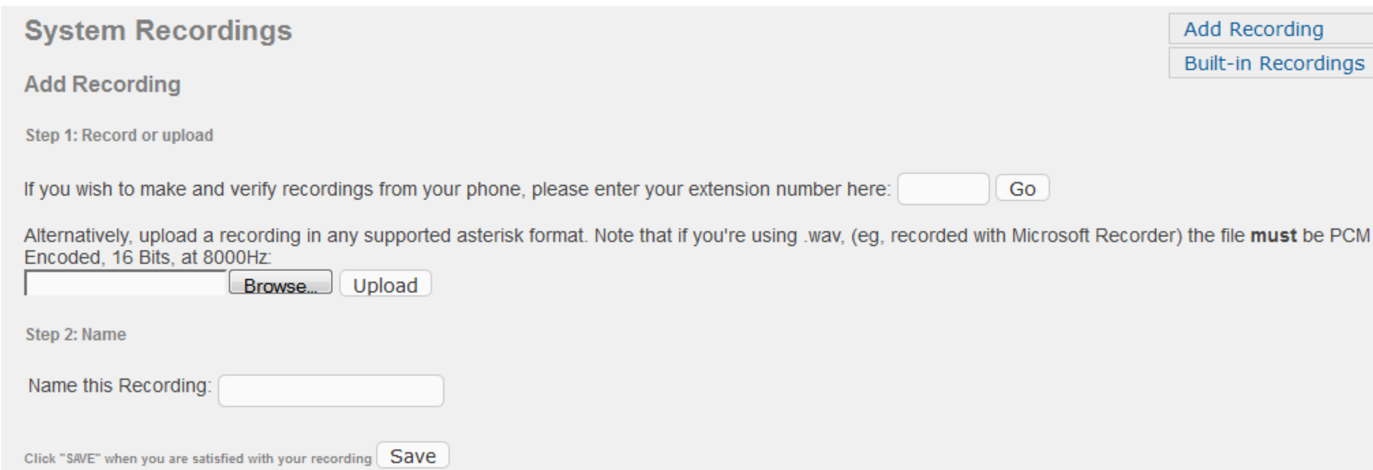


For example in above figure, we determine the group for working hours -8 am to 5pm- and the whole weeks of the month. Just consider that if you want to have plan for other hours, you should determine a time group for them either.

Now back to the time condition, you see a defined group in list of time group. Also In the lower part you specified the destination if time matches and if it doesn't.

5.10 system recording

In some part of Elastix you can add recording to the system and use them in menus and different system format. This part is in call center, telephony system setting in menu of system recording.



In this way you can transfer audio file to the telephony system. The first one is recording extensions and it lower the sound quality but it is essential. So enter your extension number to the specified place and after clicking on Go, your extension will ring. After recording you can name and save it. But if you want to put a prepared audio file on your system, choose your file on browse and upload it. After that you can name it on Elastix.

Attention: your audio file should have this format for uploading:

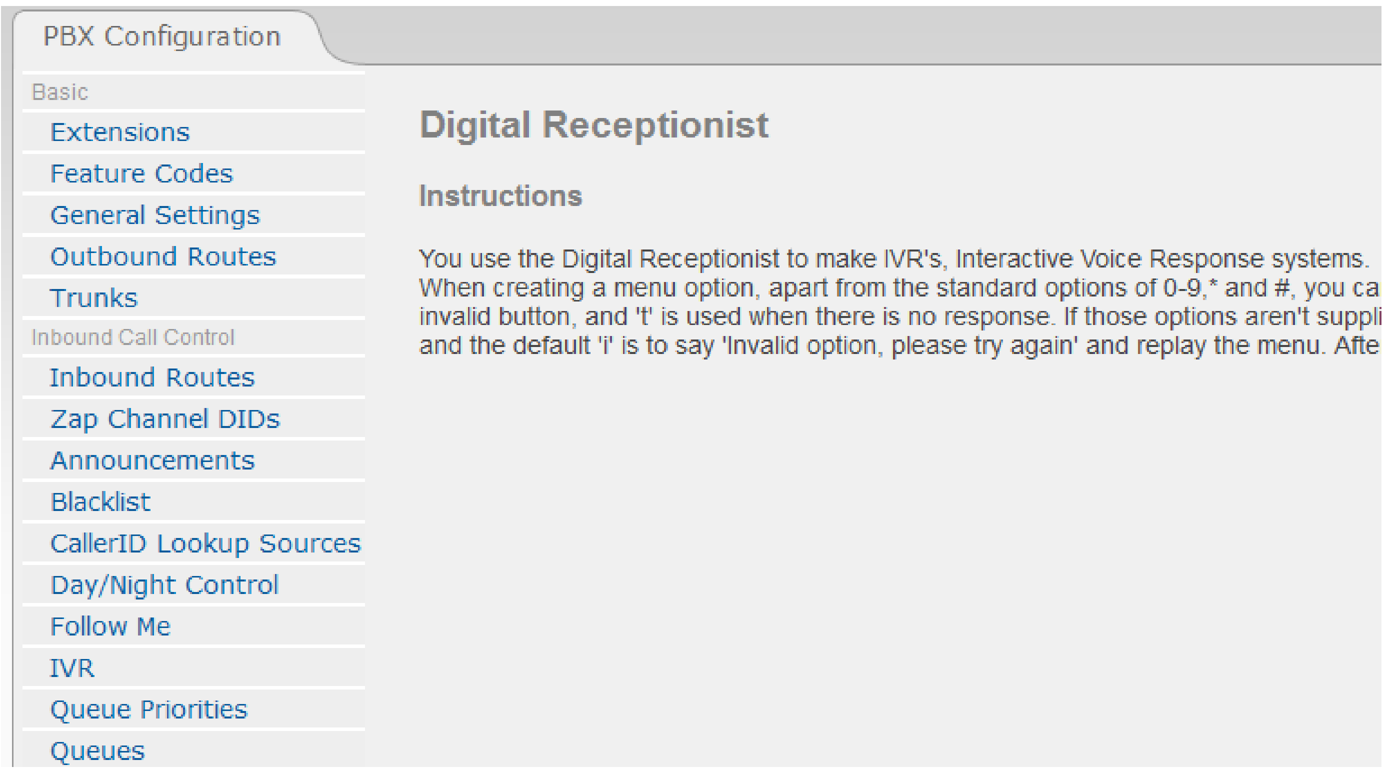
PCM Encoded, 16Bits, at Mono 8000Hz.

5.11 final destinations

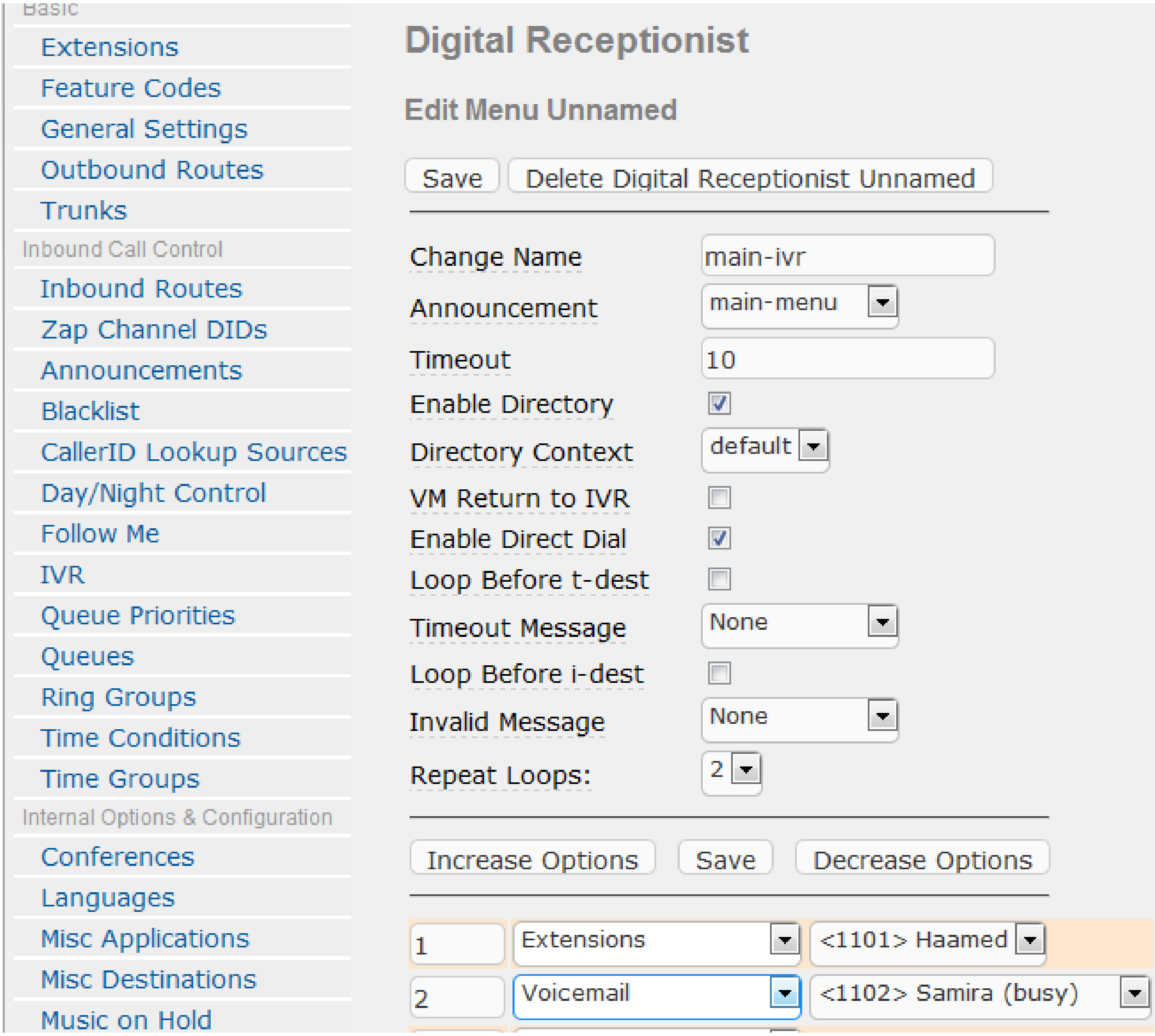
Any call from trunk that enters to the Asterisk should have a destination by Asterisk which is called final destinations. These destinations can be a simple extension, a complicated IVR or a written program by you. There are some common final destinations:

5.12 digital receptionists (IVR)

(IVR) interactive voice response is said to digital receptionist. An IVR plays the recorded text to the caller and ask them to press the key to connect to an organization, work group, a person or etc. then IVR send the call to the destination. It accepts any key to determine the call destination (ring group for the sale). In addition, 2 options of "I" and "t" can be defined. "T" is used for timeout and "I" for incorrect input. For having digital receptionist, you can click on IVR link on the left side of the page.



2part of setting of digital receptionist is shown. IVR options which is shown in the figure below with 12 adjusted options.



Change name: this option is for recognizing the menu and is not translated by freepbx.

Announcement: it is used for recorded messages for IVR. It should be recorded already and added to the message list of freepbx from system recordings.

Timeout: the entered number here is equal with the time that IVR waited for pressing the key by caller. If it doesn’t receive any, the call will be transferred to the “t” part and the call will be terminated.

Enable directory: If this option is selected, it allows the caller to press "#" key and entered to the directory system and search by name. By pressing the key pound, it plays a message to the caller that contains the entry stages.

VM Return to IVR: if it chooses, the transferred call from the IVR to the voicemail turns back to the IVR finally, after finishing with the voicemail and messages. If you don't select this option, the call will be terminated.

Enable Direct Dial: by this the caller can connect directly with dialing extension. Otherwise by entering the extension-even it exists- you hear that your number is wrong.

Loop Before t-dest: if you select this option, after considered period for timeout, menu will start playing again. It will be repeated with the specified number in Repeated Loops. If you don't select this option, the caller will be transferred to the determined destination in "t" immediately or the call terminated.

Repeat Loops: the number of iterations of IVR

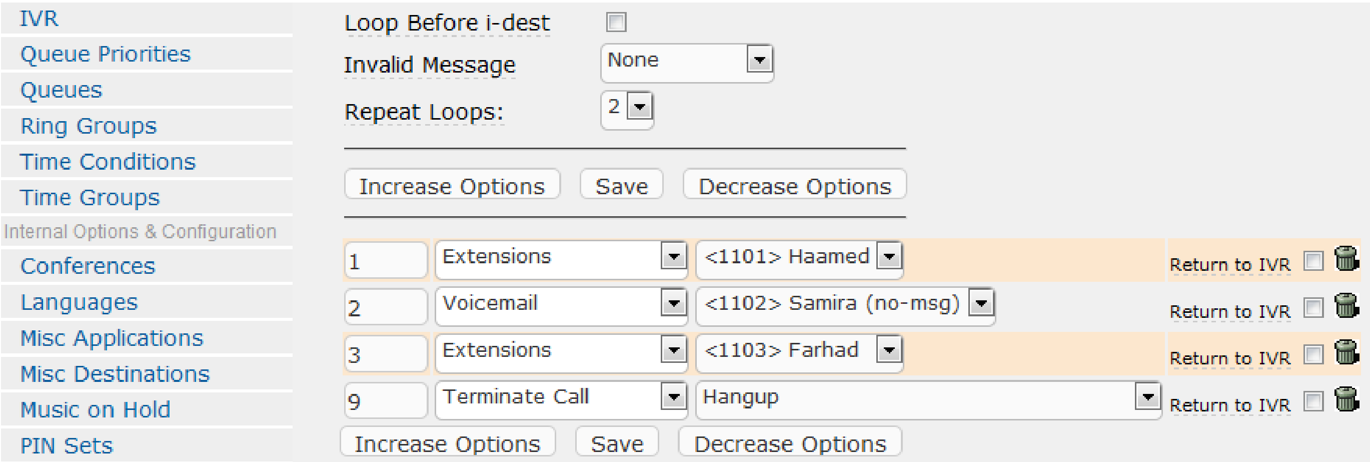
Timeout message: the message played for the caller when no key were pressed after the time entered to timeout. And it will be played when there is no destination for "t".

Loop Before i-dest: its performance is like loop before t-dest and check the key entry error. If you don't choose this option, the caller will be sent to the determined destination in "I" or the call terminated.

Invalid Message: it is the message played when the caller make a mistake in IVR menu. Consider that it will be played when there is no destination for "I".

Repeat Loops: it determined the number of repetition of voice menu if there was an incorrect incoming and after that the call will be terminated. The maximum repetitions number is 9.

The next part is IVR destinations that mapped the pressed key of caller to the intended destination. It is as follow:



In this example, key 1 sends the caller to the queue of technical support. "t" destination terminates the call an "I" destination is transferred to the voicemail of 501. The options can be increased by key of Increase Option. An IVR can have multi-digit options. Consider that IVR wait a while for one-digit incomings (for example, when a caller is allowed to enter the 1, 10, 11, 12 options, the user should wait for option 1 to be sent to its destination).

5.13 Ring Group

Ring group is a group of extensions that ring by external incoming calls simultaneously. You can add your mobile number if you want (you should have a trunk and a route, if you want your mobile phone ring).you can have a ring group for per incoming trunk or have one ring group for all trunks. For start we assume that we just have one ring group which is for all incoming trunks.

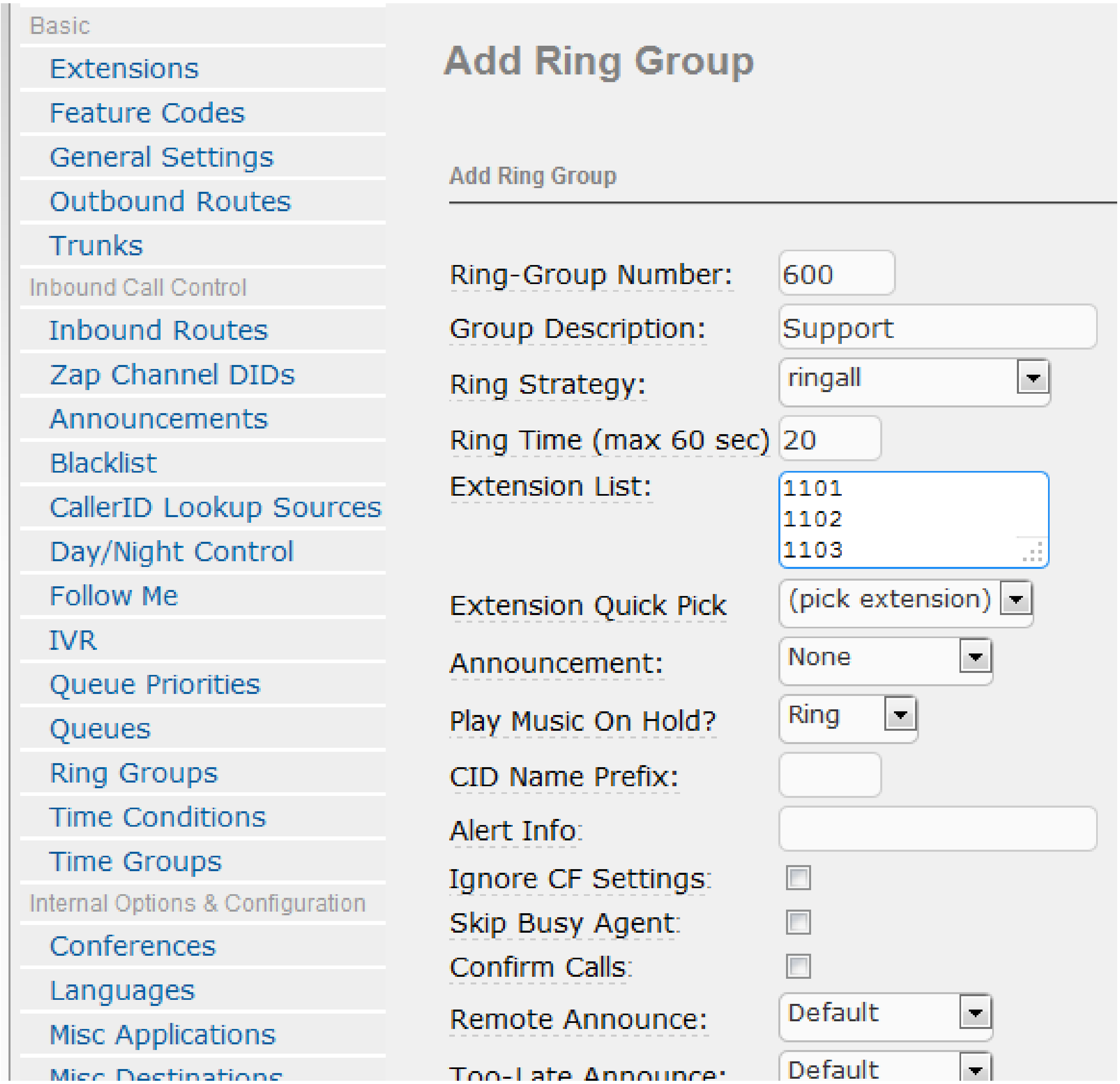
We consider the ring group 33 which is including project manager, network manager and deputy manager (one-digit numbers are not recommended for ring groups; try to dedicate at least two-digit numbers). If none of the managers reply, Asterisk will terminate the call.

Use this way to enter to the ring group:

Call center, telephony system setting, and ring group

After entering to this part, you will see this page:

Click on add ring group.



The options are:

Ring-Group Number: it is the number for accessing to the ring group. It is like extension number, both endpoints and users can dial it.

Group Description: it is used for recognizing the ring group and it doesn't have more efficiency than that. The users cannot be seen by callers.

Ring Strategy: it determines that how Endpoints should call. Endpoints will be ringing by one of the following ways:

Ringall: all the groups determined in extension list can ring at one time and together.

Ringall-prim: in this case if the first number in the list wasn't busy**,** all the numbers would start calling. If the first extension was busy or be in the do-not-disturb mode, none of the extensions ring and the call directly sent to the determined part in destination if no answer.

Hunt: if one of the group member replies, the others that specified in Extension list field starts ringing until one of the members answer.

Hunt-prim: it's the same with ringall-prim. If the first extension was busy or was in the do-not-disturb mode, none of the extensions would ring; otherwise it has the same process as hunt.

Memoryhunt: the first extension in the list rings and if there was no response, the second extension would start and similarly to the end (until one of the extensions reply or be timeout).

Memoryhunt-prim: if the first extension was busy or be in the do-not-disturb mode, none of the extensions ring otherwise it has the same process memoryhunt.

Firstavailable: the first available extension in the list will ring. If call waiting activated for the extensions, it would be consider available, however it is on another call.

Firstnotonphone: the first available extension in the list will ring. Call waiting setting is not consider here. If one extension was busy, without considering the activation of call waiting, press the submit changes key and Apply Configuration Changes Here

5.14 Queue

5.14.1 *What is queue*

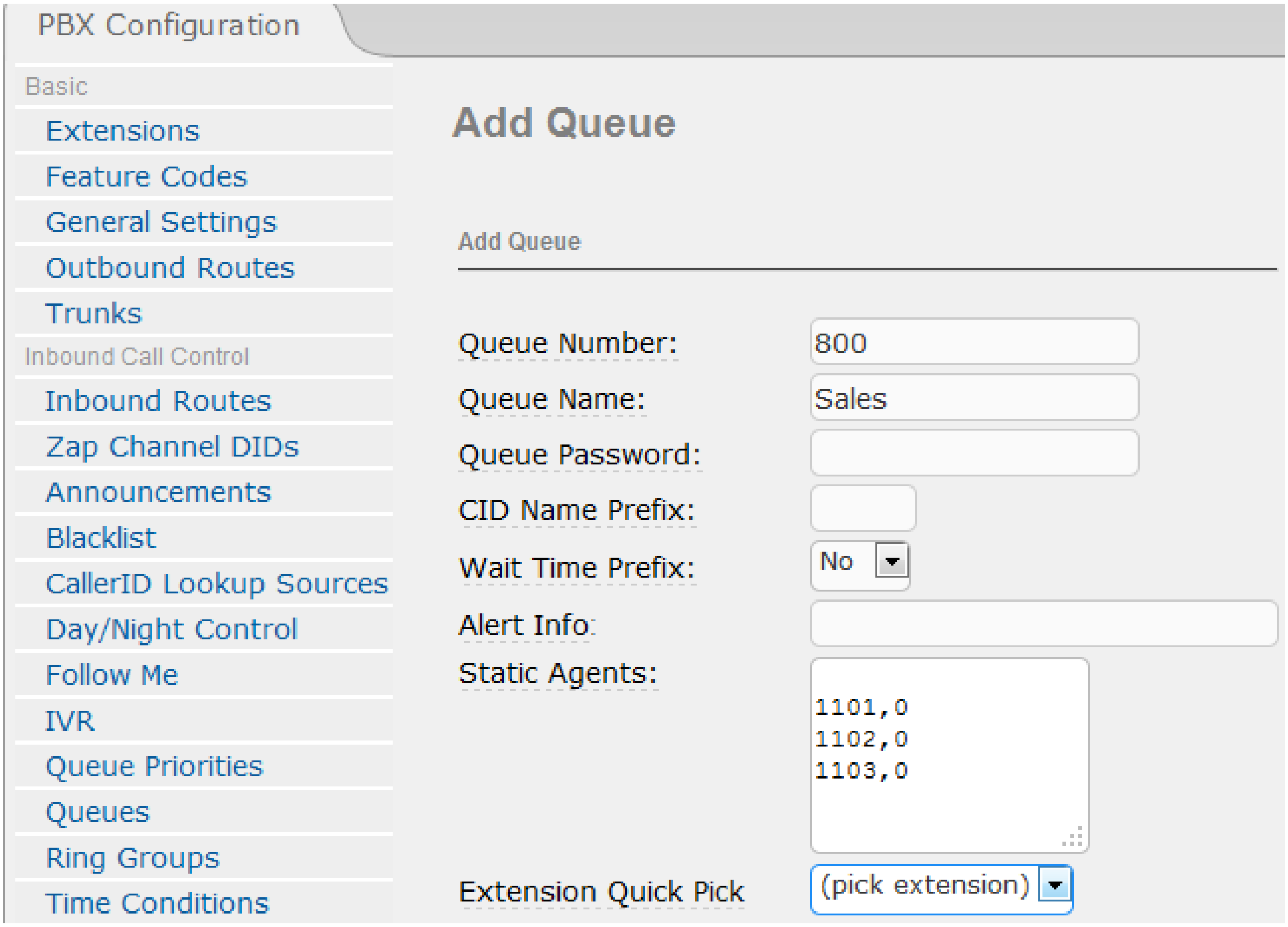
The meaning of queue is the same that we all know. Whenever your request is more than the resources or service providers, you need a queue. In the other word, in call center if you want to reply to the callers with limited number of receptionists, you need a queue. When the receptionists are busy, you keep the callers in queue until one can answer. This is the main task of queue.

5.14.2 *Kind of queue*

There are 2kinds of queue: static queue and dynamic queue. This division is not because of their function but it is according to those who are responsible to reply. In static queue, the extension in queue is responsible to answer and connect the incoming calls queue to the defined extension. In dynamic queue, the agent is responsible to answer instead of extension. You can have a static queue call center with Elastix, definition of queue and extension but for having a dynamic queue call center you need a call center module of Elastix in addition to definition of queue because it makes up the necessary equipment for defining agent and working situation. By the way for having a call center with any queue, you need its definition and familiarity with its capabilities. In next chapter, we will discuss more about the queue and its performance.

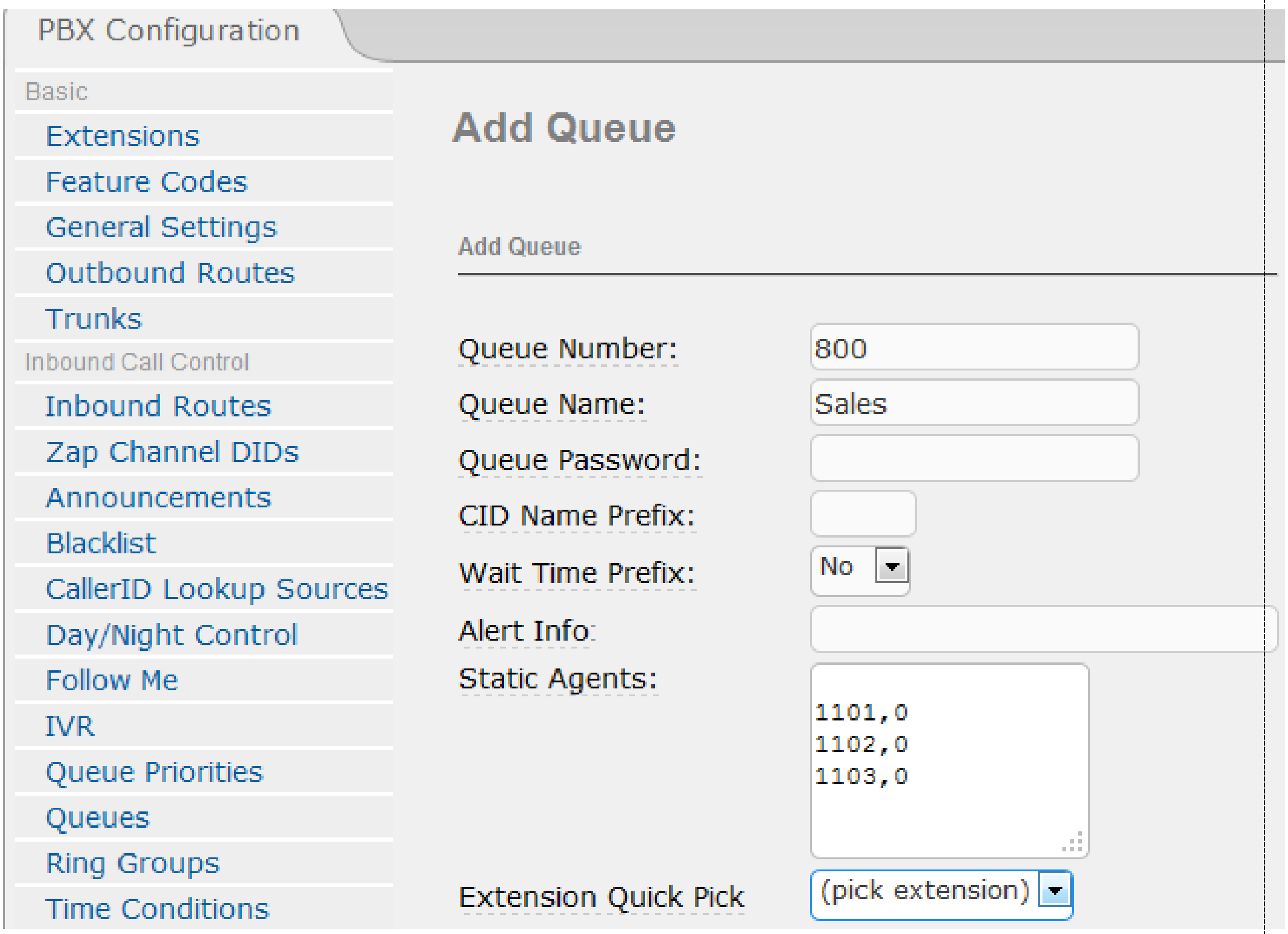
5.14.3 *Making queue*

Whether you want to have PBX or any kind of call center; you need to define a queue. It is one of the powerful features of the Asterisk which can consider a powerful tool for selling products. Facilities of queue are so complete that can set up an expensive call center and Enterprise and sell it.



In setting page of queue there are lots of features and parameters that some of them may be useless but in this chapter we try to explain then. At first we deliberate on Add Queue.

- Add Queue:

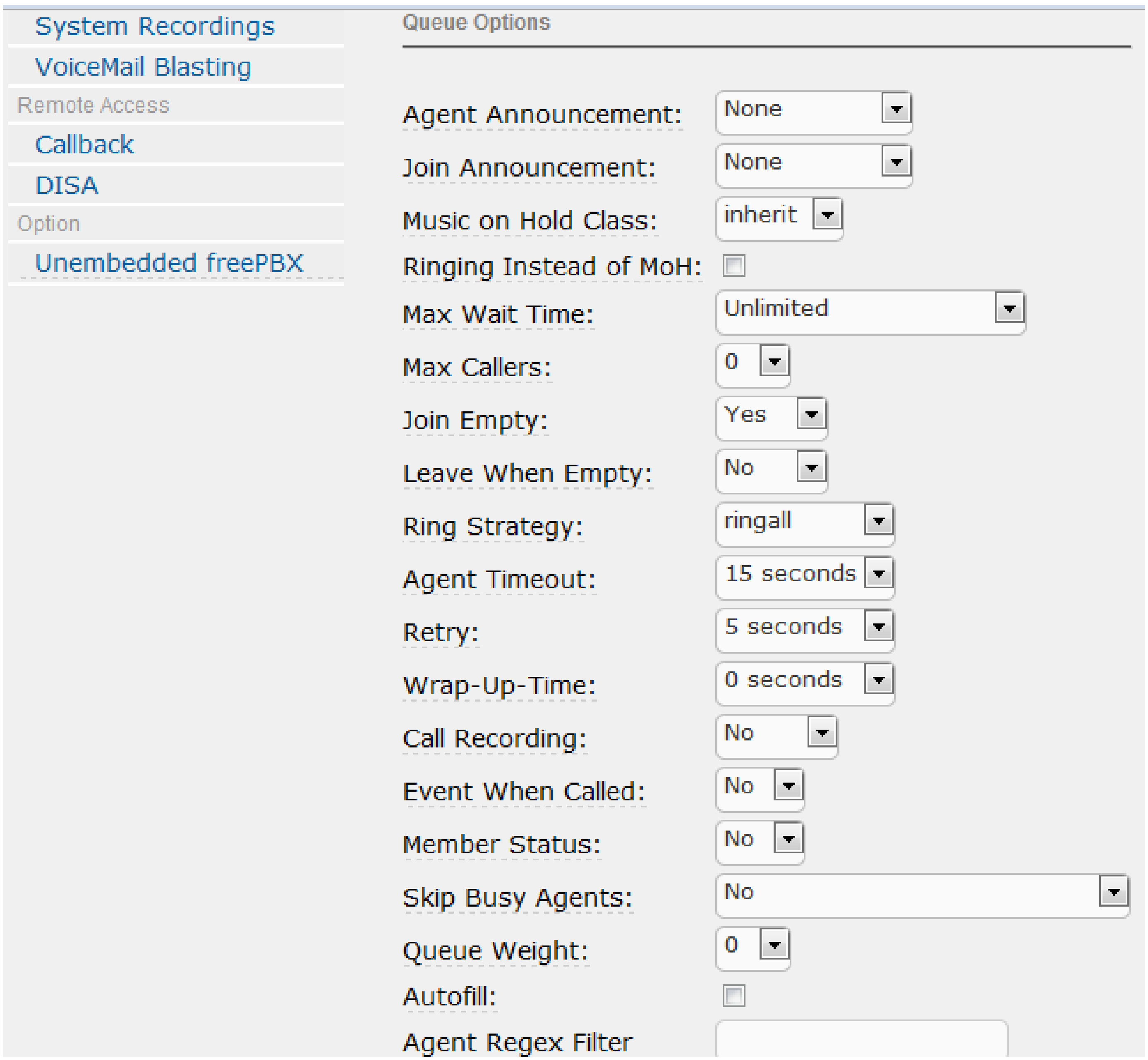


The name and the extension to reply are determined.

- Queue Name: a name for queue but try to choose a name associated with the purpose of the queue such as selling queue
- Queue Password: this is for agents who deal with dynamic queue in call center, so if you want if you want to work without call center module of Elastix and have a call center with this queue, don't fill it.
- CID Name Prefix: you can add a prefix to the number of caller for example queue name. In this case if your agent was the member of some queue, can be found from the number that this call is transferred from which queue.
- Static Agent: you should write the extension numbers that are responsible for answering this queue. You can also help from Extension Quick Pick for choosing extensions.

You should consider that if you want to use call center module, the number of defined agents should be written in that module which will be explained later.

Attention: when you use Extension Quick Pick for your extensions, it added a “0” number in front of the extension, writing that number is not required. That number is fine that its use is not mandatory. You can put a fine number. The more the number was bigger, the more calls will be connected to that extension.



- Queue Options

  The most important and most attractive part of a queue! Each of these parameters can take you close to a big contract so each function will be always remembered.

- **Agent Announcement:** playing a message for extension or agent before connecting call. It is possible that your agent be a member of some queue; by leaving a message for each queue such as "this call is from queue sale", then your agent know the incoming queue before getting the call.
- **Join Announcement:** playing a message to the caller upon entering the queue. It will be played once. It can be a welcoming message like" hi, welcome to the sell part of X company, please wait for connecting a call".
- **Music On Hold Class:** determining a music for the Callers who are waiting in queue. This is the chosen music for Music On Hold of a system as a default.
- **Ringing Instead of MoH:** the ring sound will be played instead of music for those who are waited in queue.
- **Max Waited Time:** the maximum time that a caller waited in queue. When it finished, the caller

  Will be out of queue and sent to the determined destination in Fail Over Destination.
- **Max Callers:** determining the capacity of queue for admission. For example you consider the capacity of your queue 10, the 11th caller is not allowed to enter and after hearing the message of full queue, come back to the queue of previous menu.
- **Join Empty:** default of this option is "yes" it means if the defined extensions and agents were not activated, should the caller enter to the queue? But if you don't want the caller enter to the queue when there is no activated receptionist in queue, change this option to "no".
- **Leave When Empty:** it is the same with join empty but there is a difference that in this case if someone waited in queue and suddenly the agent was not available, the waited one should be thrown out of the queue or not.

- **Ring Strategy:** This option includes algorithms for how to connect the caller to the extensions of the queue.

Ringall: all the groups determined in extension list can ring at one time and together.

Ringall-prim: in this case if the first number in the list wasn't busy**,** all the numbers would start calling. If the first extension was busy or be in the do-not-disturb mode, none of the extensions ring and the call directly sent to the determined part in destination if no answer.

Hunt: if one of the group member replies, the others that specified in Extension list field starts ringing until one of the members answer.

Hunt-prim: it's the same with ringall-prim. If the first extension was busy or was in the do-not-disturb mode, none of the extensions would ring; otherwise it has the same process as hunt.

Memoryhunt: the first extension in the list rings and if there was no response, the second extension would start and similarly to the end (until one of the extensions reply or be timeout).

Memoryhunt-prim: if the first extension was busy or be in the do-not-disturb mode, none of the extensions ring otherwise it has the same process memoryhunt.

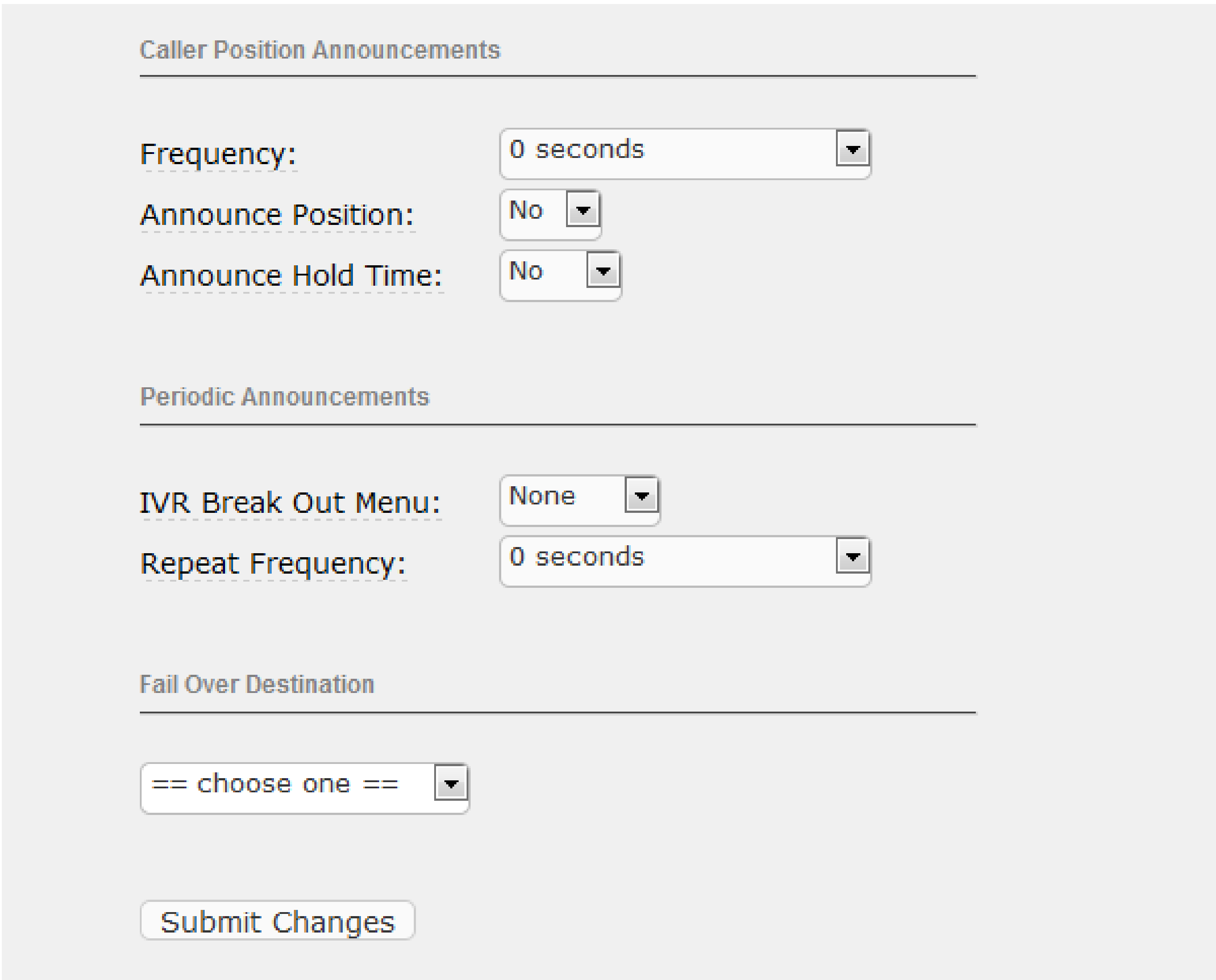
Firstavailable: the first available extension in the list will ring. If call waiting activated for the extensions, it would be consider available, however it is on another call.

Firstnotonphone: the first available extension in the list will ring. Call waiting setting is not consider here. If one extension was busy, without considering the activation of call waiting:

- Agent Timeout: the frequency (second) that the extension of the agent rings until the system timeout.
- Call Recording: recording the conversations of the queue. Audio files will be saved in /var/spool/Asterisk/monitor. Never forget that

recording the put extra pressure on server, so select a server capacity in terms of both processing power and disk space.

- **Queue Weight:** you consider a weight for a queue. You extension could be in extension of some queues simultaneously. So the weight of queues determines the priority for sending a call to an extension.



- Caller Position Announcements

- **Frequency:** determining the period for playing a message in second (choosing 0 for inactivation)
- **Announce Position:** declaring the position of a person in queue for example "you are the 5th person in the queue"

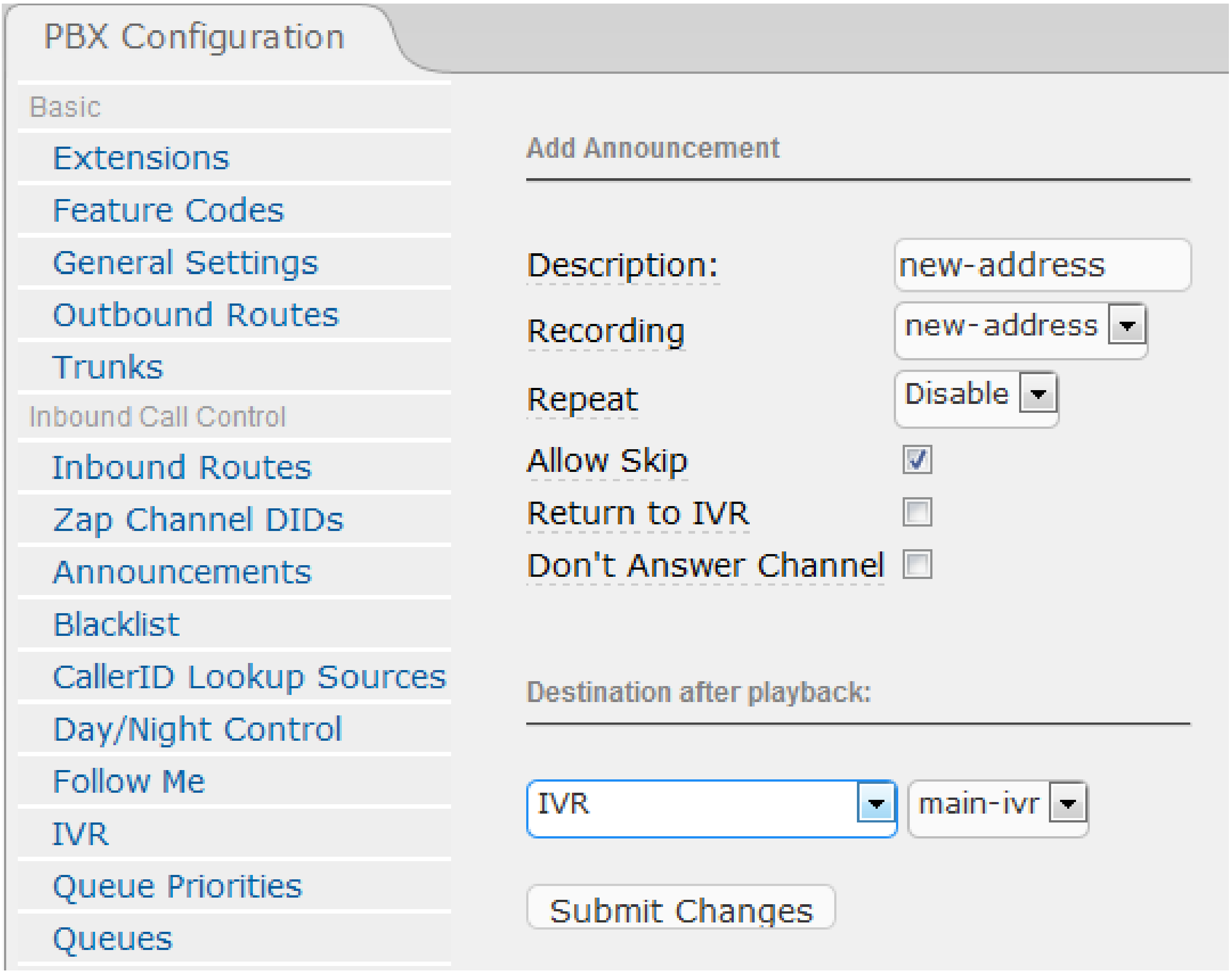
- **Announce Hold Time:** declaring the estimated time for waiting. If it was less than 1minute, it wouldn't announce. If you choose once, the message will be played just once.
  - Periodic Announcements: you can choose an established IVR in menu of IVR to be played for people in queue alternately. It can be used in this way: for example you want that an IVR played for the waited people who say "for returning to the main menu press key 1, otherwise wait". You can play any IVR for people in queue.

- **Fail Over Destination:** if for any reason the queue functions wrong, you can send the people in queue or those wanted to be, to the final destination and don't miss any call. Bear in mind that in final destination, you always will see the items in a list you've made, for example if there is no announcement on your system yet, you won't have it on destination.

5.15 Announcement

Sometimes it is necessary for you to play a message for the caller in telephony system, for example you want that if someone presses key 8 on IVR, the address plays for them or agents play an address or a message with transferring a call to a number so they use announcement.

In menu in "call center", "telephony system setting" choose "announcement".



At first choose a name for it. The other configuration is:

Recording: from list choose a message which was loaded through menu of recoding.

Repeat: choose a key to repeat a message when a caller entered.

Allow Skip: if it was activated by pressing any key the message can be cut and directly send to a next destination.

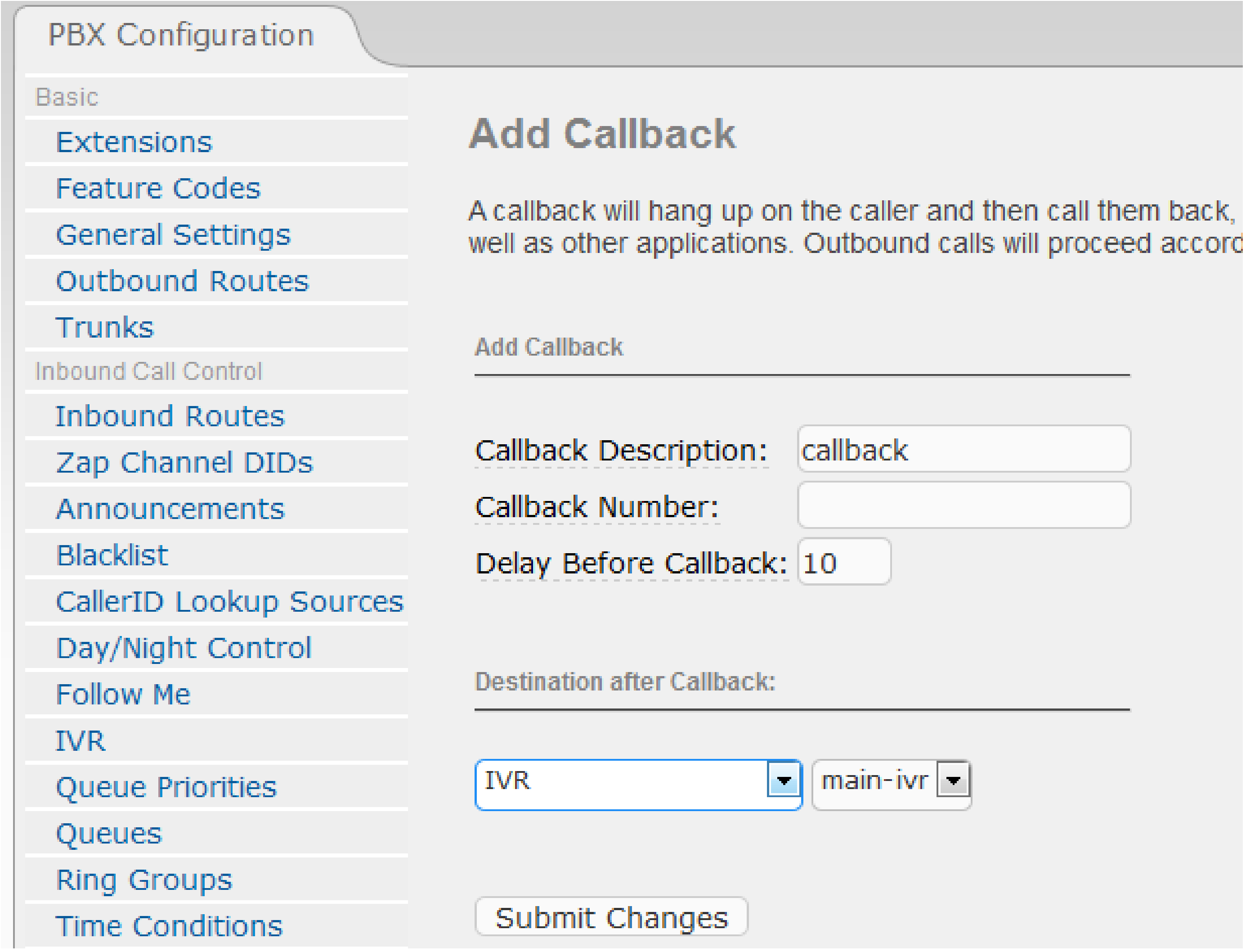
Return to IVR: if the caller was sent by an IVR to this message, the next determined destination (last option) cancelled and the caller will be back to the IVR. So we determine in Destination after playback that after playing message caller directed to which destination.

5.16 Call back

A callback will hang up on the caller and then call them back,directing them to the selected destination. It is used in situation that the caller cannot access to the VoIP endpoint and don't want to pay the cost of long distances. This is useful for reducing mobile phone charges while inbound calls are significantly cheaper than outbound calls. Destination of callback can be any resources defined in PBX (like extension, voicemail, IVR, queue or…) or be used like a complex with DISA, which explained later, that caller receive a beep sound and be able to dial.

For having a callback click on callback in internal option and configuration and you should fill four options in next part.

A) Callback description
B) Callback number
C) Delay before call back
D) Destination after callback



In callback description a name is determined for call back setting.

In call back number a telephone number is put that a system dials it for connecting a call. But you should consider that the number format should be the same with one of the formats of call back (for example if you don't have any format or pattern for 11-digit numbers and enter 09121111111, callback setting doesn't work and assumed as incomplete number). If you put this part empty, the number of the caller ID of the caller is used for calling (be sure that the caller ID be correct and don't registered like "unavailable" or "unknown".

If you set a time in Delay Before Callback, there will be a delay (in second) between cutting a call and the process of starting the next callback. It is obvious that this part is optional and you can leave it empty. But consider that most of the telephony systems have some seconds wasting time to be ready for call back after cutting a call, so it is good to wait some second that the phone (mobile) accept the next call and not to be busy. Destination after callback is used for setting a call target that should transfer the caller to it after terminating the call. In this part any call destination is usable.

After setting click on submit changes and apply configuration changes. So the call back is saved and activated.

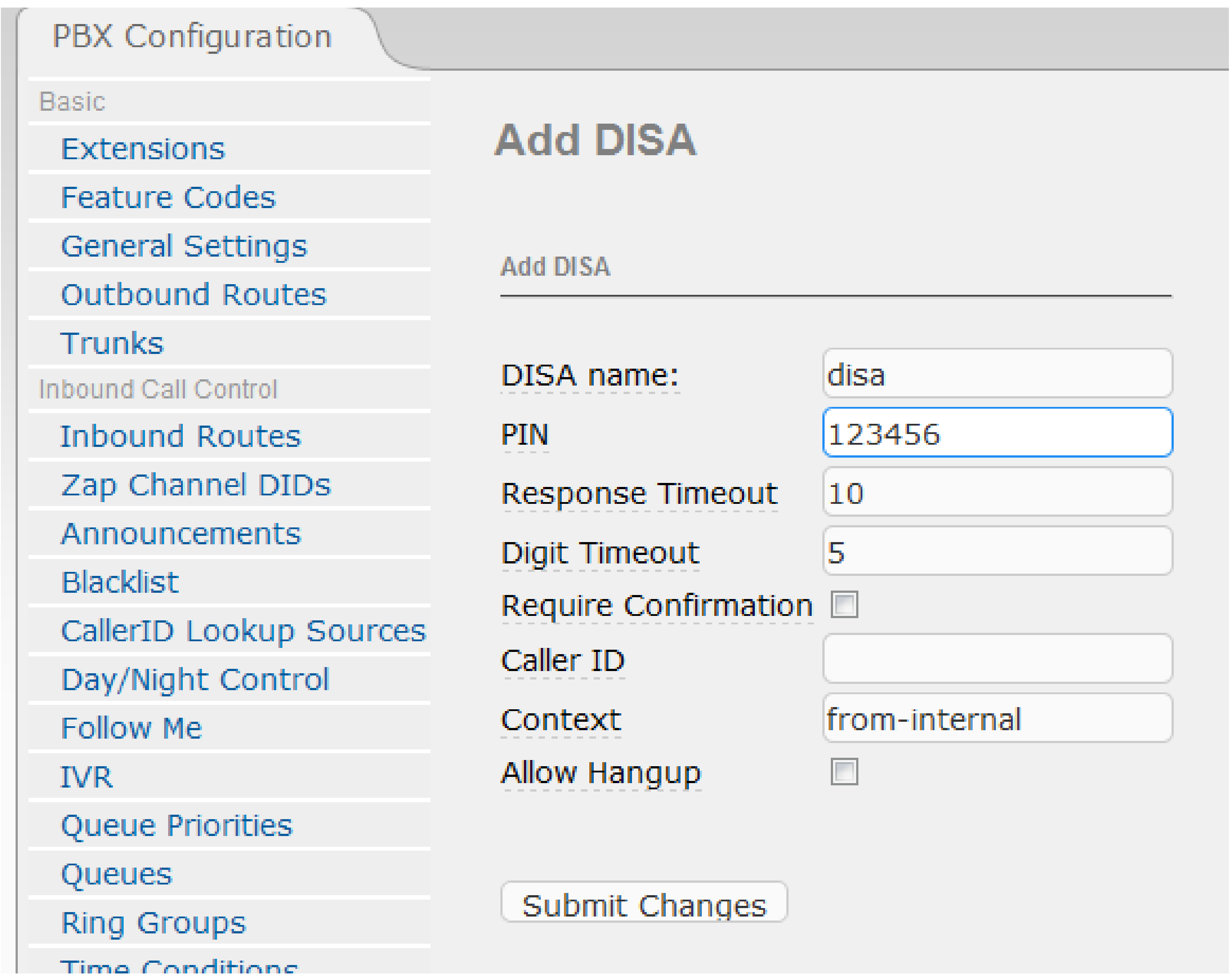
A few cases of application of the callback can be summarized in the following:

- Consider a company that its employee could access to the voicemail from everywhere. Calling with toll-free of company for checking the voicemail cost a lot for company. In this case after hanging up on any caller, call them back and direct the conversation to the (Misc Destination) 98*. The callers receive a call from the number they dialed and they extension number and the password is asked from them, so they can access to their voicemail.
- There are some companies that the cost of their VoIP trunks of the calls is cheaper than calling with mobile of staff with company. Mobile of the staff doesn't have the fee of dialed calls. In this case a callback is set for each of employee that after cutting the call, call back to their mobile phone and DISA destination give a beep sound from PBX; in the other words, by dedicating one of the VoIP trunks of a company, he is allowed to call with no cost with his mobile phone. In next chapter we will discuss fully about DISA.
- Large companies need to collect calls from all over the world (for example a credit card company that needs to get all the missing contact card, forgetting the password or stolen password). By using the VoIP trunks of the country that caller called it reduce the international cards significantly. In this case it calls back to the caller and put him in the call queue. Callers receive a call from the

number they called and connected to the representative of the company once prepared.

5.17 Accessing to dial tone from outside (DISA)

DISA is used when a free line (beep sound) is dedicated to a caller from PBX. By having beep sound a caller can use all the accessible endpoints of VoIP. It means that a caller that hear beep sound can dial any extension he wants, check the voicemail or even dial from the attached line of PBX. For having DISA or the service of accessing to dial tone from outside, click on menu in left side of the page on the DISA, below the internal option and configuration as follow:



In the following eight options will:

DISA name: it is used for specifying the settings of DISA in different part of system as call destination.

PIN: it is used for recognizing the caller when accessed to the DISA call destination. If it is not empty, it will ask to enter pin code from the caller. If it isn't equal with the pin, the call will be terminated and the caller is not allowed to access to the DISA call destination. By the way, PINs can be separated by comma. (For example by entering 9012, 5678, 1234, caller will hear beep sound by using each of 9012 & 5678 & 1234 numbers).

Response Timeout: it determines the time (in second) that system is waiting for receiving an entrance from the caller before terminating the call. This time is for entered and to be entered numbers. This time is considered 10 second as a default but it is not enough and the caller will be in hurry. 15 or 20 second is enough for entering the numbers without waiting too much.

Digit Timeout: it determines the time that system should wait between entering the numbers and dialing. It means after entering the numbers by caller and the time he is waiting for connecting, how long (in second) a system should wait to send the numbers to Asterisk for connecting a call? The default time is 5second and it is enough for the callers to enter the numbers. The caller won't last more than 5second for entering the numbers.

Require Confirmation: when this option selected, system ask the caller to press key 1 whenever he hear beep sound. If it doesn't, the call will be terminated. This option is usually used when the caller is sent to the DISA from inbound route directly. In this case the caller immediately will be sent to the destination without any beep which timeout quickly and the call terminated without the caller being aware of what happened.

Caller ID: it is used when we want to consider that the outbound route be something other than caller ID of the caller for the DISA call destination. The format of this option should be like <##########>

“caller name” that instead of the caller name should be the name of the caller and instead of <##########> telephone number of the caller (for example <09111234567> “ Saeed”). If the option was empty, the number of the caller should be sent for destination as Caller ID. (Likely the number that you put in this option won’t be sent to destination. So it is better to leave it empty, this setting depends on your trunk, for more information refer to “trunk” subject).

Context: it determines the context that Asterisk put the caller in it in DISA calling. (More information)

Allow Hang up: if it was chosen, the caller can enter the terminating code (** as a default) and cut the call and in this case hear the beep sound. It can assure the caller that the call is terminated.

After finishing the setting, click on submit changes and for confirming (click on orange border of DISA) apply configuration changes and it is activated.

6. Fax Server

As you know Elastix have a good and powerful server for sending and receiving fax. This fax server use Hylafax software. Elastix can receive faxes from difference ports such as LAX, SIP & ZAP trunks.

6.1 Making a IAX Extension

First of all you should define an IAX extension. This extension is like a modem for connecting Asterisk and Hylafax. For any fax server you need a lax extension so go to the call center, telephony system setting, and extension.

Choose Generic LAX2 Device and click on submit.

The only setting that should be entered or changed is as follow:

User Extension: extension number

Display Name: a name

Secret: extension password

Record Incoming: put it Never

Record Outgoing: put it Never

Now click on submit, your extension or your modem is ready!

6.2 Visual Fax

In second step you should make a visual fax. Go to this menu. Fax, visual fax. There is some information required for making a visual fax which is your fax server either:

Visual Fax Name: "a name"

Visual Fax Name (LAX): "The lax extension you made"

(LAX) password: "the lax password you made!!"

Destination Email: "the email you want to use for sending fax to it"

Code of Country: "can be 98"

Code of Area: "can be 021"

Caller Name: "name of your company"

Caller Number: "phone number of your company"

After making this, go to the list of visual fax. That should be shown here and Running and idle on ttyLAX1 should be written in "position" field. Now your system is ready to receive fax. For testing you can call to the lax extension and hear the sound of fax. On your IVR you can consider a key like 6 for the fax, anyone press the key 6 will be connected to your fax. You can use this way for making more faxes and have several visual fax system.

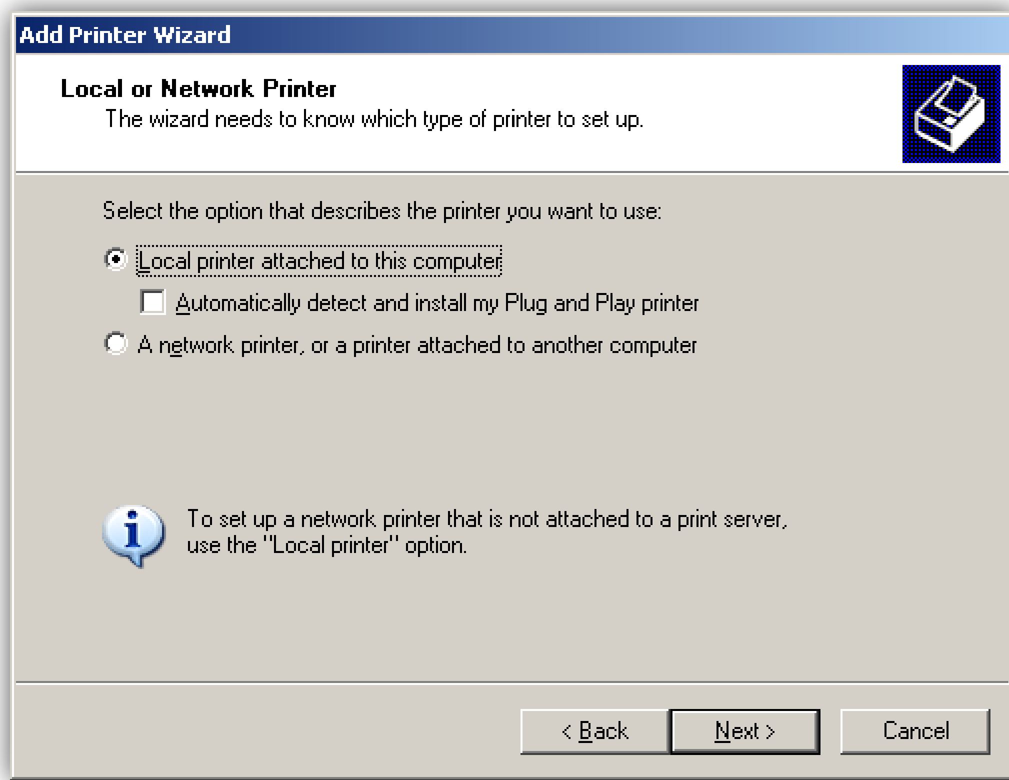
6.3 View the Faxes

Received faxes are like PDF and after receiving in fax menu, view the fax can be seen.

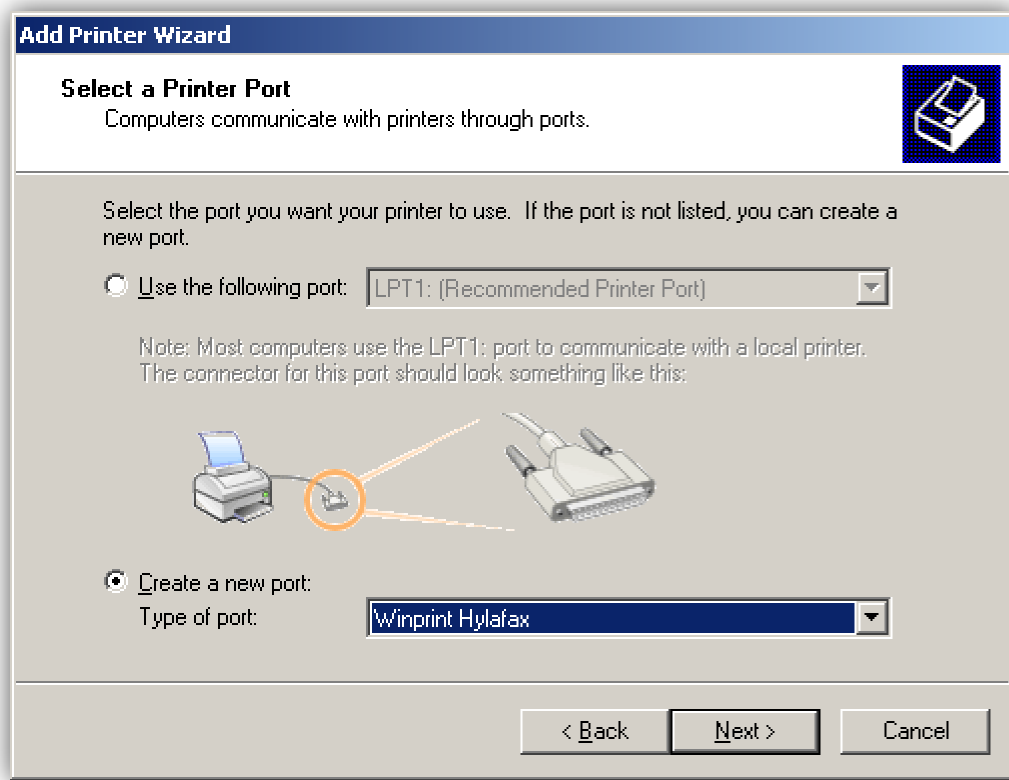
6.4 Programs of sending fax

- Sending fax by Winprint HylaFAX:
  Winprint HylaFAX is a windows GUI interface for sending fax through server. By using this software you can fax your data, as easy as printing, with giving destination fax number. For receiving this software please refer to VoIP-IRAN in sourceForge, files, utilities and fax.
  https://sourceforge.net/projects/vaak/files

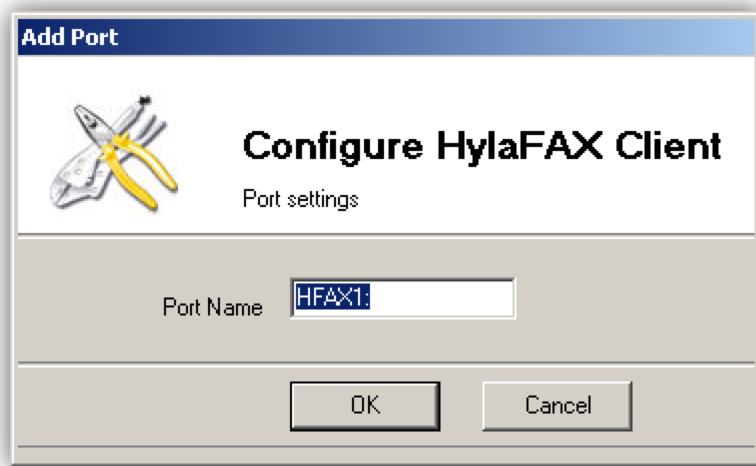
By installing this file port of WinPrint HylaFAX is added. After setting up the exe, go to control panel, add printer and click on printer add a. I the page below choose Local Printer but since it's not a printer don't choose install my plug and play printer automatically detect and.



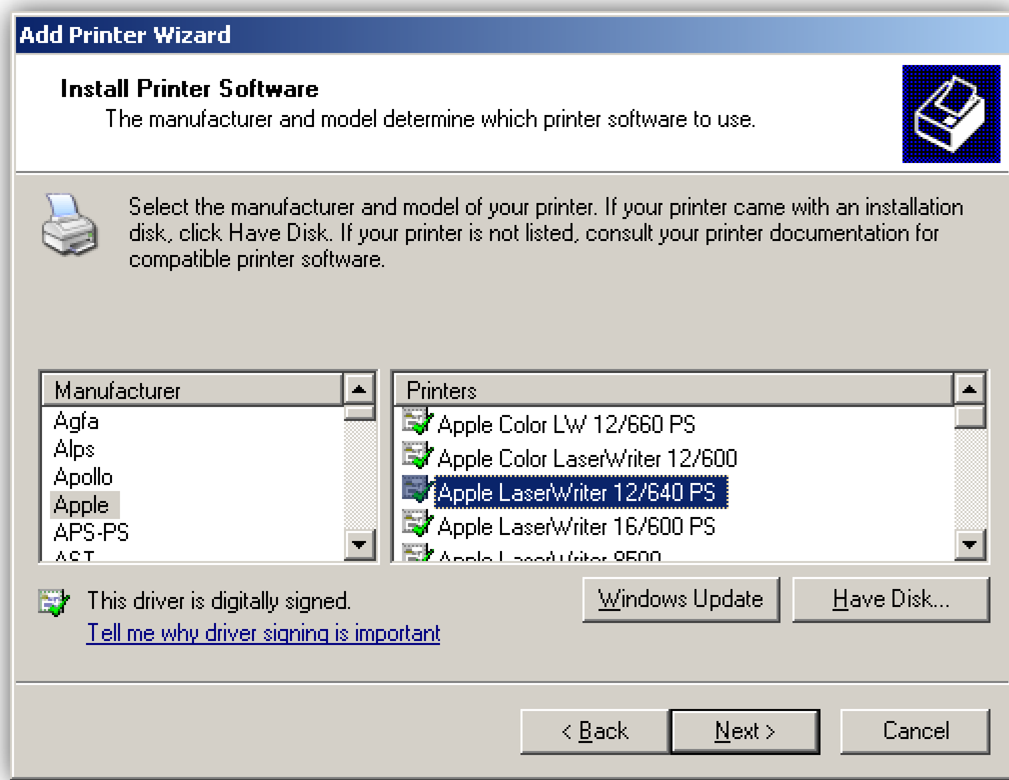
Click on Next and in Select printer port, choose create a new port type. If it is successfully installed, you should see the Winprint Hylafax in the list. Select it and click on Next.



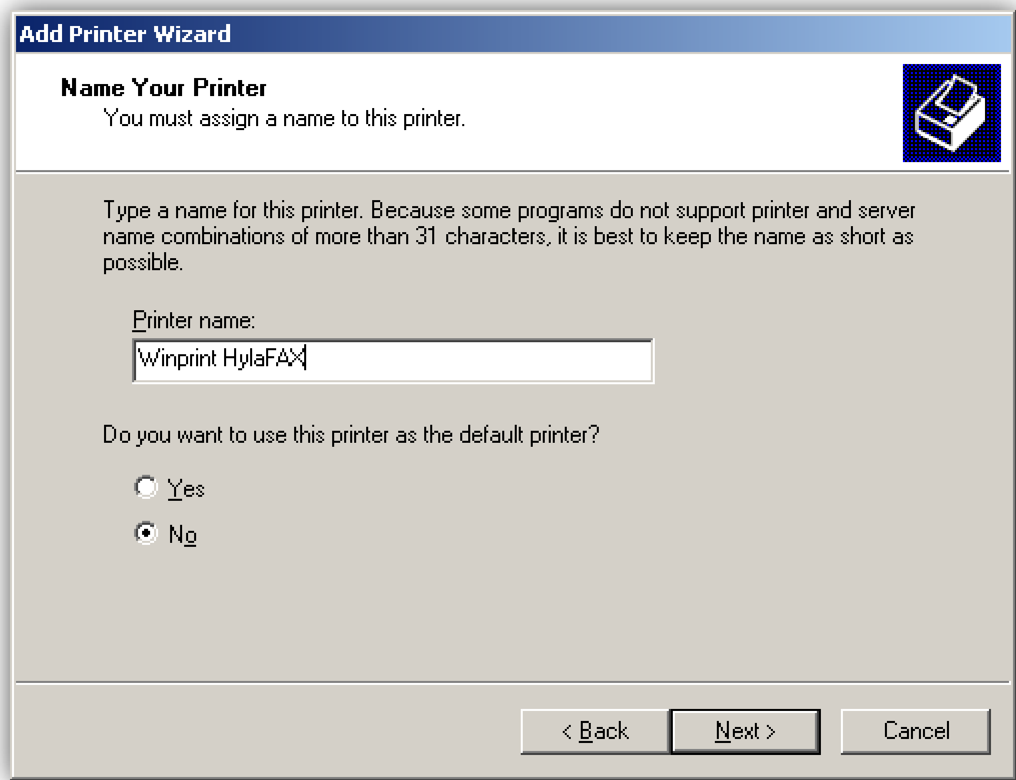
The naming box will appear as above. The name is not important but it is better to use :HFAX1 to determine it.



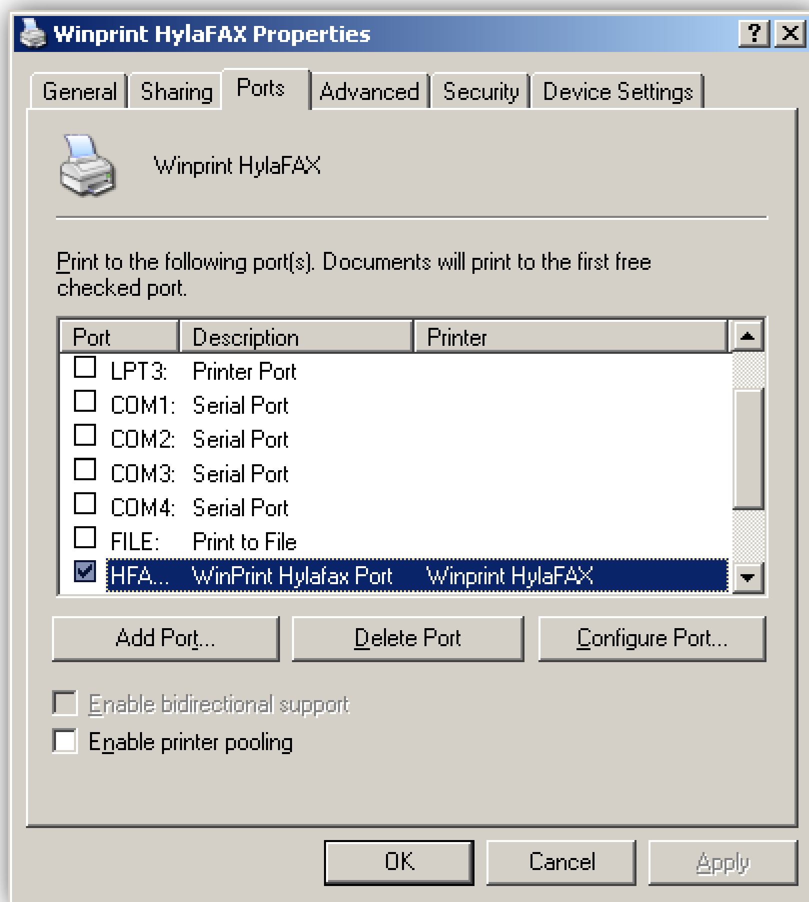
Choosing the type of printer, as it was shown choose Apple LaserWriter 12/640 PS from Apple.



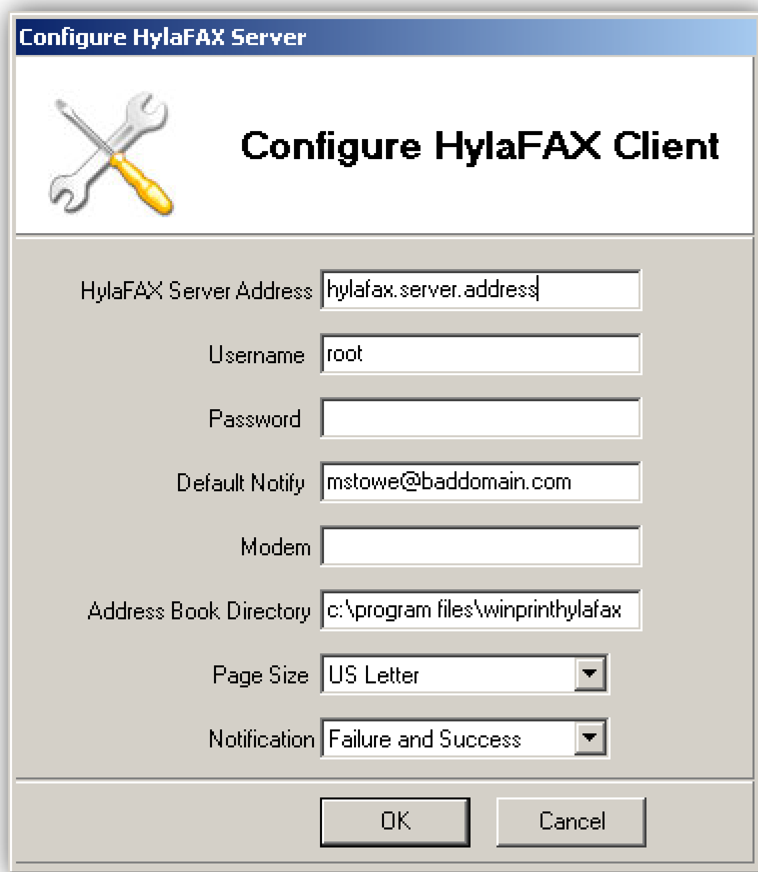
You can name whatever you want.



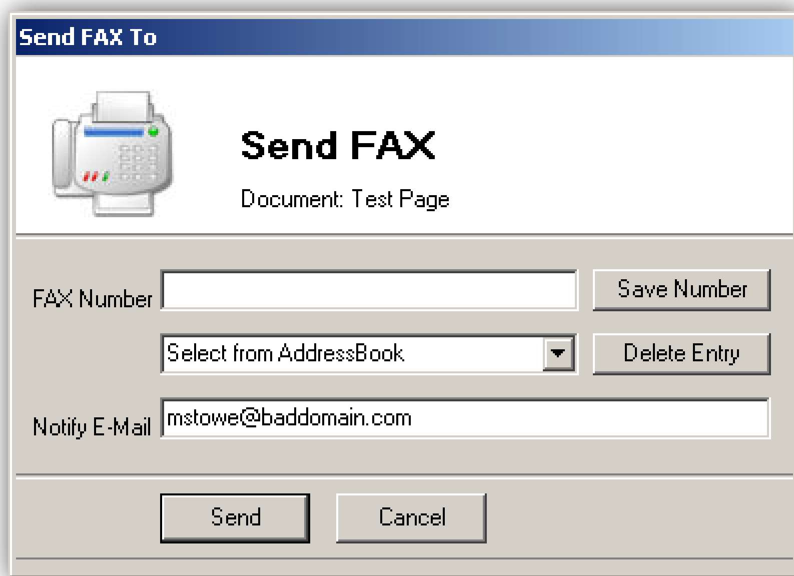
Don’t share it. You can install Winprint HylaFAX on any system and directly connected to the server of Hylafax. Don’t print the test page because you have not set the printer yet. When you install it, right click on it and choose properties. Find the tab of ports and tick the port you have made.



Click on configure port for connecting with fax server.



Enter IP of the server in hylafax server address and write the username and password of the server. One of the problems that may occur is not having permissions to access the hylafax server. To solve this problem go to var/spool/fax/etc/hosts.hfaxd/ and add IP of your computer. Now you are ready to send a fax. For testing choose any application you want and click on print and choose your printer, winprinthylafax. Whenever you send something to this printer, the page below appears:



Enter the fax number totally. Fax will be sent by halafax and its server send an email to confirm the address entered to Notify E-mail.

7. Callcenter module

7.1 introduction

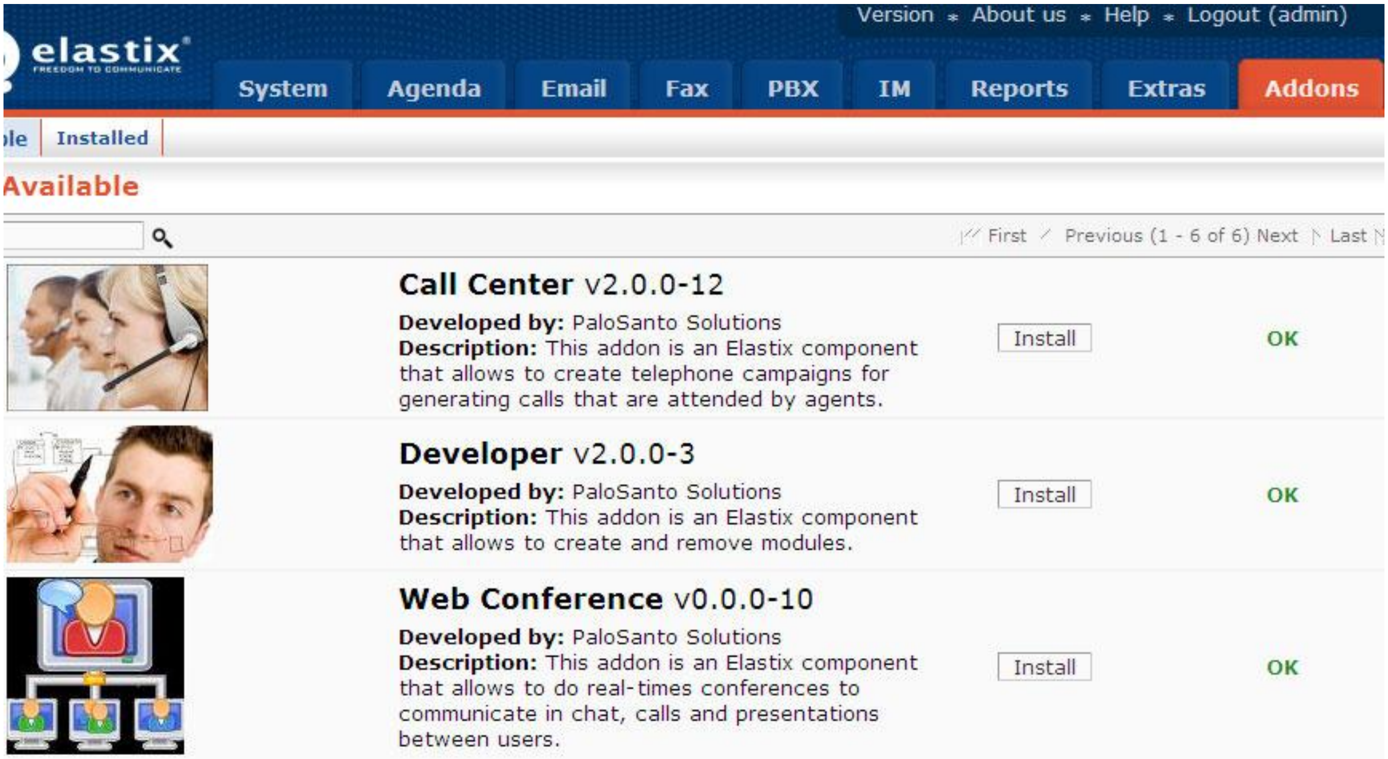
Call center module is one of the add-ons of Elastix that can be installed optionally. This module allows having a call center with dynamic queue by defining an agent. This module has different part and services such as browser page of agents, do automated calls advertising (Telephone marketing), Very detailed reports of agents, their operation and performance and many other useful items that will be explained in next chapters.

7.2 Concept of call center

Generally the concept of call center is systems that can service the callers with limited number of receptionist. Normal Pbx systems are not in this category. The most important feature that distinguishes a call center with a normal pbxs, is Queue. It allows the system to service the callers with much greater number of receptionists.

7.3 installing call center module

After installing Elastix go to add-ons menu and wait until it connects to the Elastix. Note that it may take several minutes; by the way be sure that your server is connected to internet. After connection click on Install.



After finishing the installation you can see the added menu of call center in main menu.

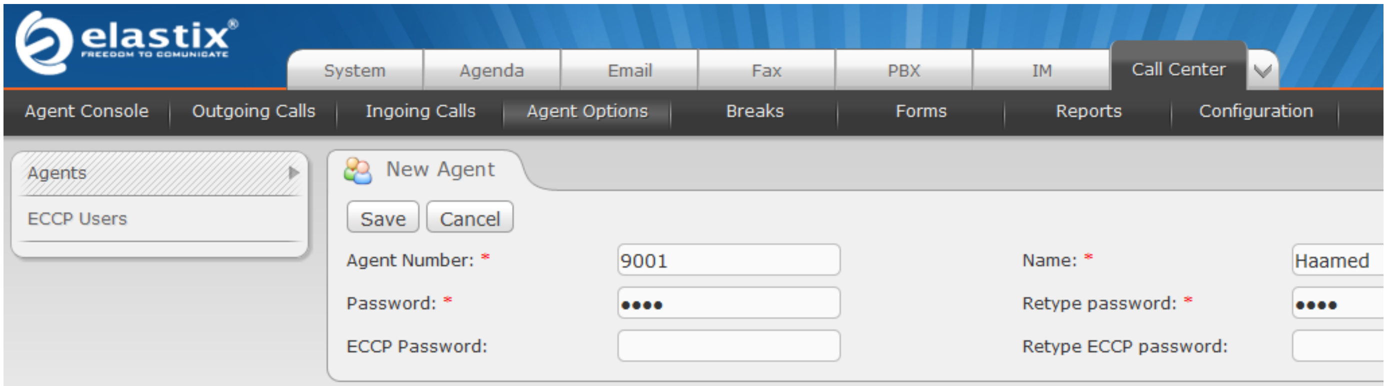
7.4 use of call center

Call center module has two applications:

- **Management of ingoing calls:** it is receiving call from queue and accountability of it to agents, in this operation there are some application for agents which will be explained later.
- **Outgoing calls:** in this section you can determine some numbers; call center automatically call them and connected them to the agents. About the application and how to activate will discuss later.

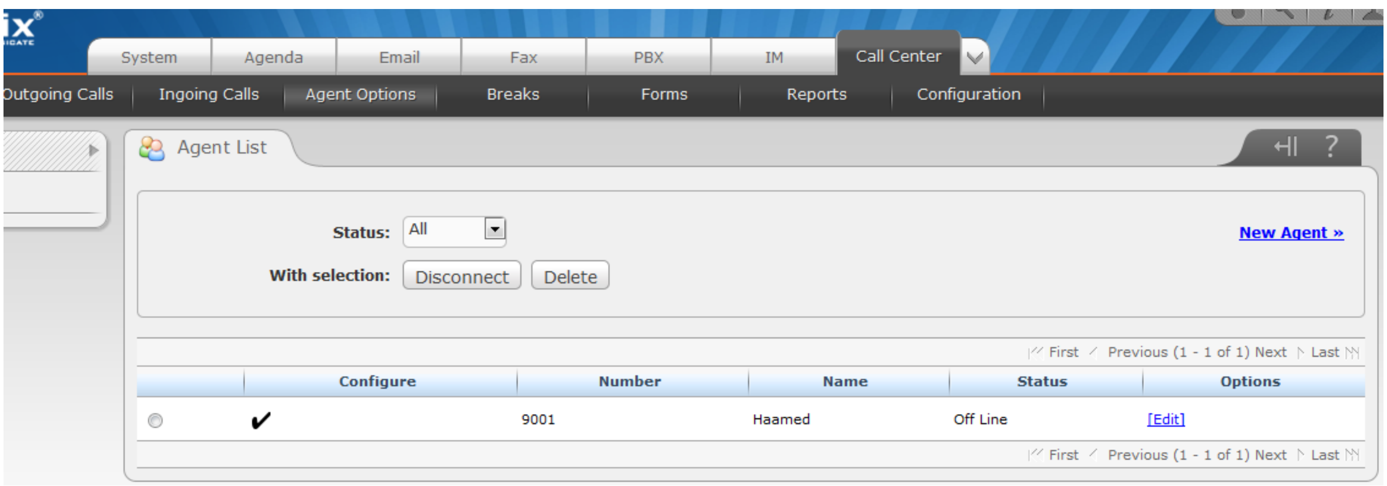
7.5 making agent and its configuration

I'll try to explain the menu based on priorities and performance. In first step we should make an agent. Go to call center menu, agents, click on New Agent. The page below will be open.



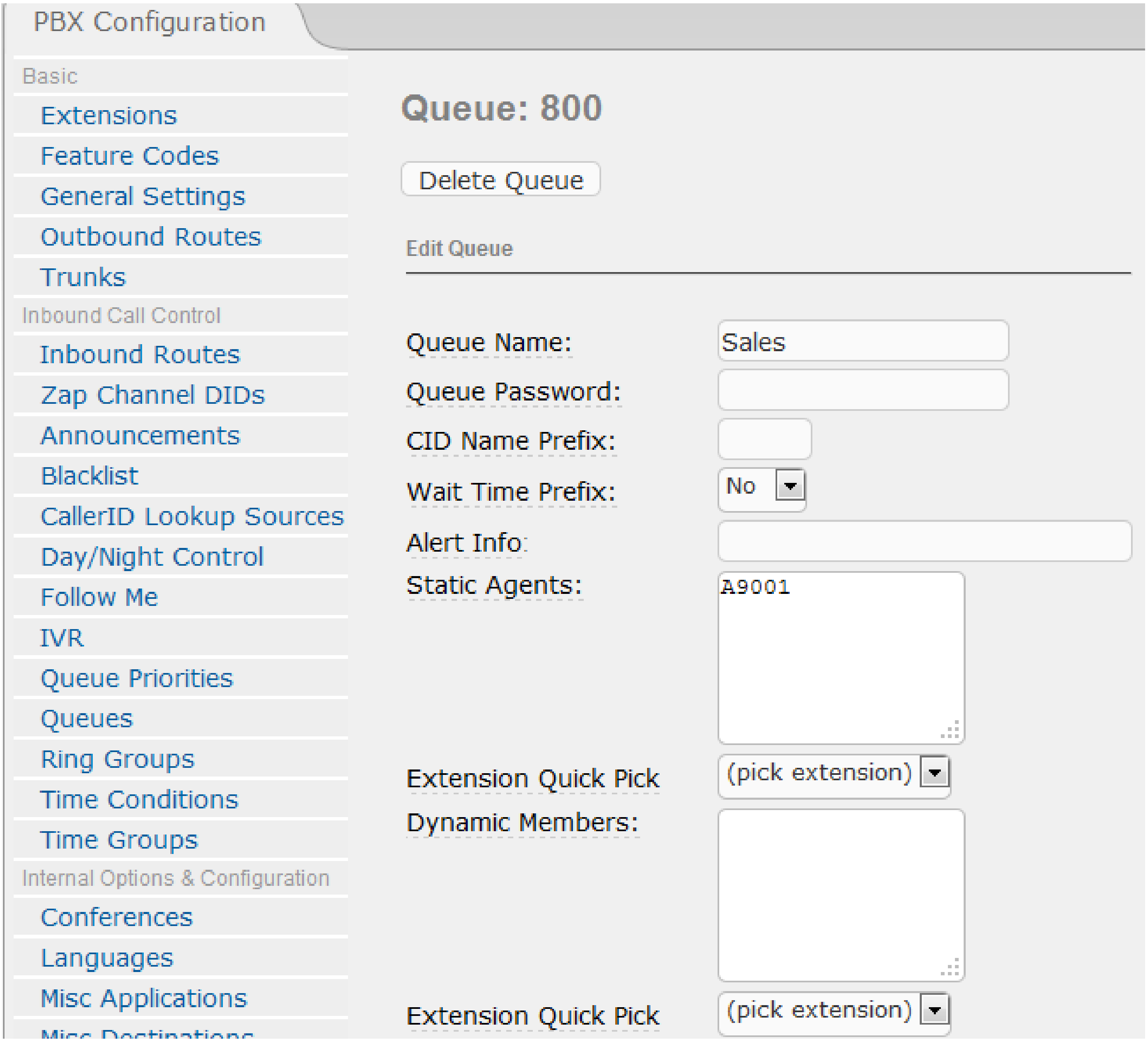
- **Agent Number:** unique number for agent that can be entered to the agent console. It is good that your agent number be more than 3 digits (it is not required).
- **Password:** password for agent number.
- **Retype Password:** enter it twice for reducing the risk of confusion.
- **Name:** a name for your agent. It can be good.

After filling out and saving, you will see your agent. It will be offline because you don't enter through console.



It is the important section and cause many cannot install a call center. In this stage you should enter your defined agent in a predefined queue in menu of telephony system of PBX.

For this purpose we enter to the queue and write the agent number with an A character before that in a static agents as follows:



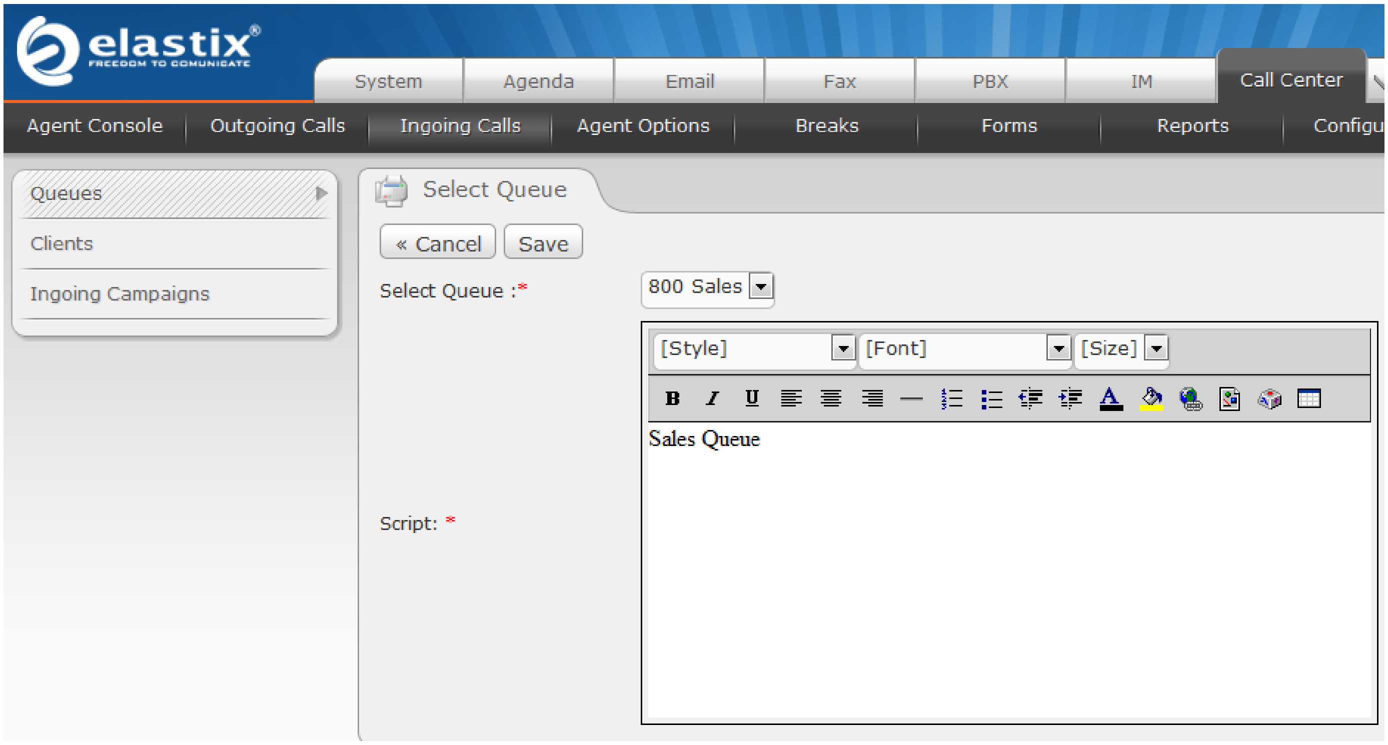
Your agents are in the queue, but still there is something to set up a call center with the least setting and it is adding your queue to call center module.

7.6 ingoing calls

It has two sub-menus which one of them was used to define queues in call center and the other for entering the names and numbers you work with (your clients).

7.6.1 *Queues*

Queue is a part of a system that manages the ingoing calls and connected to the agents. In this section you can add the queues that are defined already in telephony system menu (PBX) & Queue and active or inactive them.



A. Select queue:

- **Select queue:** select the queue from the list.
- **Script:** text played for agent.

B. View the queue:

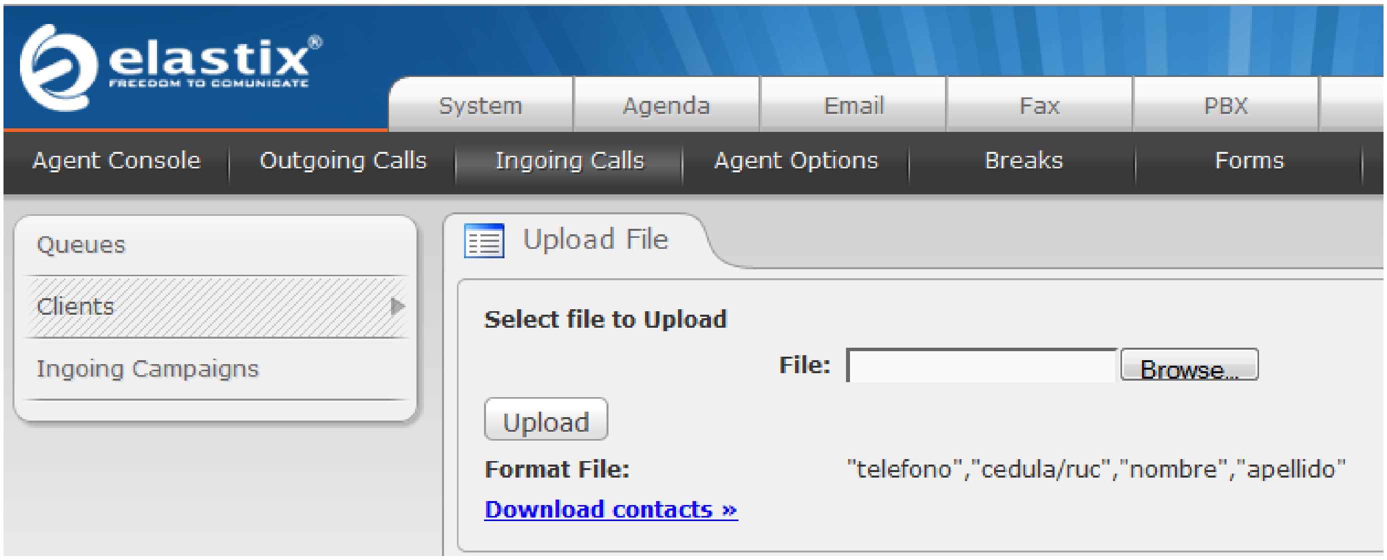
- **Queue list:** Queue list that transferred the ingoing calls.
- **Queue:** name of queue
- **Status:** being the queue active or inactive
- **Options:** there are 2 selection: one of them is "view" for viewing the details and likely changes in queue and the other is "activate" for activating the deactivating queues.

7.6.2 *Clients*

In this section you can save list of clients and their phone numbers. In fact you make a phonebook. Call center module use this when receiving calls to show the details of the caller.

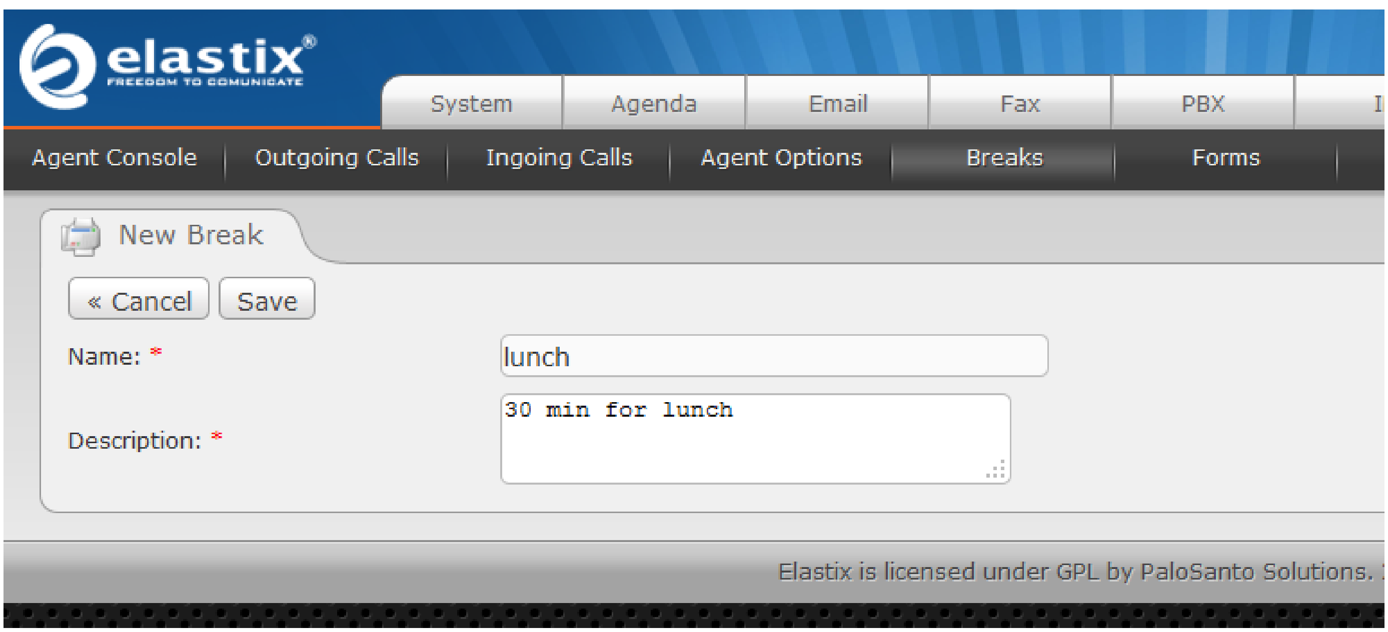
You can load your list in a CSV file in this format:

"Phone number","company","name","last name"



7.7 Breaks

In Break you can clear a reason for resting or exiting the agent. Agents can be free for a while by choosing Breaks for example we define Breaks for lunch time of an agent.

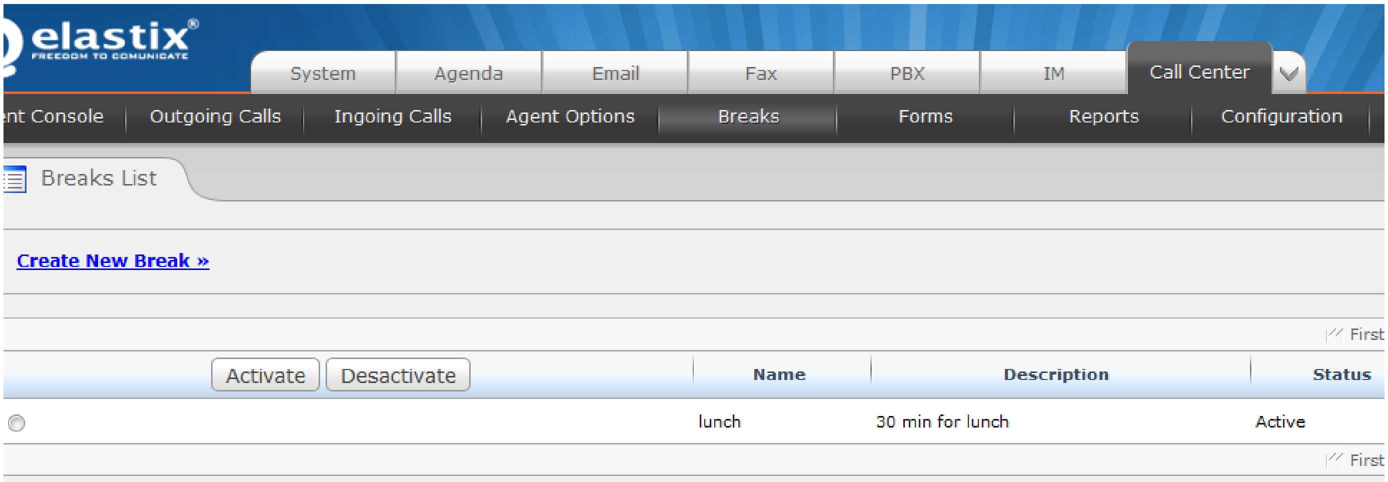


2parameters is asked for making Breaks:

- **Name:** name of your Breaks such as lunch time, resting, emergency!!

- **Description:** an explanation about the Breaks

After making Breaks you can active or inactive it in options which is activated as a default.

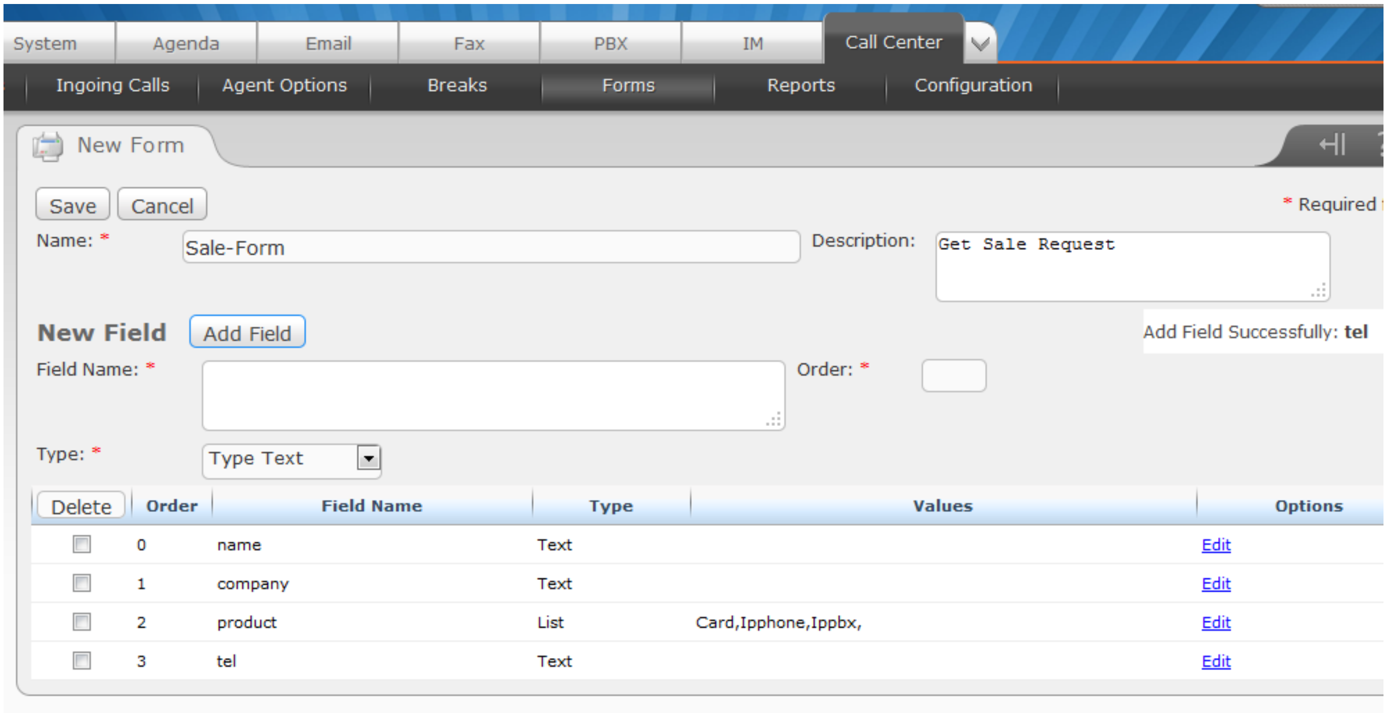


7.8 Forms

You can make a form with objects you have. Use of a form is when a call is connected to an agent, and he can select a form during a conversation with client and include information in it. For example you design a form for order product and agent can fill it during the call.

A. Form designer

In this sub menu, by clicking on Create new form you can enter the necessary information for designing a form. The details are shown here:



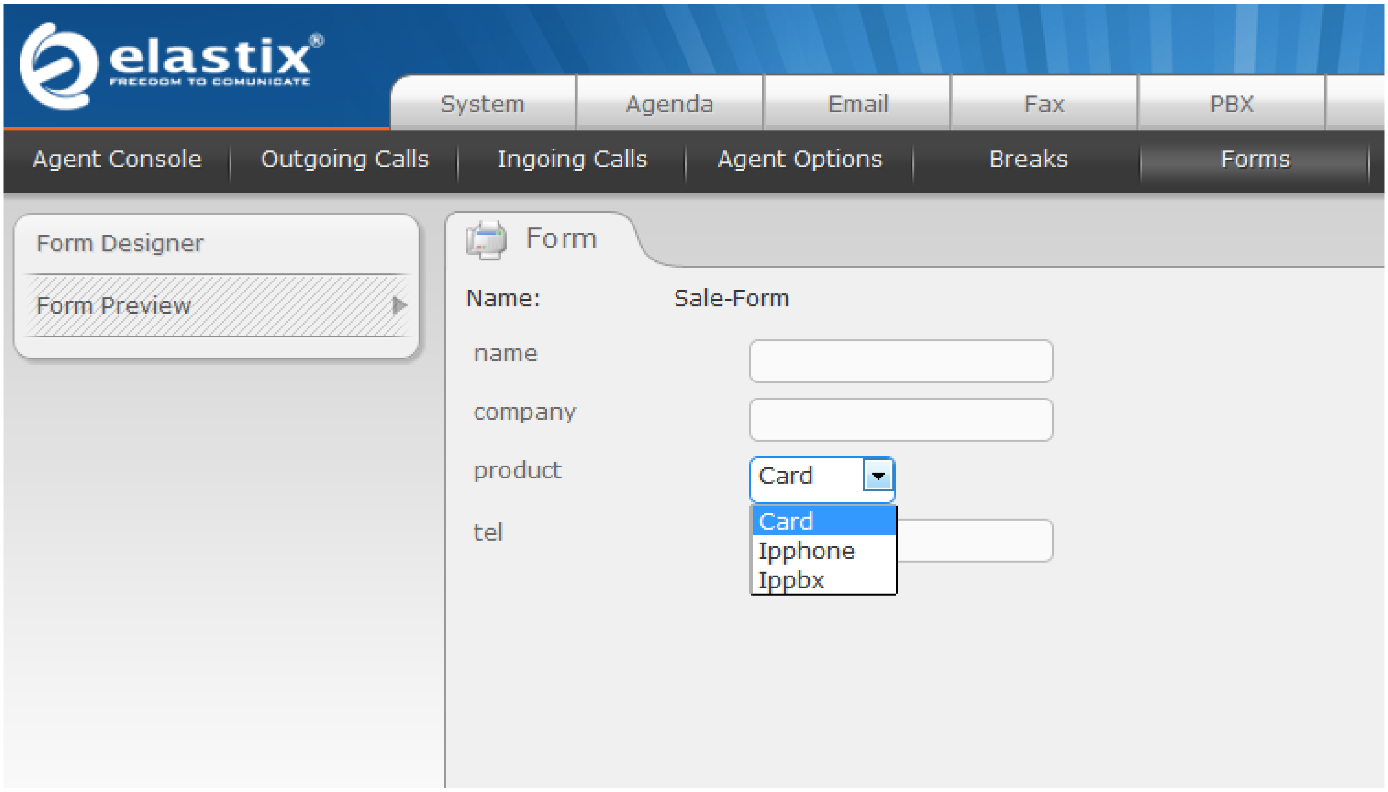
Name: select a name like: selling form

- **Description:** a short explanation about the form such as selling form of phone card
- **Name of Field:** the name of field which is supposed to be shown.
- **Order:** Show the priority of field in the form
- **Type:** type of your field. You can have different type of field according to your needs such as:
  - Type label: it is used for showing a text such as description or giving an attention to an agent
  - Type text: a field with capability of filling a text, short texts such as name and last name or name of company
  - Type list: it can list your products or anything which an agent wants to select one of them.
  - Type date: date field
  - Type text area: it’s a field for entering the long texts such s address or description.

A designed form is shown for selling a phone card:

B. Form preview:

In this sub menu you can see the forms as agent can see. Image below is the form Preview that we made.



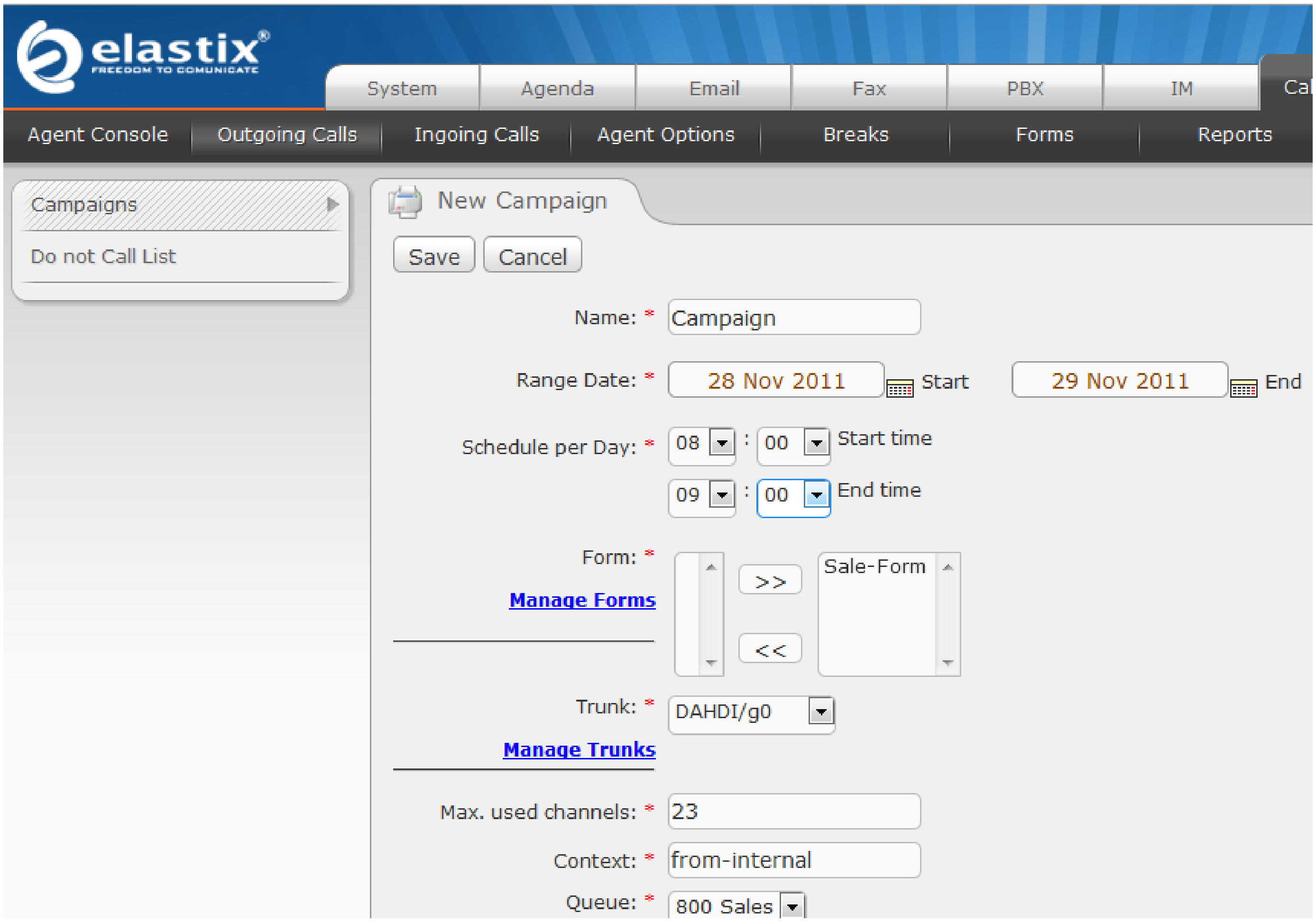
7.9 outgoing calls

A. campaigns:

In this section system calls the lists you have given automatically and connect it to your queue. They name it campaigns because the most widely use of this application is group call or campaigns. Another use of this example can be expressed: campaigns help you to declare an announcement, warning or new services to your pervious customers.

B. Making campaign:

After clicking on Create New Campaign, a form with required field for making a campaign will be open.

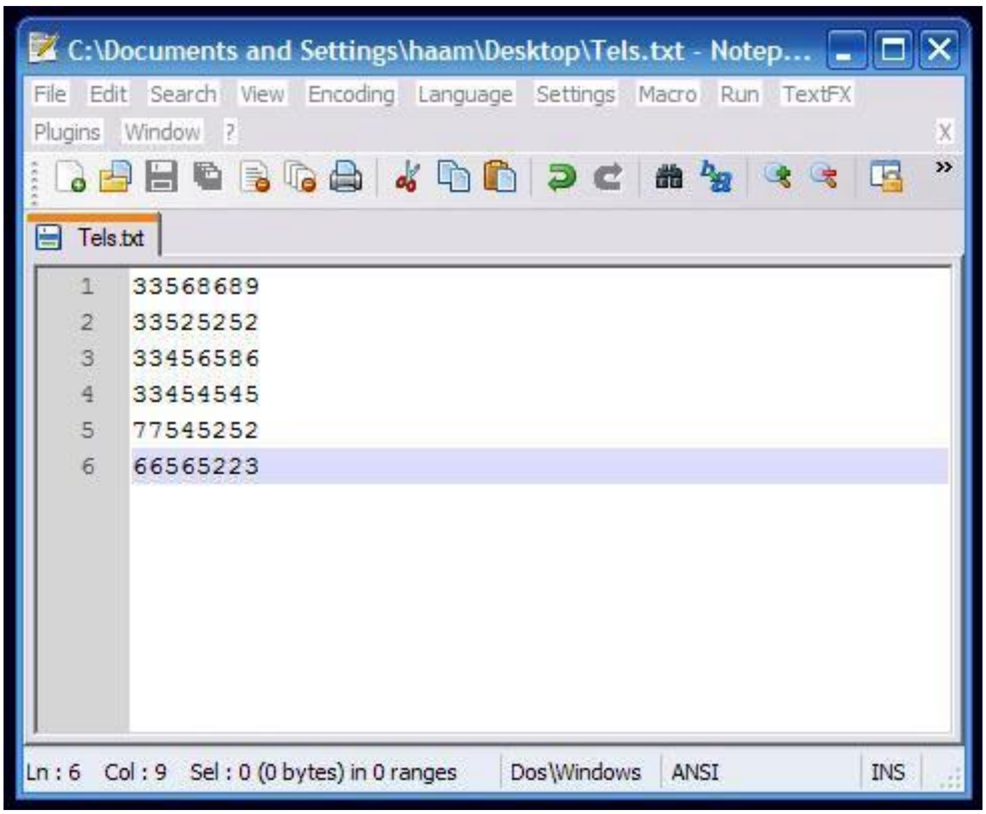


Description of the field is as follow:

- **Name:** a name for this call group such as " new cards of sangoma"
- **Range Date:** determining a time-span for calling, start date for starting the calls and end date so that if the calls didn't end until that date, system wouldn't continue the call.
- **Schedule Per Day:** calls for daily schedule
- **Form:** you can choose either form you have made for collecting information. After connecting to the agent, the selected form will be shown to insert necessary information.
- **Trunk:** selecting the trunk that you want the system use for outgoing calls.
- **Max Used Channels:** it is possible that you don't want the campaigns have negative impact on whole lines of the company. So you try not to busy the lines and consider some restrictions. In this

field you determine the number of trunk channel that system has right to use for this purpose.

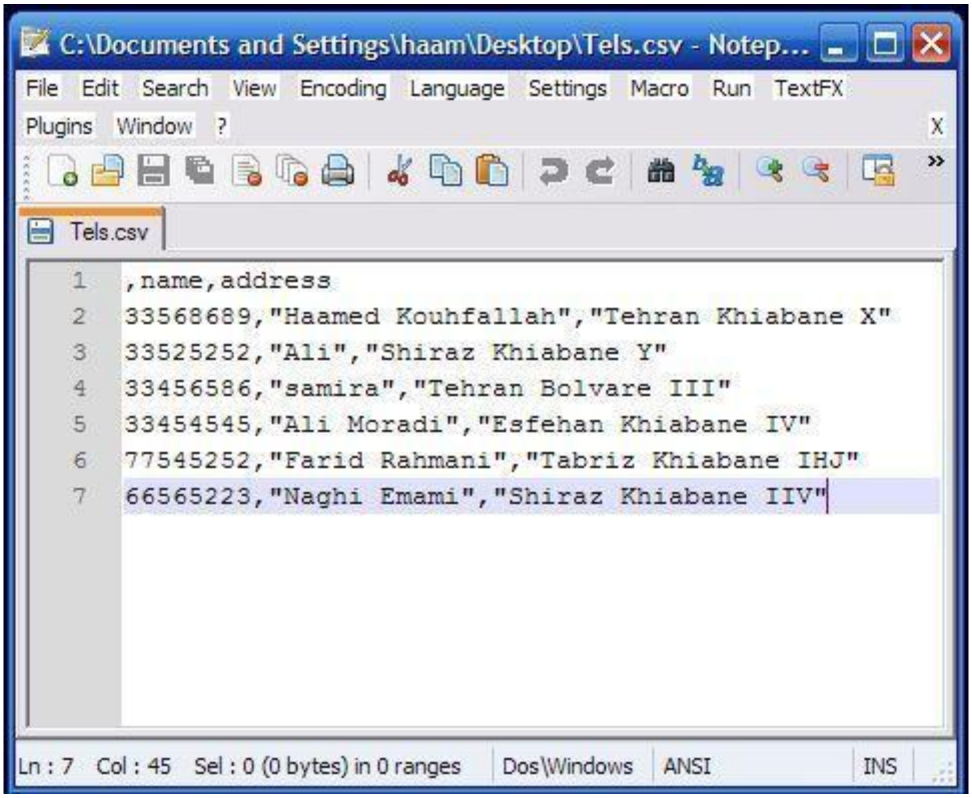
- **Context:** for this campaign the name of context is Form-Internal and there is no need to change it.
- **Queue:** determining a queue for accountability. After calling, system put this numbers into the queue.
- **Retries:** the frequency that system tries to call a number (System for various reasons may not be able to call numbers such as: being the line and the number busy, wrong number and the number not respond…)
- **Call File:** a text file of CSV which includes phone numbers.



Example: a simple phone file

Note: phone numbers should place in a column. If they have titles, the first line should start with ",".

Note: first column should always include phone numbers. So if you have several columns, you don't need to write the title of the phone numbers.

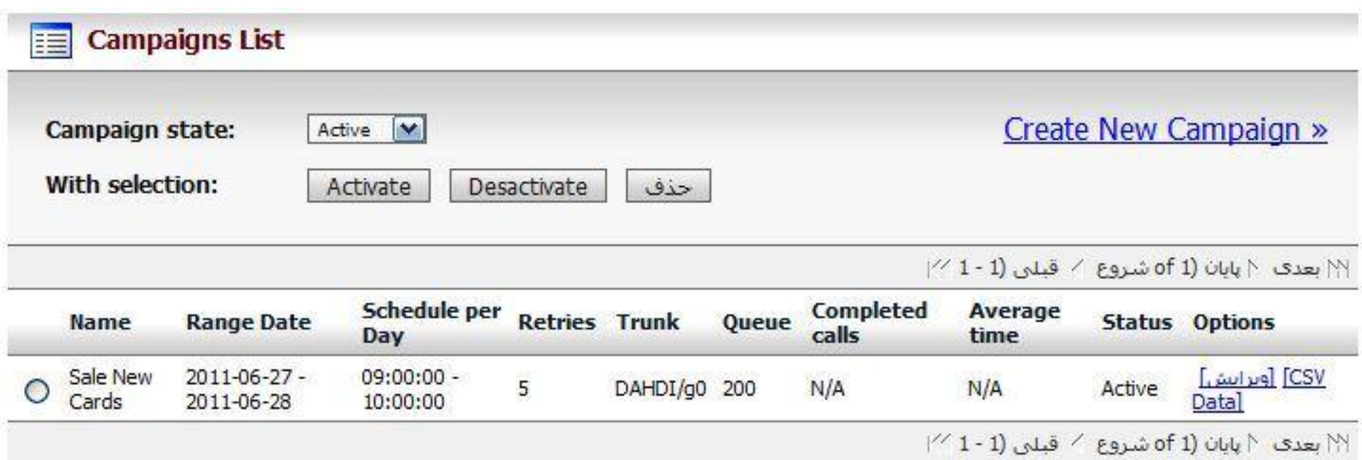


Example: file of customer list with the name and address

- Script: a message for agent when a call is connected.

C. View and change:

The example is as follows**:**

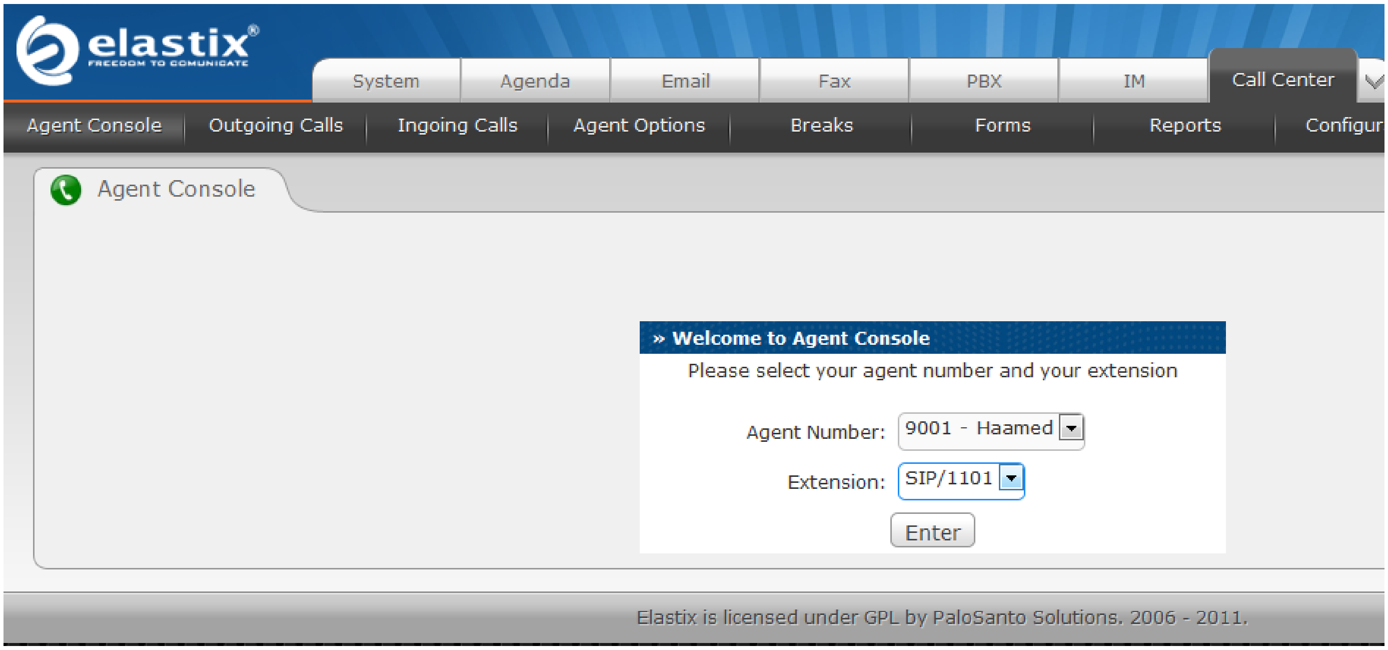


As you can see you can activate or deactivate this Campaign.

10.8 Console Agent:

Console agent is an environment that a made agent can enter with a number and extension. As soon as the agent entered, he is responsible for accountability of the calls in queue.

A. Entrance:



- **Agent number:** it is the same number you made in agent menu.
- **Extension:** an extension number that agent wants to be accountable.

After pressing Enter, the entered extension starts to ring. After answering, the system asks your password that you entered in agent number and console page will be open and you will hear music in your phone until you connect to the first call. In fact an agent should never hang up the phone!!!! So agents use headset.

7.10 Console environment:

After entering the password through phone, console of agent will be open as follow.

(In next version of the book will be completed)

8. Custom Context Module

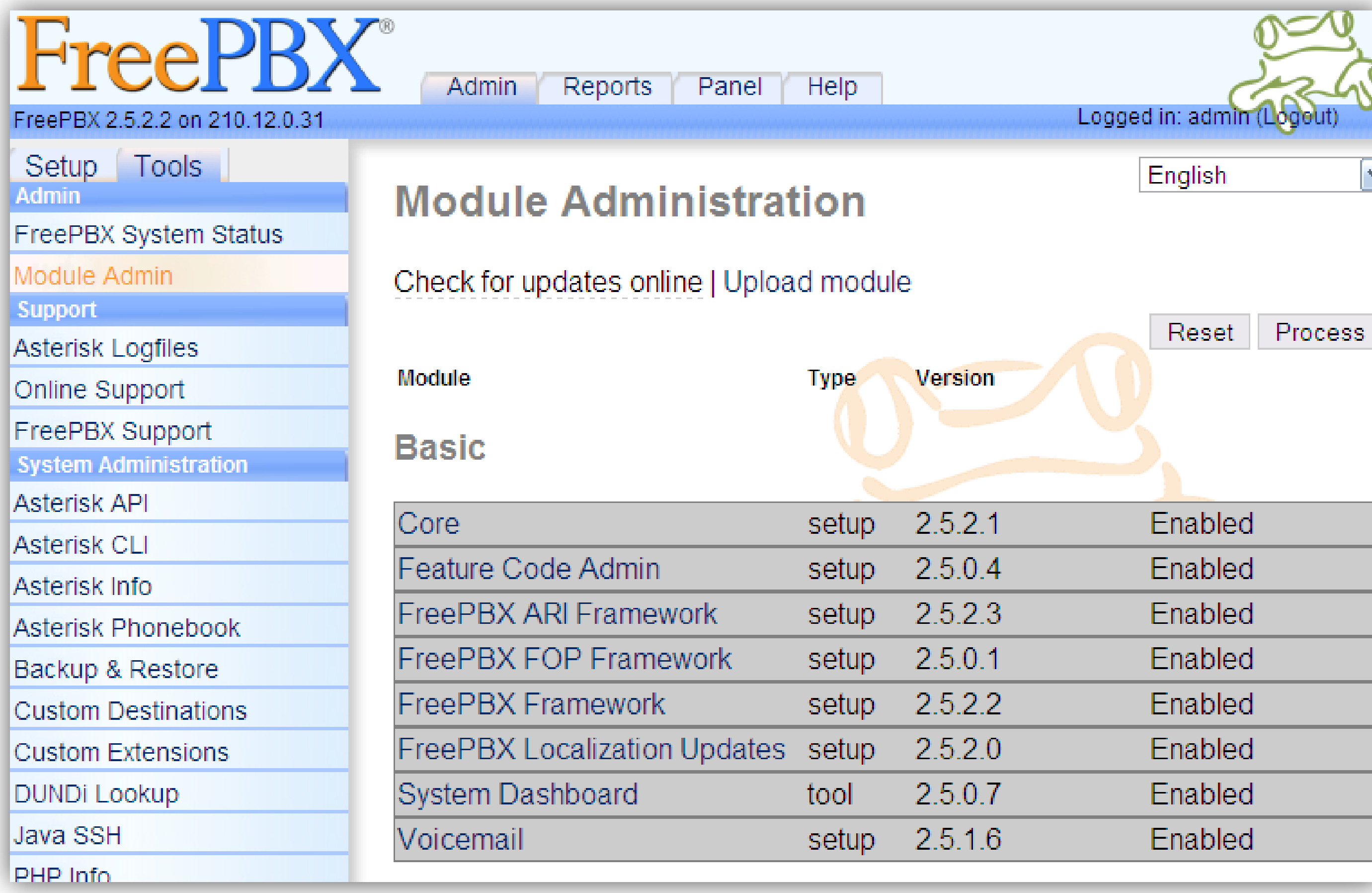
This module is a side add-ons of freepbx which is not installed on Elastix as a default. You should receive it from Freepbx.org or VoIP-IRAN on sourceforg.net:

http://mirror.freepbx.org/modules/release/contributed_modules/

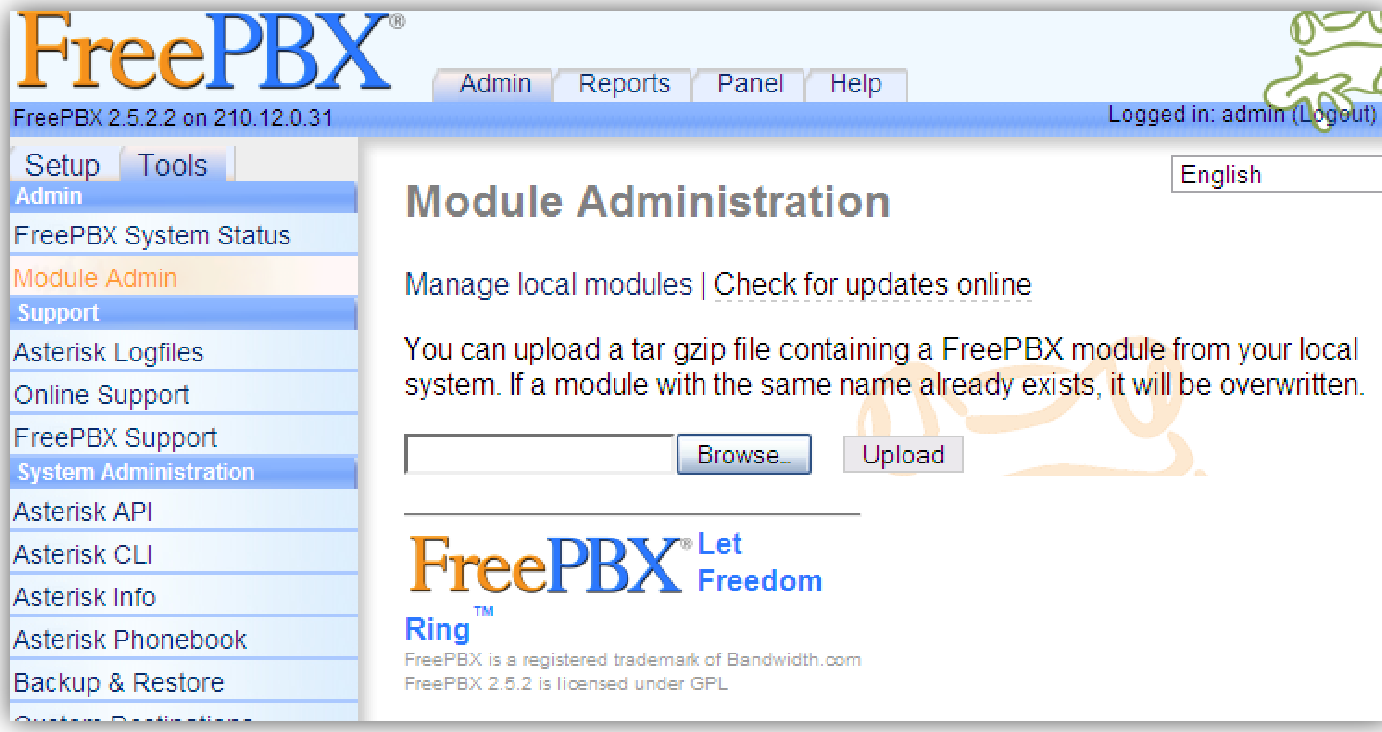
You can classify the extensions with this module and added application to freepbx and supervise on their calls separately. This module is appropriate for those wanted to do have several separated group work on an Elastix server. Other features of this module are: classifying the extensions for having different group on a telephony system, restrict to access some of extensions to other system resources such as trunks, fax systems and…

For installing this module you should download the link and then enter to freepbx. Go to menu of "call center","telephony system configuration" and unembedded freepbx. For entrance use admin user and password (if you use Elastix version 1.6 or older, the password is admin too).

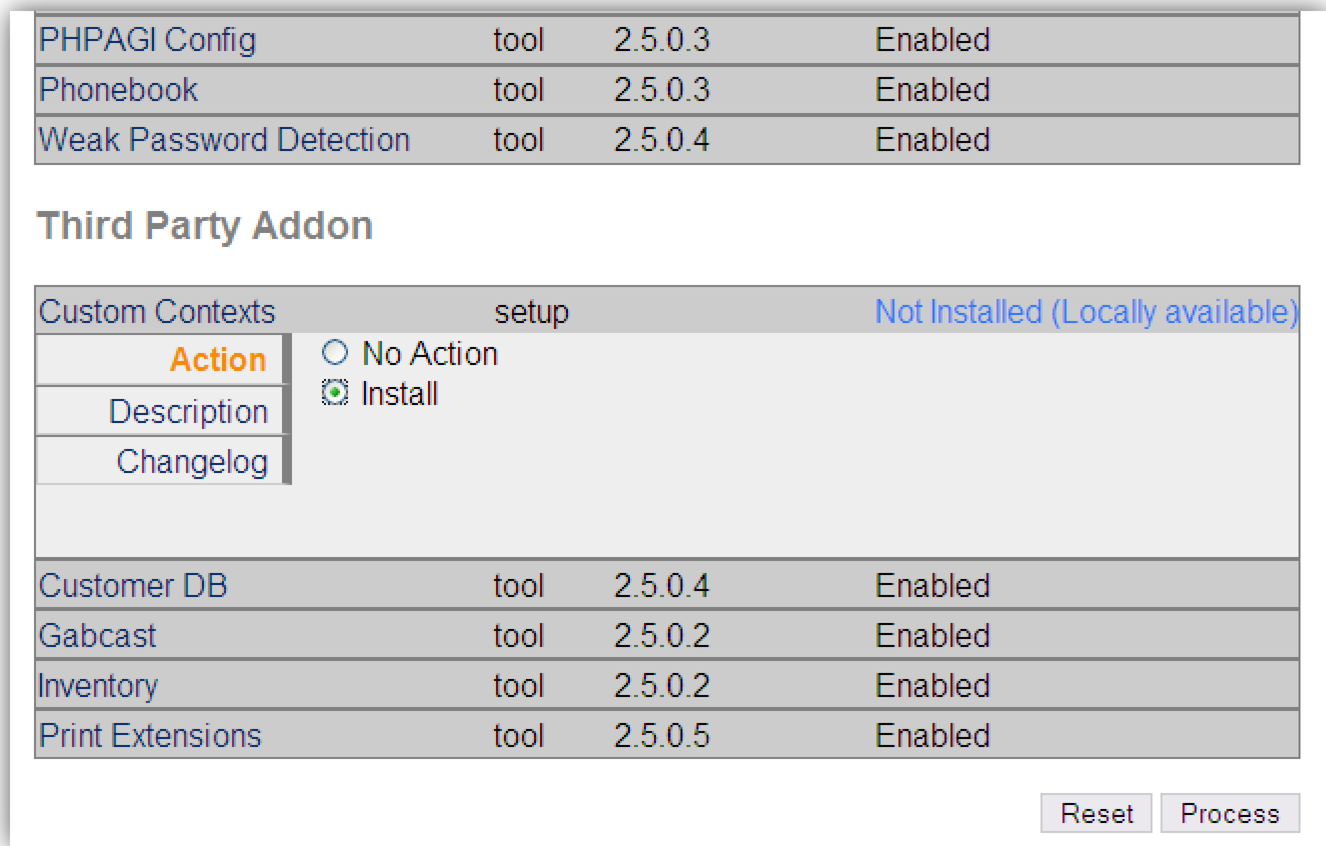
In freepbx for installing the new module you should go to "tools", "module admin" and click on Upload Module.



After clicking on Upload Module you can load the received file of Custom Contexts.



After loading, in menu of Module Admin and list of Modules and in Third Party Add-on, custom contexts is added. Select it, click on install and then process.



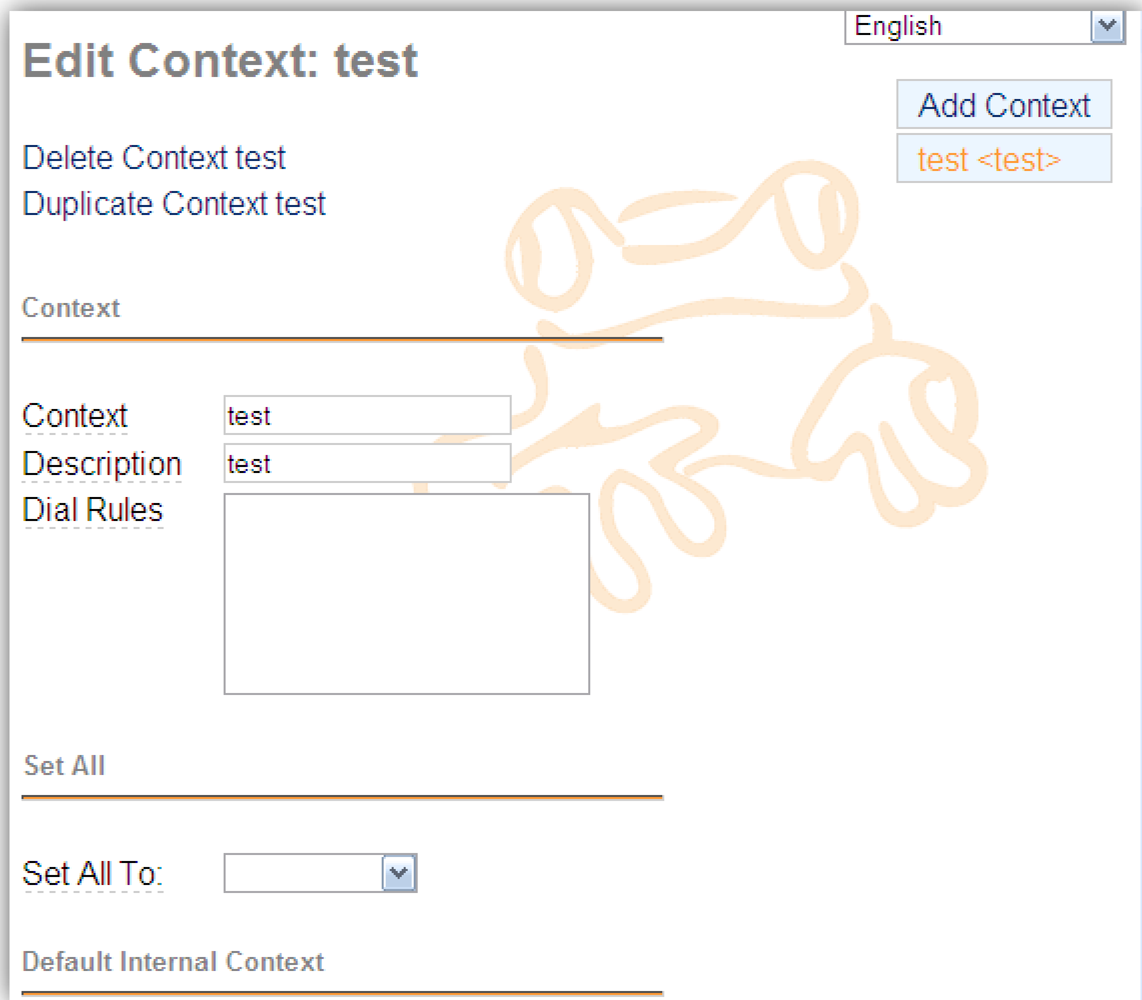
After installing in set up menu, at the end of the list, in third party add-on custom contexts is added. You can define new contexts there. In defining a custom context you may face the following:

Context: select a name. It is used in defining an extension and determining contexts.

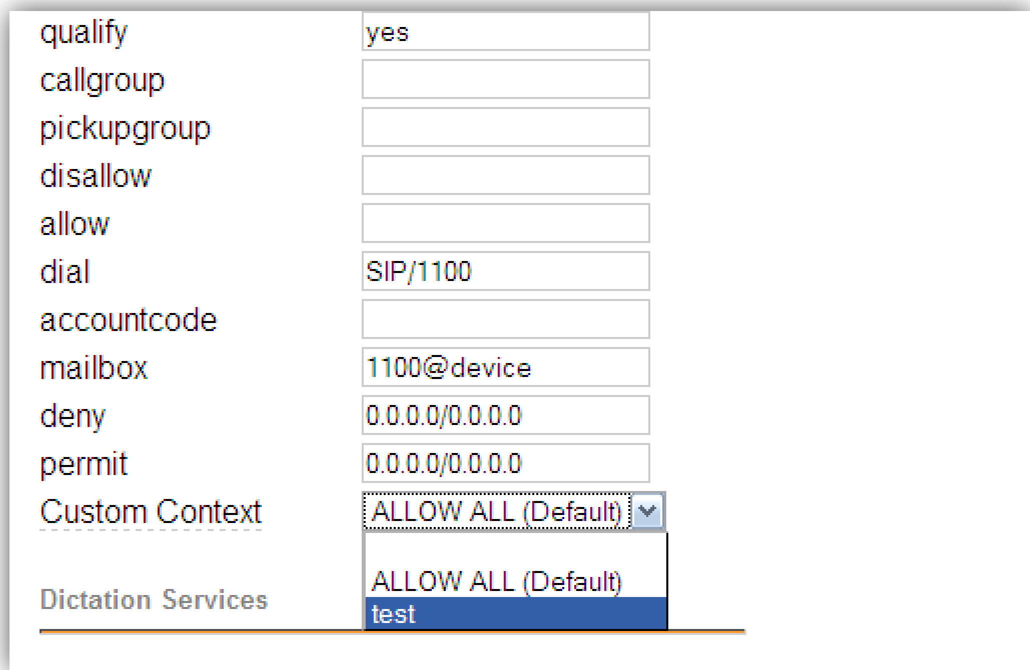
Description: an explanation for contexts.

Dial Rules: if it defines you can run the rules for dial plan. You can use dial patterns, for example if you put 2xx, it means the entire dialed plan are 3-digit and start with number 2.

You can see a list of telephony system application that by Allowing them you let extensions with this contexts use them.



After defining custom contexts go to extensions. You will see an extension or changing in defined extension, custom context is added that you can determine this contexts or this extension. In fact you will put an extension in a specific group that you defined.


qualify
yes
callgroup
pickupgroup
disallow
allow
dial
SIP/1100
accountcode
mailbox
1100@device
deny
0.0.0.0/0.0.0.0
permit
0.0.0.0/0.0.0.0
Custom Context
ALLOW ALL (Default)
ALLOW ALL (Default)
test
Dictation Services

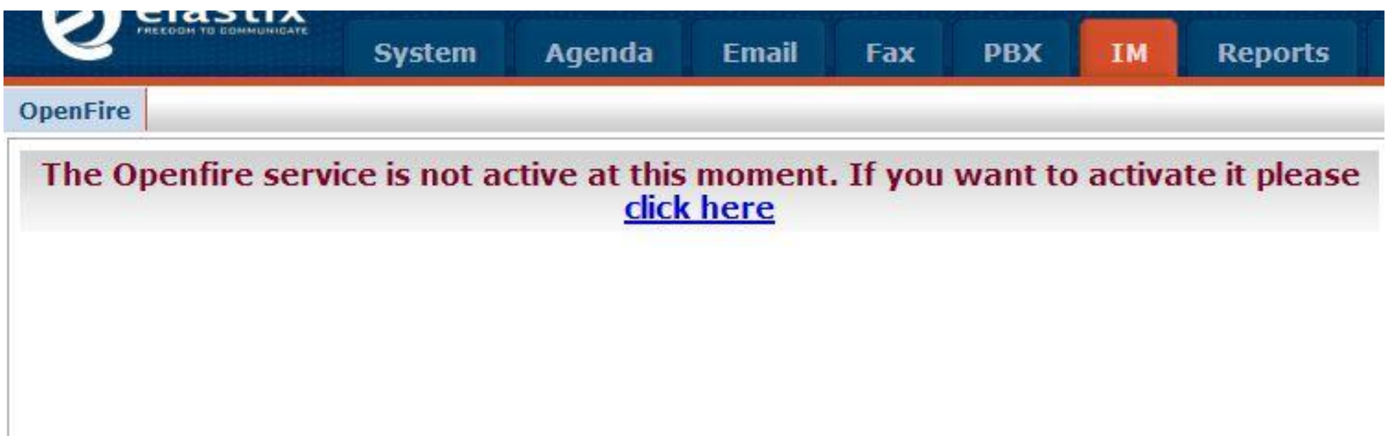
9. Instant messaging with OpenFire

It is a popular chat program and use Jabber/XMPP protocol for exchanging data. After installing this program you can have services such as Google talk, yahoo messenger and etc. name of client program installed in staff computers is SPARK which they will have these features by the configuration you did:

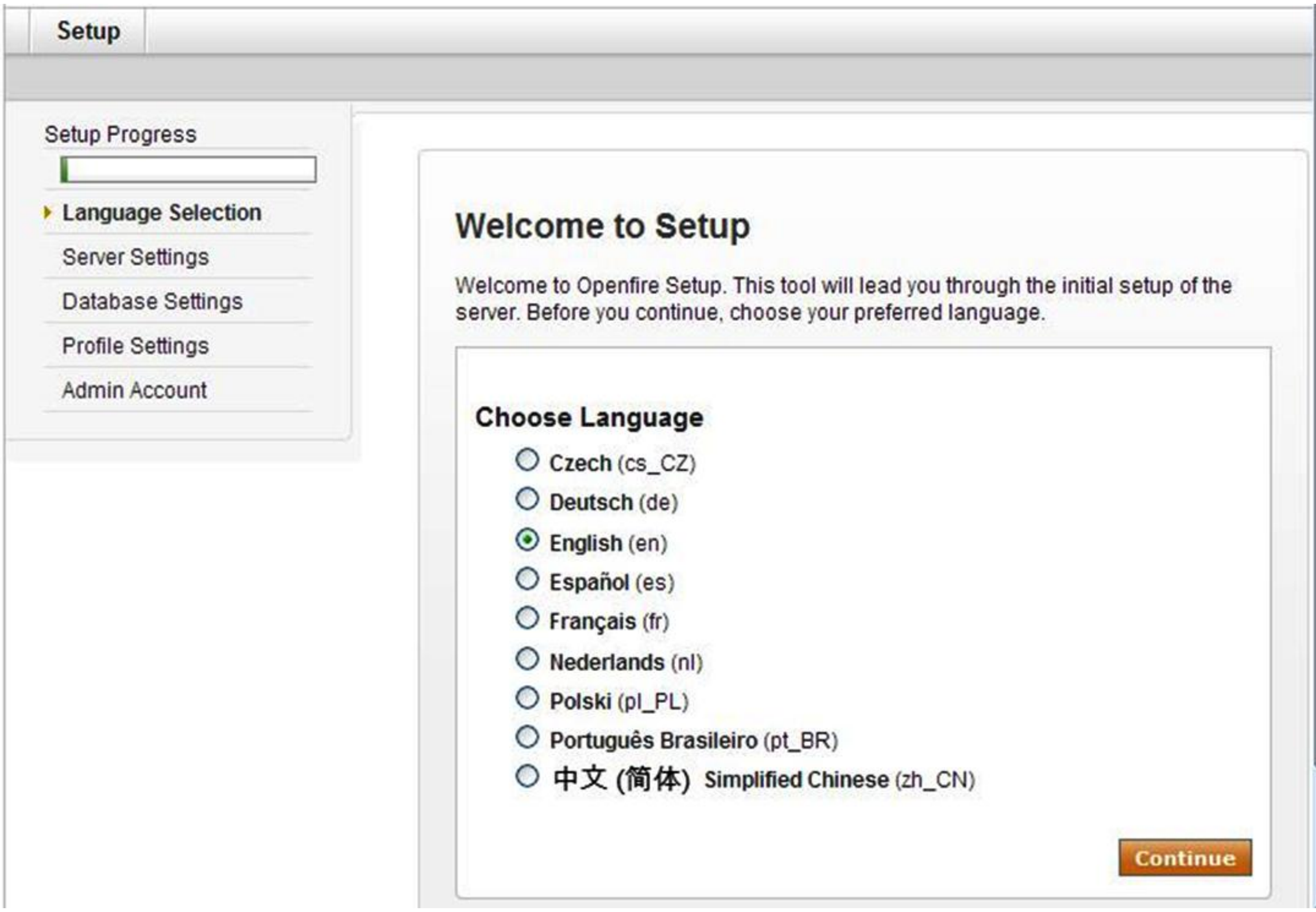
- Chat
- Exchanging the file
- Calling an extension by pressing a key
- By installing add-ons you can add your friends with Yahoo ID, MSN ID and … in fact by having a proper configuration and internet you can add your friends in yahoo messenger and chat with them.

9.1 Installing Openfire

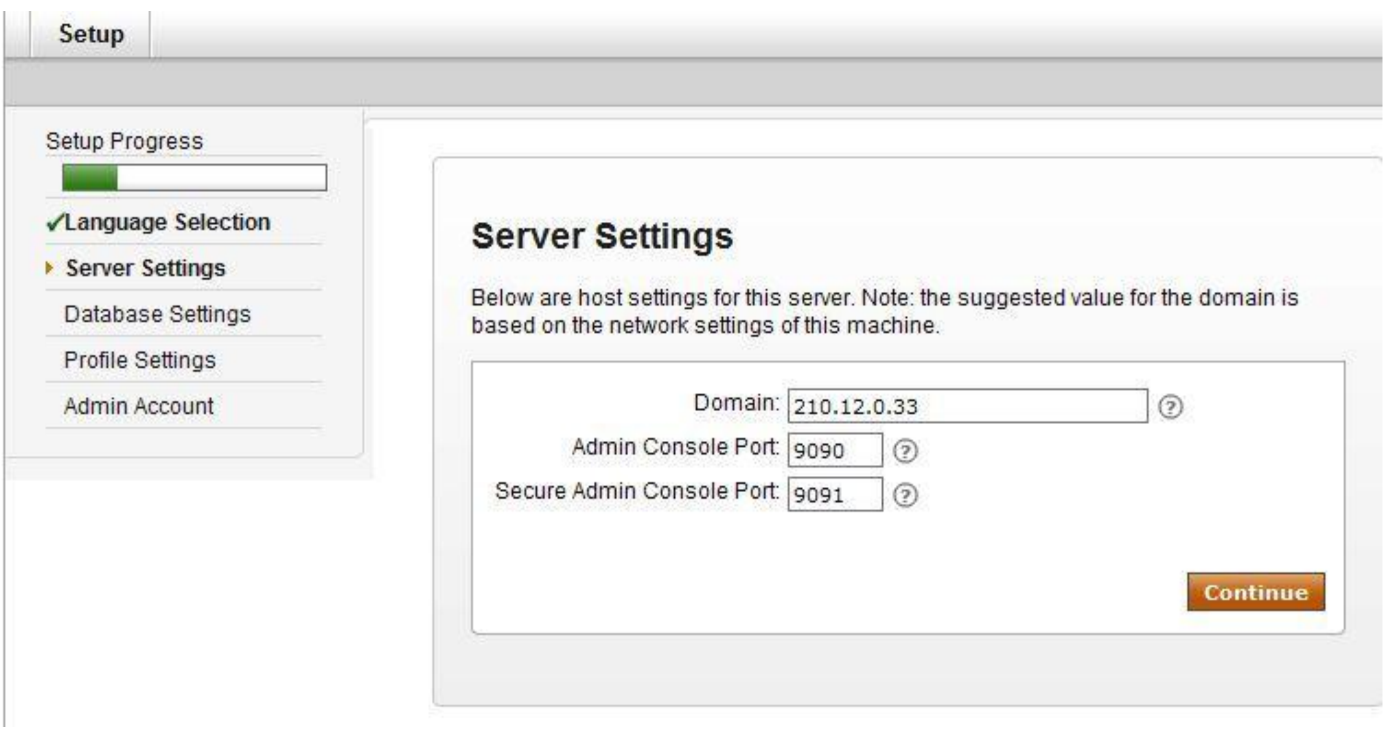
After clicking the instant messaging tab (openfire) you will see this message, because it is not installed on Elastix as a default.



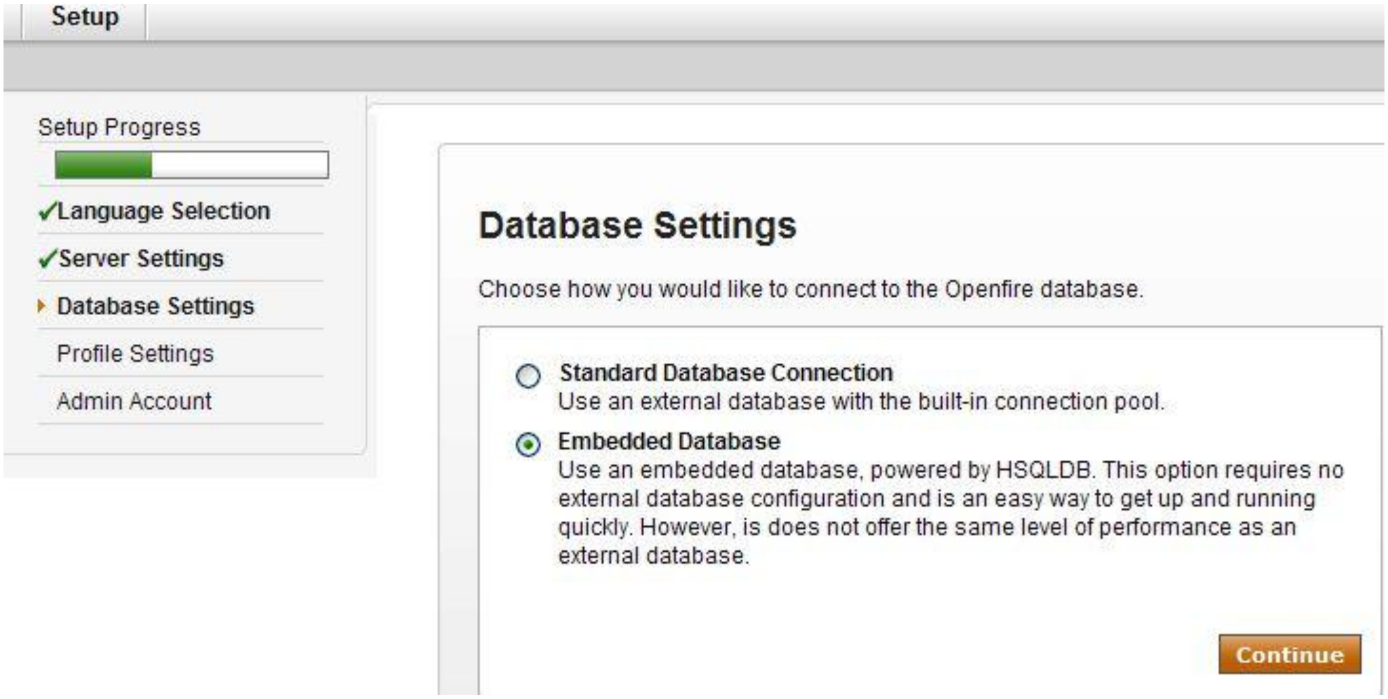
"Click Here" and installation progress will start.



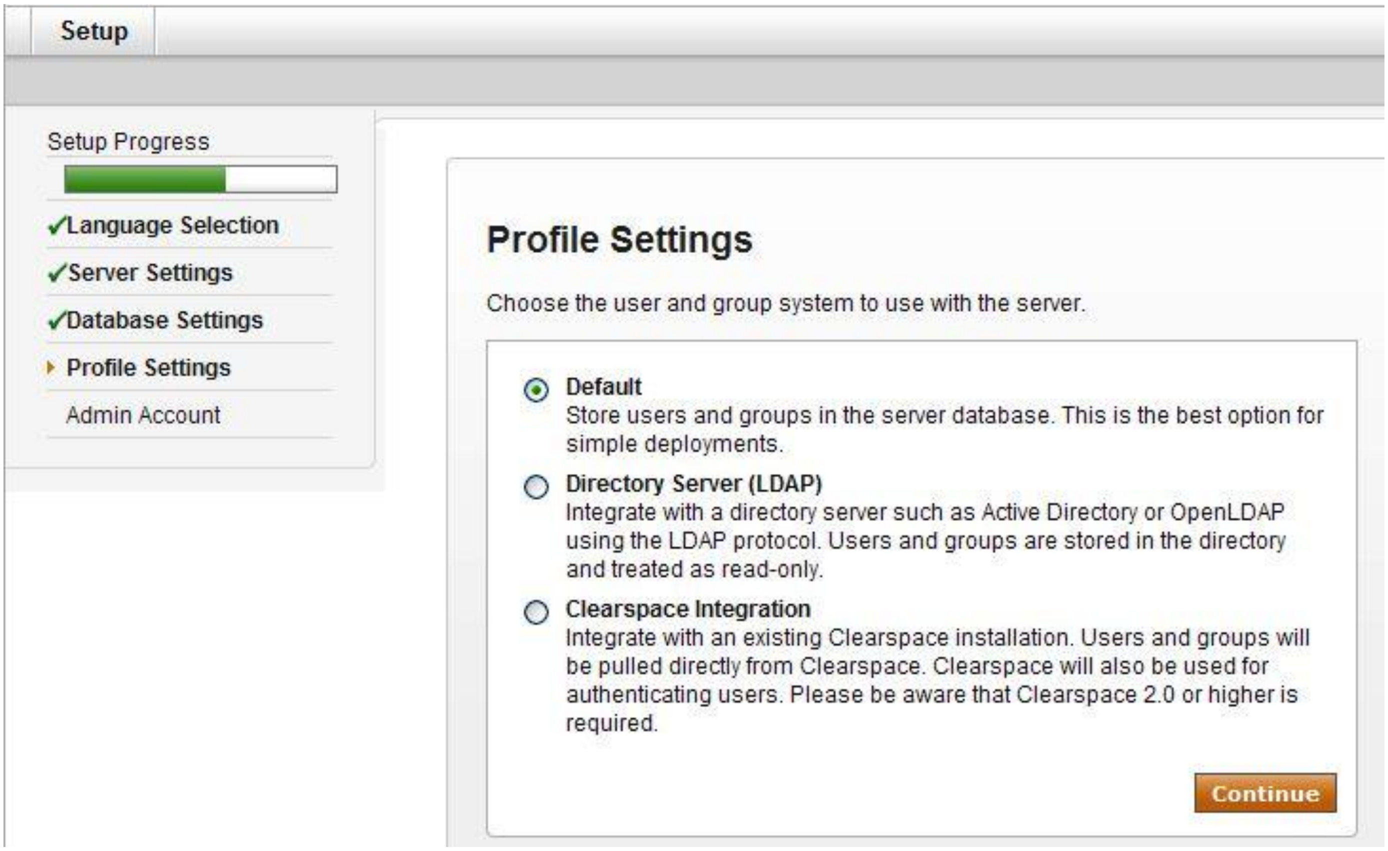
In first stage you should select the language.



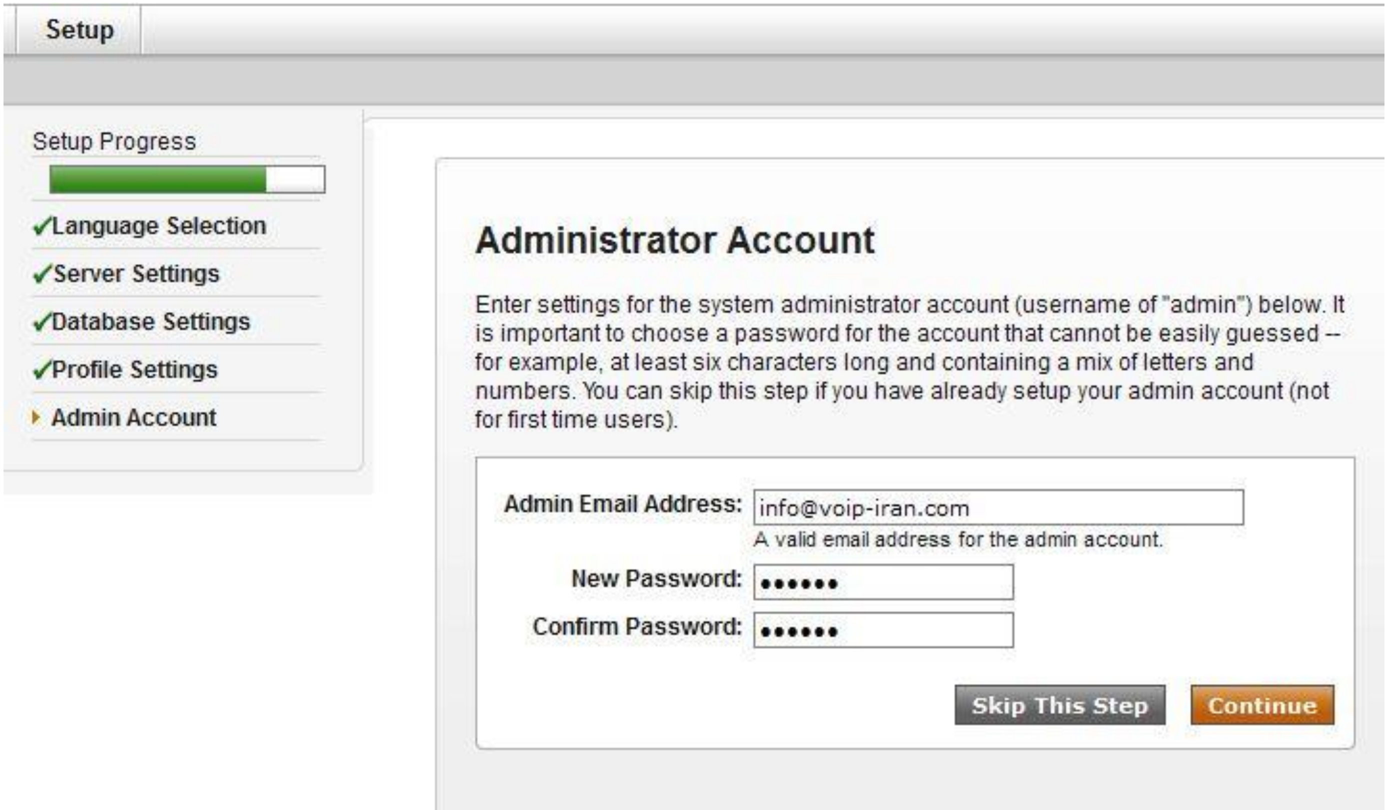
In this section you should fill the domain part with the name of Host or your IP server which is recommended don't change the name system recognized! You can change the console ports if you want but it is better to use default ones.



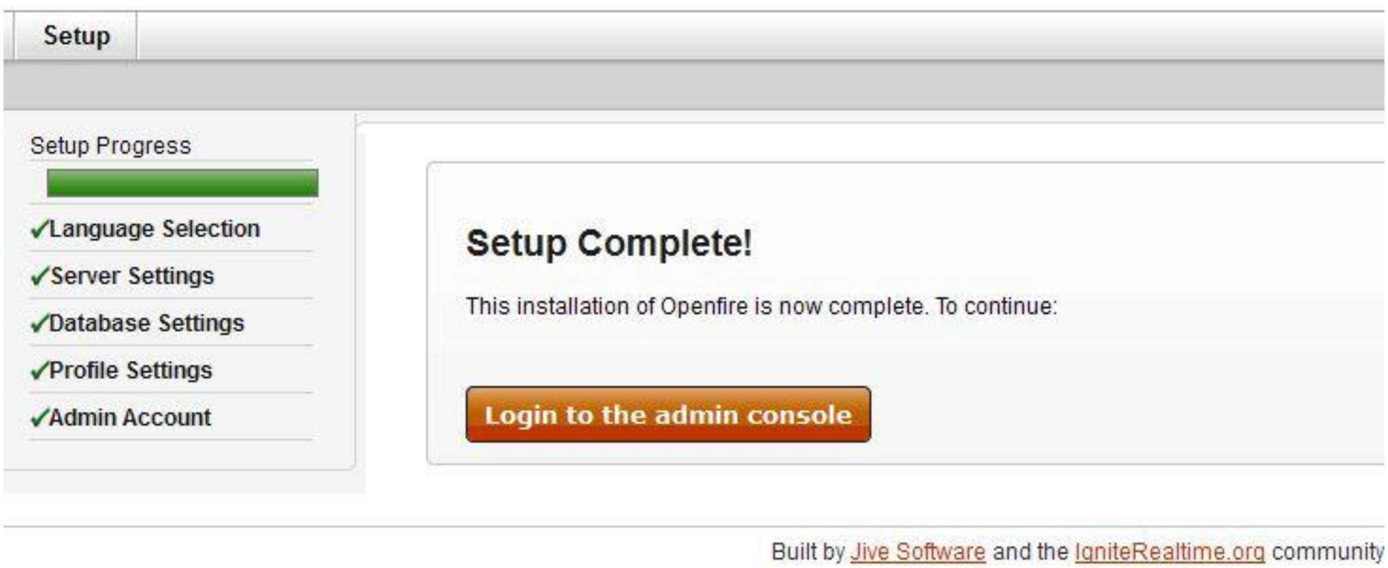
In next stage you should select how you want to be connected to database. The first one asks you a lot of question about connecting to the database which waste your time!! Select the second one and continue.



In this stage it is asked that where do you want to store users item 2 and 3 is used when you want to store them in a directory server or clearspace otherwise select Default.



Determining an email for admin user and select a password for admin user (this password has nothing to do with your email), try to choose a password you can remember!! Because its retrieving is very difficult.



Congratulation!! Your openfire installed now by clicking on "Login to the admin console" go to the page of management.

Enter with admin user and password you have selected in previous stage.

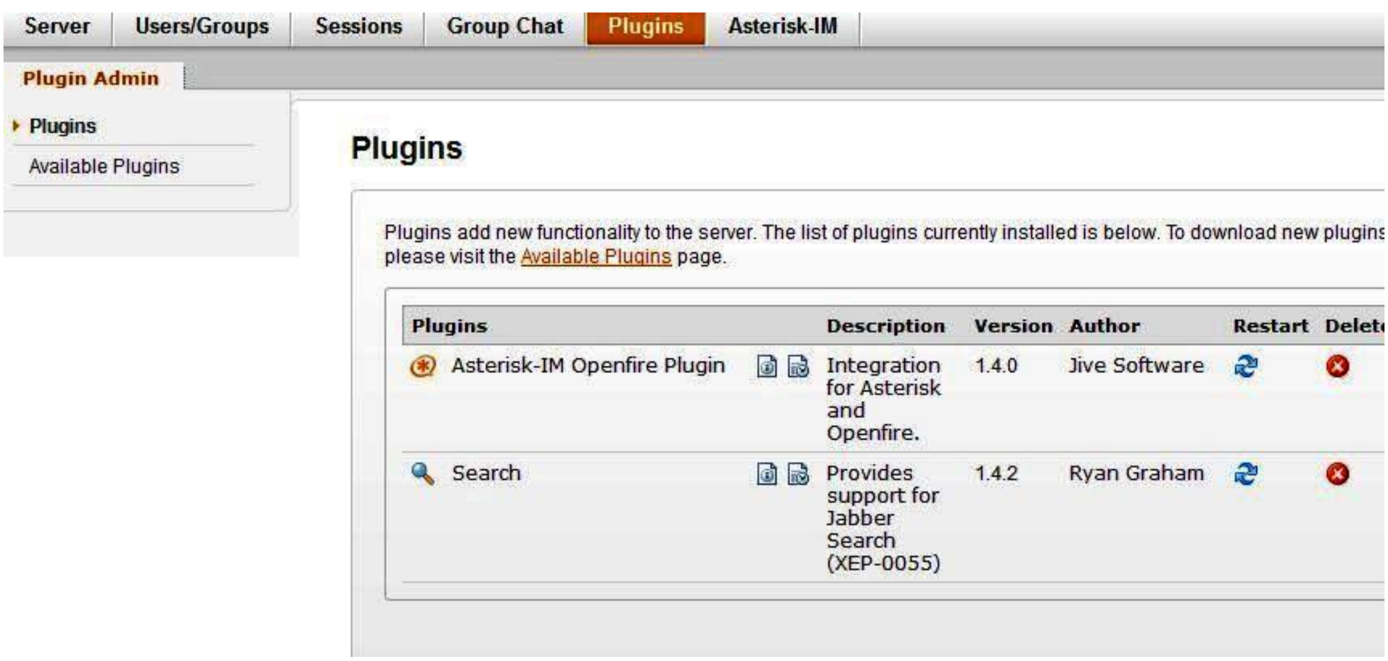
Don't be sad! You are not supposed to change all the setting! And never try to update an openfire, this program will be update with any new versions of Elastix. Updating manually may cause many problems so don't risk.

Now we go to plugins to install some add-ons (in this stage you need internet, if you don't access, you need to download the add-ons and upload here).

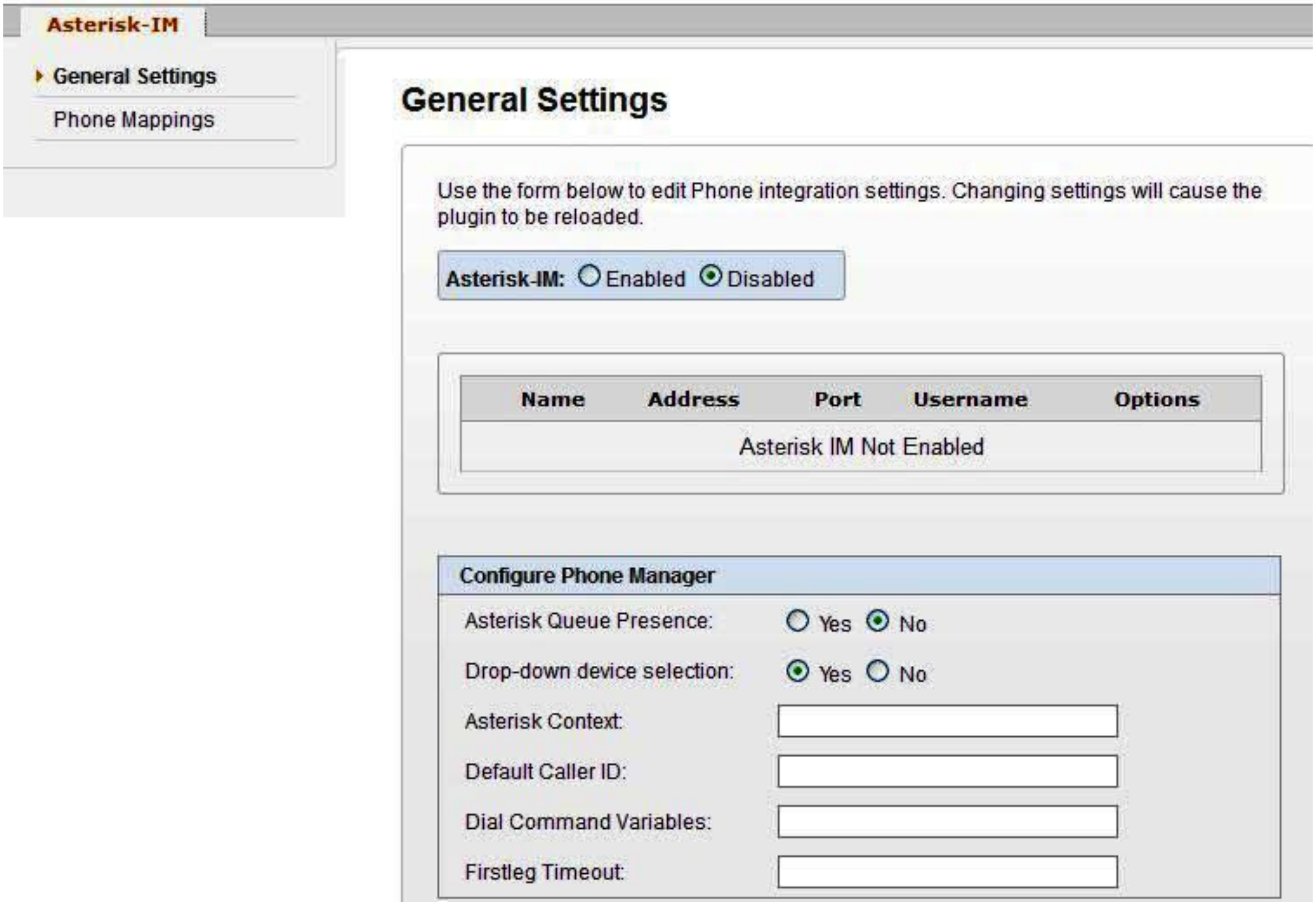
After clicking on available plugins, lists of add-ones will appear. For installing the add-ons (Asterisk-IM openfire plugins) click on green sign (+), after installation, these add-ons will be added to the list of plugins.

Install these add-ons either:

SIP Phone Plugin, Presence Service, and IM gateway



By installing new plugin, Asterisk-IM will be added to the menu. Enter to in to change some part of it.



Enable Asterisk-IM and in change the Asterisk queue presence and drop-down device selection to "Yes" and save it.

Asterisk-IM has a bug that should be fixed before using! We should have small change in file. For convenience you can use Winscp for editing the files and console environment or nano program. But for changing, open the file below:

```
/opt/openfire/plugins/Asterisk-im/database/Asterisk-im_hsqldb.sql
```

You will have something as follows:

```
create table phoneServer (
    serverID bigint not null,
    serverName varchar(255) not null unique,
    hostname varchar(255) not null,
    port integer not null,
    username varchar(255) not null,
    password varchar(255) not null,
    constraint phoneServer_pk primary key(serverID)
);

create table phoneDevice (
   deviceID bigint not null,
   device varchar(255) not null,
   extension varchar(255) not null,
   callerID varchar(255),
   isPrimary integer not null,
   userID integer,
   serverID bigint not null,
   constraint phoneDevice_pk primary key (deviceID)
);
create table phoneUser (
   userID bigint not null,
   username varchar(255) not null,
   constraint phoneUser_pk primary key (userID)
);
create unique index phoneUser_username_idx on phoneUser(username);

INSERT INTO jiveVersion (name, version) VALUES ('asterisk-im', 2);
```

Omit the word unique in third line.

After changing the file should be like that:

```
create table phoneServer (
    serverID bigint not null,
    serverName varchar(255) not null,
    hostname varchar(255) not null,
    port integer not null,
    username varchar(255) not null,
    password varchar(255) not null,
    constraint phoneServer_pk primary key(serverID)
);

create table phoneDevice (
   deviceID bigint not null,
   device varchar(255) not null,
   extension varchar(255) not null,
   callerID varchar(255),
   isPrimary integer not null,
   userID integer,
   serverID bigint not null,
   constraint phoneDevice_pk primary key (deviceID)
);
create table phoneUser (
   userID bigint not null,
   username varchar(255) not null,
   constraint phoneUser_pk primary key (userID)
);
create unique index phoneUser_username_idx on phoneUser(username);

INSERT INTO jiveVersion (name, version) VALUES ('asterisk-im', 2);
```

The bug is fixed! Now reboot Elastix to continue.

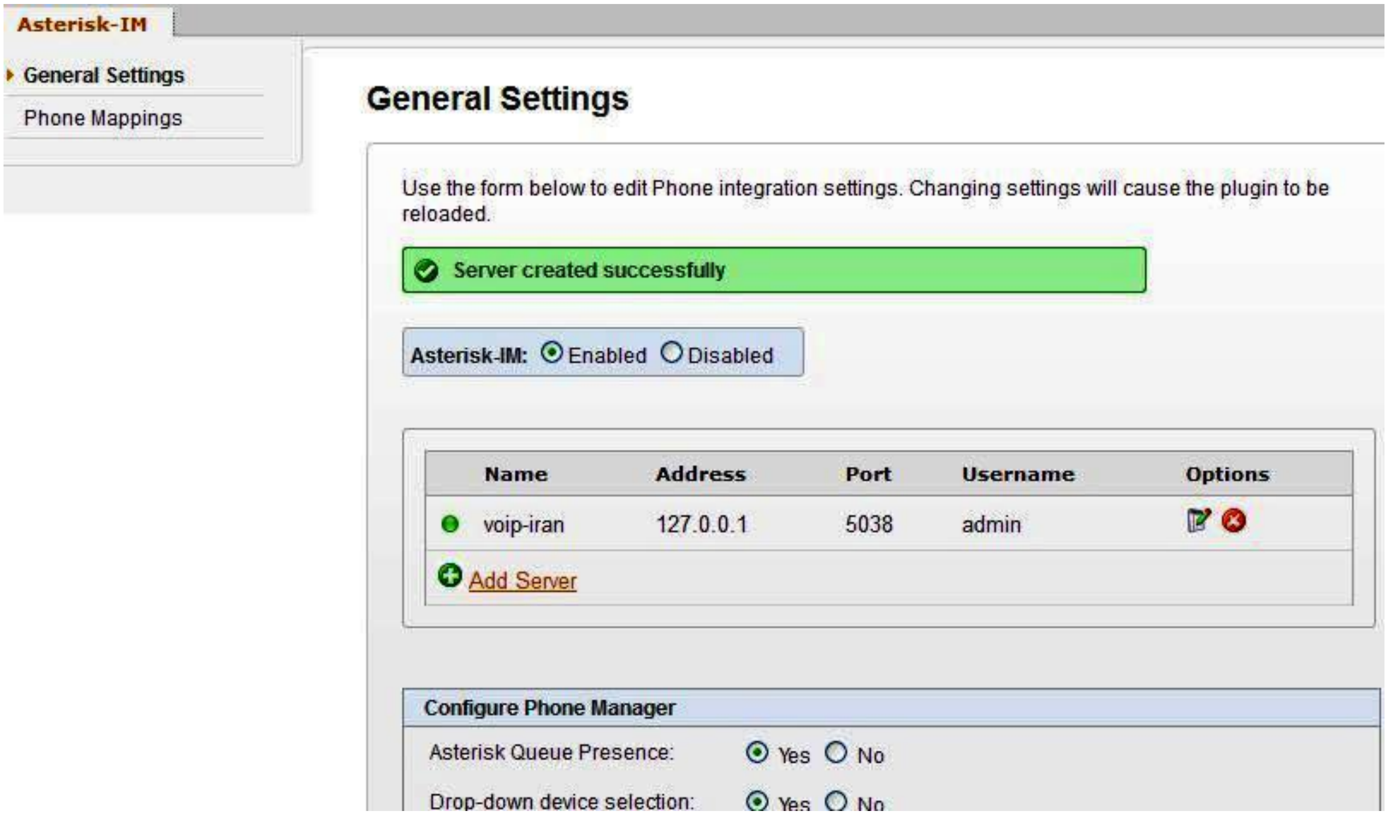
After rebooting, go to the Asterisk menu, general settings and click on Add Server and make the necessary adjustments.

- **Server name:** name of your server which is the Host name. in this example it is voip-iran.
- **Server address:** IP number 127.0.01
- **Port:** 5038
- **Username:** admin
- **Password:** as a default is “Elastix 456”

Configuration should be as follow:

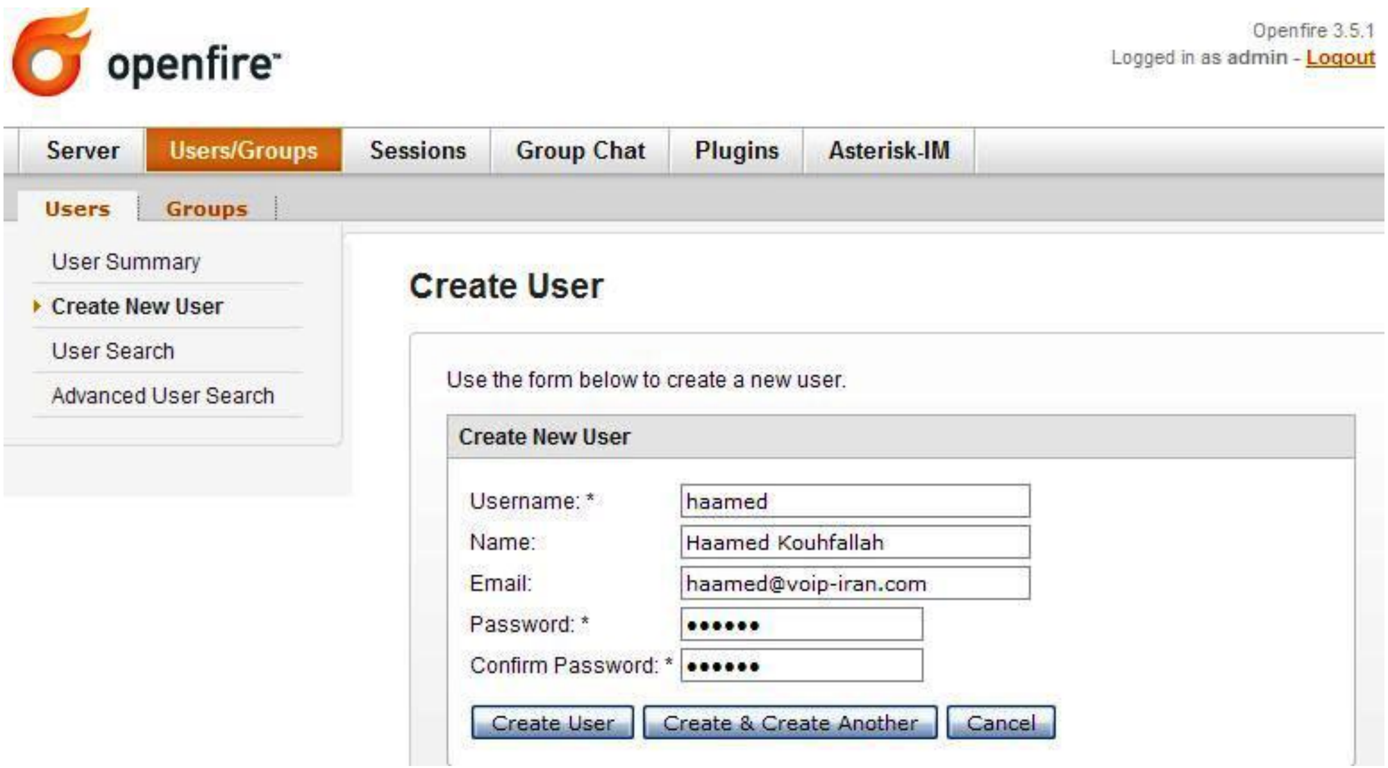
After clicking on Create Server, the server should be made and shown with a green ball. If there was nothing after creating server, be sure that you made a mistake in fixing bugs and rebooting the system. If it was created but it was shown with a gray ball, so probably registering on Asterisk is the problem. Once more check your server settings, especially the password. To ensure that the password, Elastix456, is correct you can check the correctness in the following file:

```
/etc/Asterisk/manager.conf
```

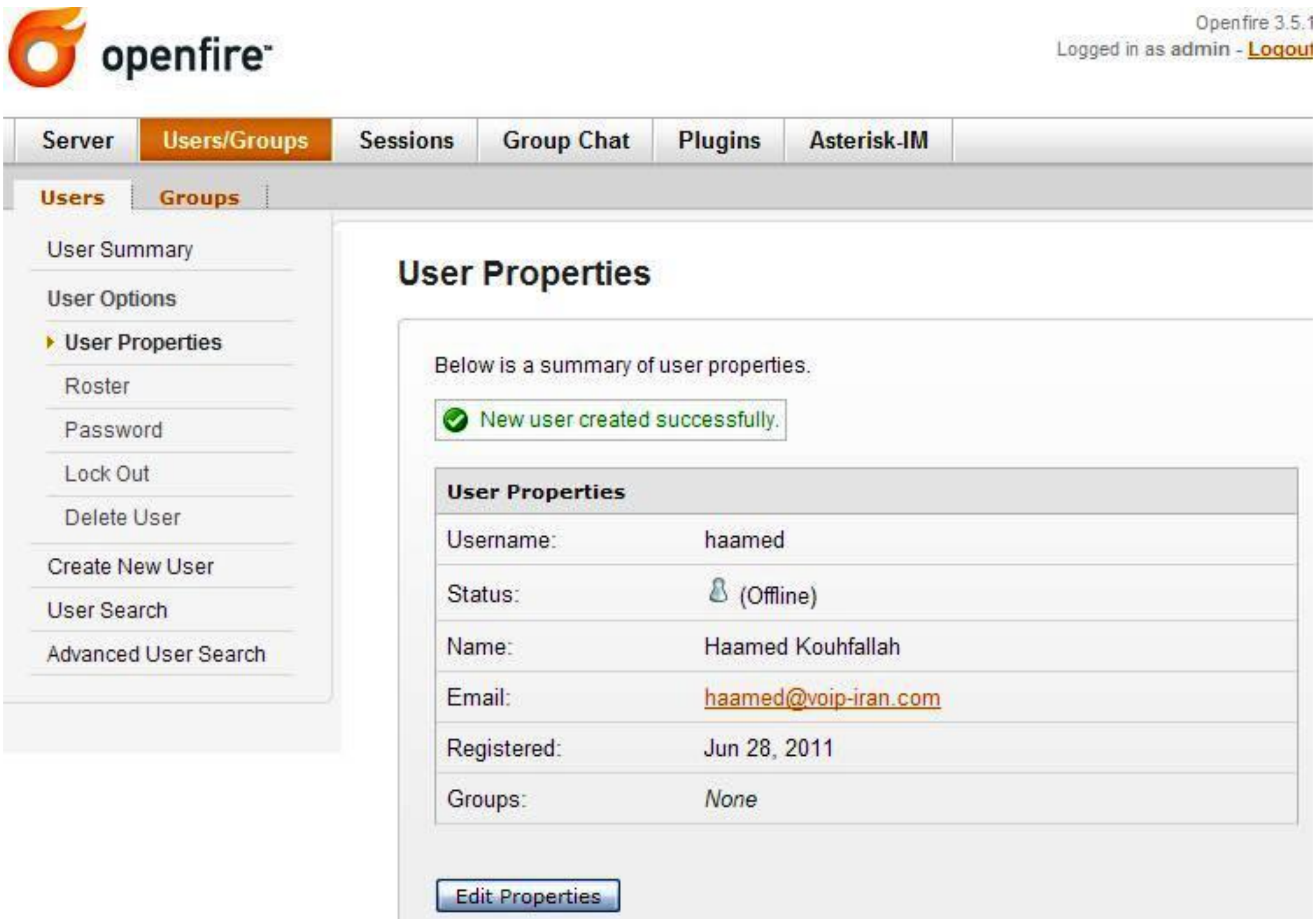


Now we are going to define a new user, the user that should be defined on SPARK and registered on the server. Go to users/groups menu. As you can see there is a default user which is admin.

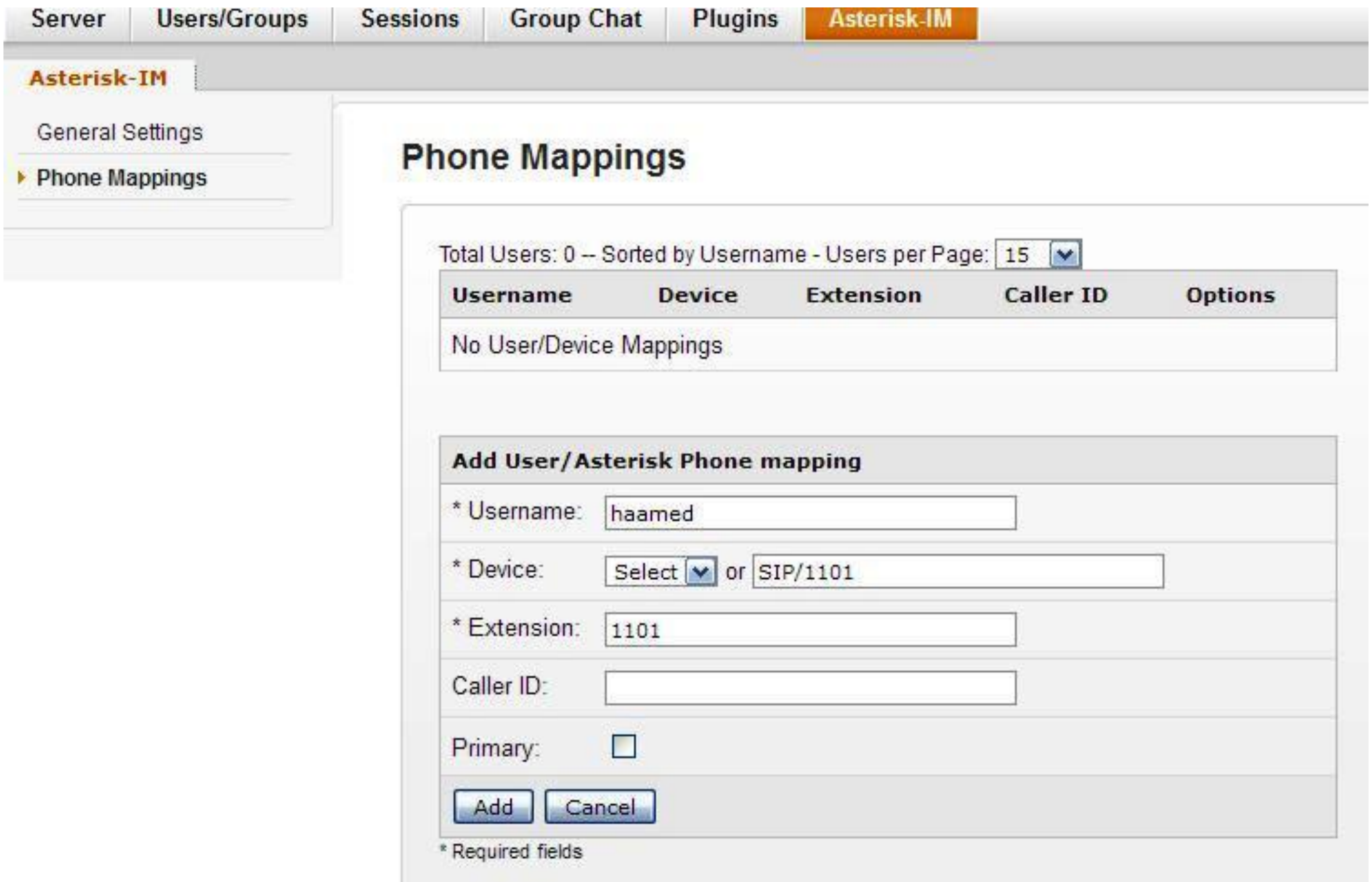
Click on Create New User.



Fill the necessary fields. For convenience it is better to enter your username with small words. You will see following page after filling and clicking on Create User.



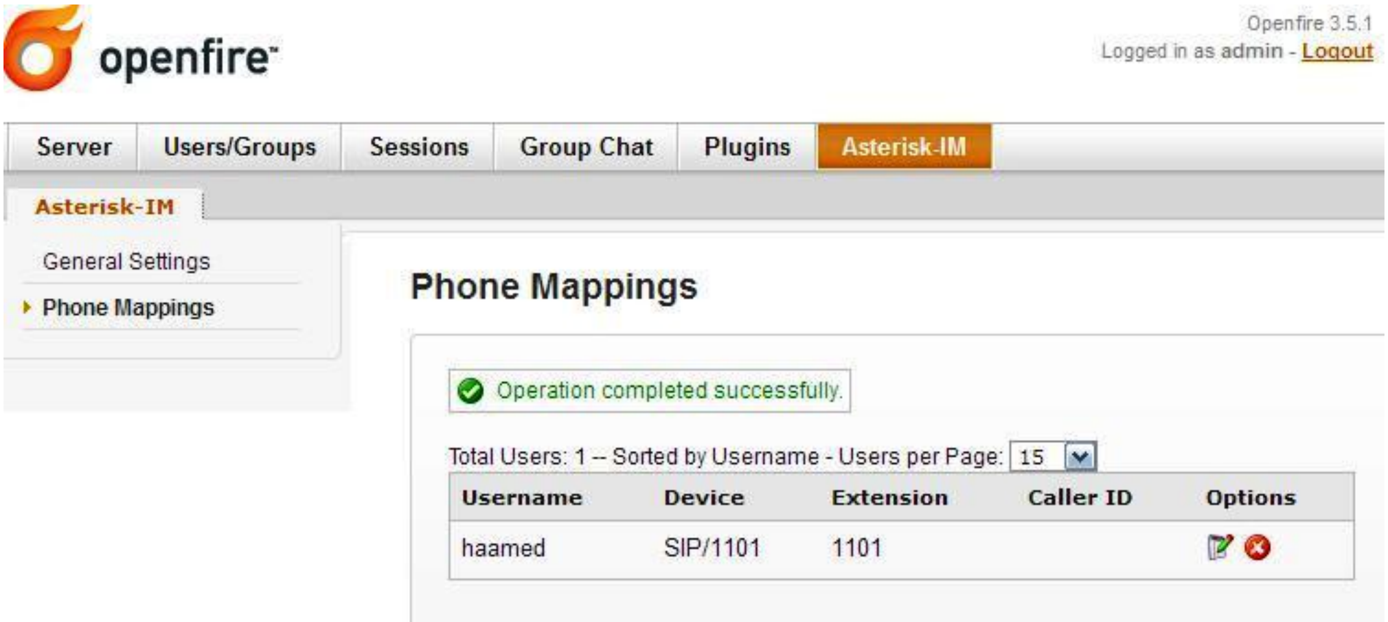
We have a user but if we want to connect our user with one of the extensions of telephony system, we should make a map (in fact we want to consider an extension for this user). go to Asterisks-IM below the Phone Mappings.



Username: the user we already made.

- **Device:** defined extension in telephony system which SIP/ Protocol is set there. You can use from the extension of the list.
- **Extension:** Intercalate the extension number.

The other 2 items don't need any changes. Now click on Add.



Well we have done here. Now we should install a SPARK IM client and test it.

9.2 Installing SPARK

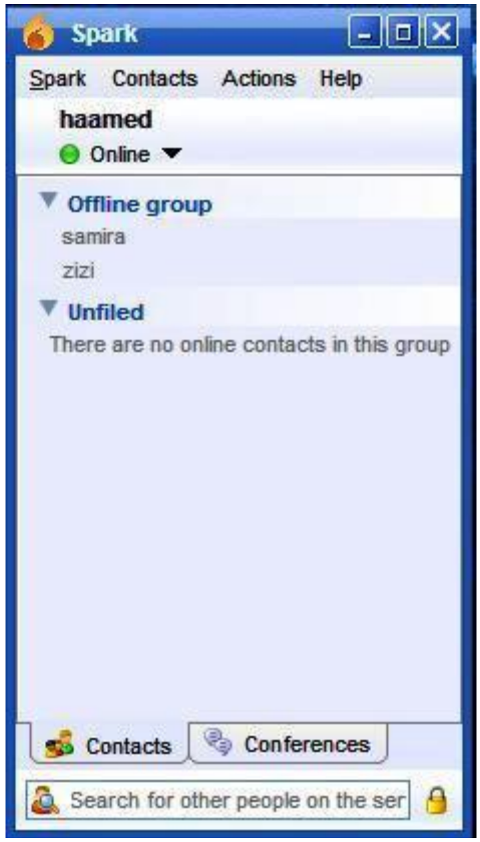
As we mentioned earlier, SPARK is an IM client that you can be registered on openfire and communicate with others exactly like Yahoo Messenger. Receive the latest version of SPARK from this link and install it.

http://www.igniterealtime.org/projects/spark/index.jsp

After installation you will see the first page.

Enter username and password, for server also enters IP of your telephony system and then login.

At first go to menu contacts and add others with their users. Of course other side should allow to be added. You can have conference with this program, exchange the file and more other features. It is a very exciting service for any company, I hope enjoy.


Spark
Spark Contacts Actions Help
haamed
Online
Offline group
samira
zizi
Unfiled
There are no online contacts in this group
Contacts
Conferences
Search for other people on the ser

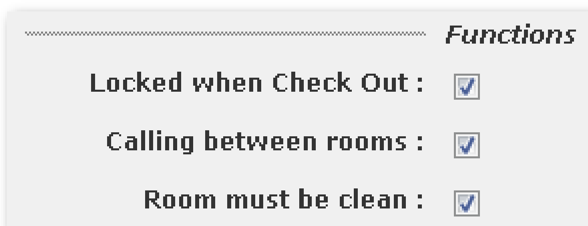
10. RoomX Module

10. 1 Configuration

- **General**

Setting operating mode.

Here, you could select 2 operating mode. (Hotel, and Hospital). Today, only one operating mode is enable.



- **Functions**

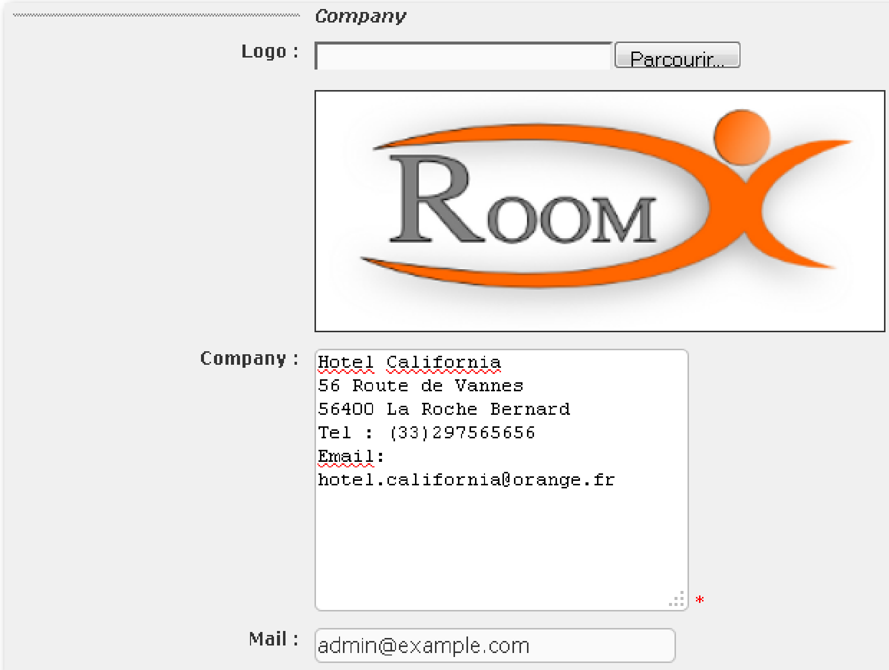
You could select 3 basic functions in RoomX application.

Locked when checkout. When the room will billed, this room will locked. So impossible to calling a number.

Calling between rooms: When checked, the room is able to call another room, but only if this room is included into the same group as the called room.

Room must be clean: The room appear into the list of available room only if the room is cleaned. Else, the room will not appear into this list. However, this room could appear if you need to make a booking about this room.

Company.



You can customize your company header, like the logo (png, or jpg file extension), the company address, and the professional mail of company.

- **RoomX Dial Plan**

You can customize or change the prefix of each RoomX function. 3 Prefix exist right now.

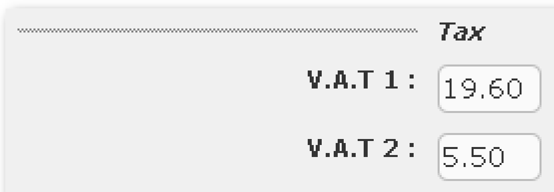
Mini-bar, is able to add some drinks on the room, and will used during the billing. When the chambermaid will cleaned the room, she could check the mini-bar and enter all drink used by the guest.

Eg: ***37**122*

(**1** could be a Coca, and **2** could be a Whiskey). So 1 Coca and 2 Whiskey, total 3 drinks.

Room Clean Prefix will used when the room will cleaned by the chambermaid.

Reception is here to giving a phone number to the reception..



- **Tax**

Two tax values can be entered. The first value is used by the outbound calls during the billing.

10. 2 Models

- **Model List**

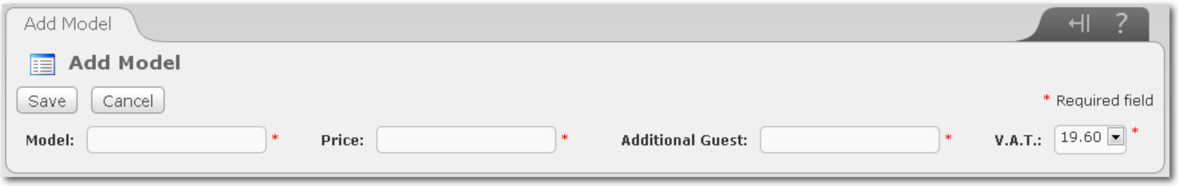
First Previous (1 - 2 of 2) Next Last

Delete	Models	Prices	Additional Guest	V.A.T
☐	Simple	60.00	0.00	19.60 %
☐	Double	100.00	10.00	19.60 %

First Previous (1 - 2 of 2) Next Last

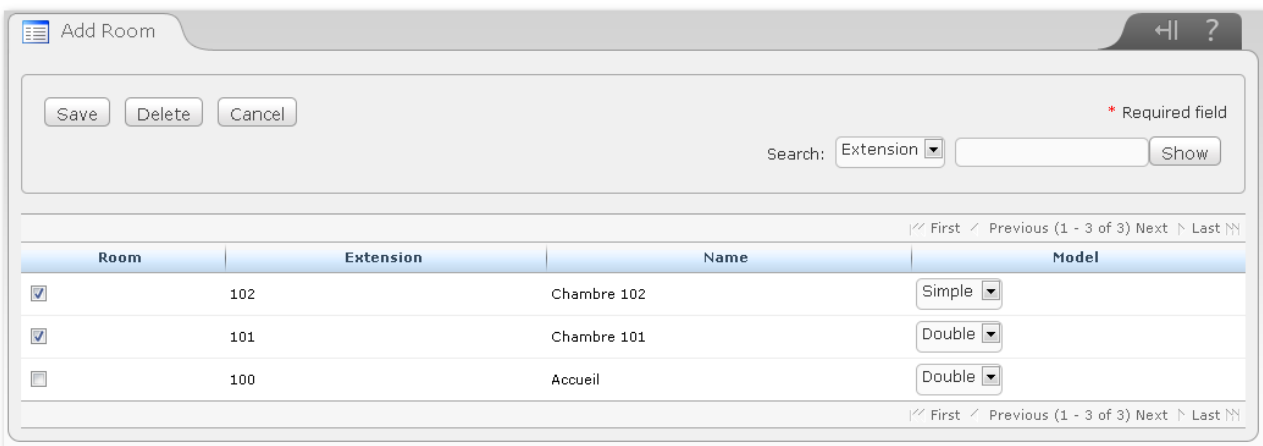
Models display all models already recorded into RoomX configuration. You could delete a room or more just selecting the checkbox at left of row.

- **Add Model**



Before to add any room, you must create some models to putting them on each room.

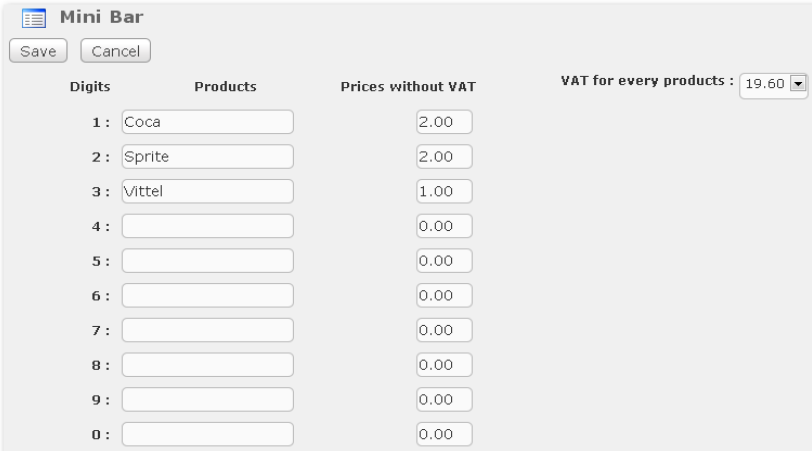
Just putting a model with its price, and why not, enter a price to additional guest if you want, and select the V.A.T used by this room. (2 V.A.T. are enabled).



- **Add Rooms**

With this module, you can select the rooms that you want from Freepbx, and into each line, select the type model for room. Don't forget, try to prepare a good list of name for each room. (eg: room 100, room 101..etc). This name will be used by RoomX if no name is entered.

10. 3 Mini-Bar

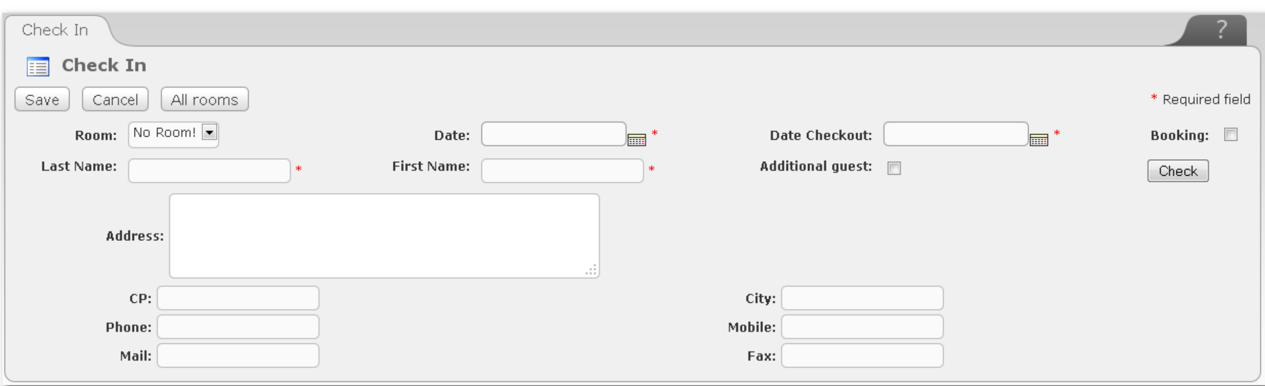


This menu affecting a product on each key with its price. You can enter 10 different products on this module.

2 V.A.T can be selected.

When the chambermaid check a mini-bar, she dials the prefix (*37 by default) and presses the number of product used. The entering is ended by the * key. If the # key is pressed, all products will not billed.

10. 4 Checking



You can making a checking with this menu. You can see some fields to enter different values.

Room, displays all available rooms into this list, and all cleaned room (*if this options is enabled or not into config parameters*).

Date, it's the Checking date (the current date by default).

Date Checkout, is needed to have a reference in the case where another room will booked. This information is not used for the billing.

Of course, you must enter a **last-name** and **first-name** to making a checking.

This first part is required and needed to making a checking. The others fields below are facultative.

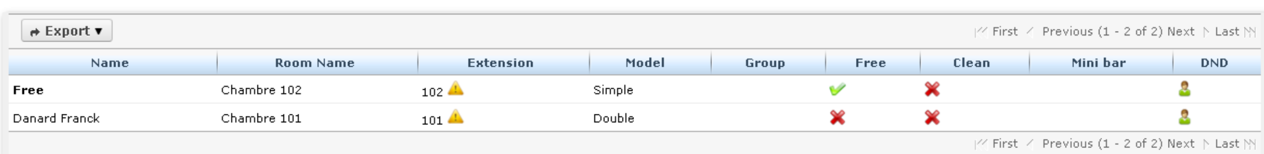
However, only one field is needed in the case where you want to sending the billing by mail. In this case, you must enter this field. (**mail**). No billing will sent by fax yet.

If you want to make a booking, you must click on Booking and why not, maybe click on

to have the full list of rooms (free and not cleaned).

When no room are into the list, check if all rooms are cleaned and available. If it's done, then your hotel is full.

10. 5 Room List

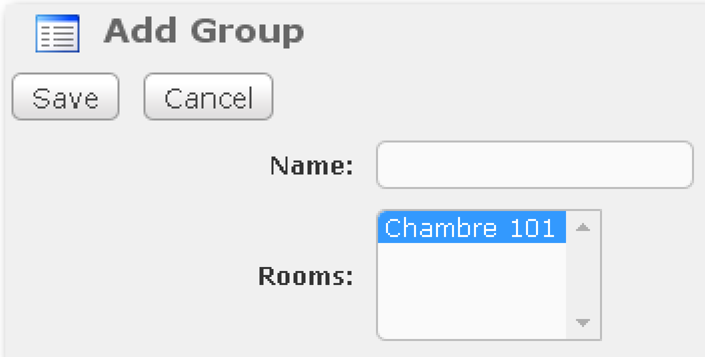


Name	Room Name	Extension	Model	Group	Free	Clean	Mini bar	DND
Free	Chambre 102	102	Simple					
Danard Franck	Chambre 101	101	Double					

You can see the room's status currently in your hotel. The guest name, the room name, if it's free or busy, cleaned or not. If the guest used the mini-bar or not. If the room is on DND status or not (Do Not Disturb). And you can see if the room is included in a group or not.

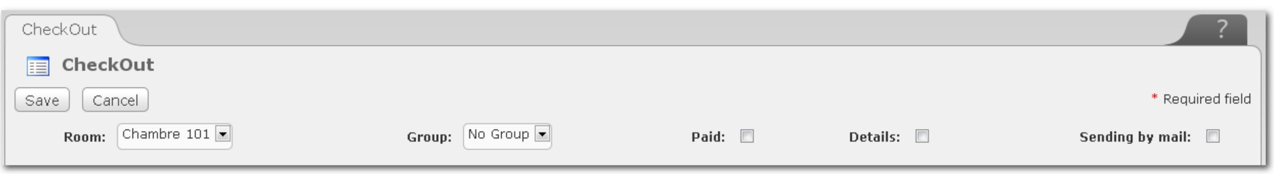
If the phone device is a SIP phone, you can know if the phone is connected or not. In this case, you have a small yellow triangle beside the phone number.

10. 6 Group List.



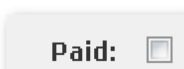
Here, you can see all group already existing, and you can add lots of rooms into a group in the same time. Just selecting several rooms maintaining the [shift] key and click on the rooms that you want.

10. 7 Checkout



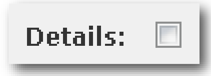
You can do 2 types of checkout. A classic checkout by room, and a checkout by group.

Checkout by group will take all room in group, and will make the



checkout, room by room.

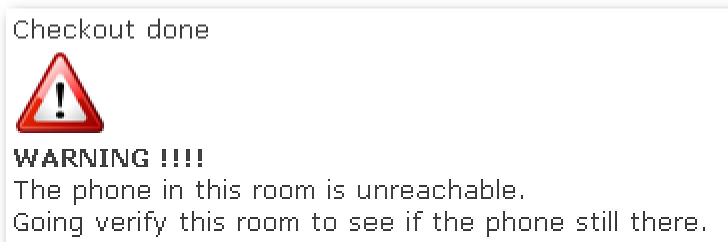
If paid is checked, the billing is paid by the guest, else, this billing is taken like not paid.



If you want to have all calls details for the room, check this box.

If you want that guest receive its billing by mail, check this box.

When you click on save button, these information will appear:

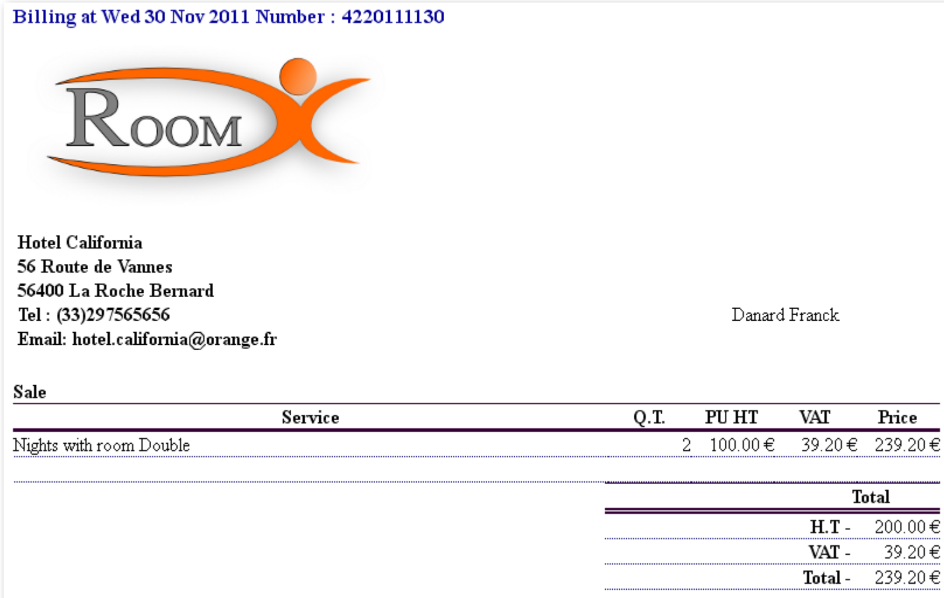


If the phone is unconnected, you could see this information:

Else, you will have only:

Next, you will there information below:

Click on display to see the billing, and you could print this billing by the way.



Billing at Wed 30 Nov 2011 Number : 4220111130

Hotel California
56 Route de Vannes
56400 La Roche Bernard
Tel : (33)297565656
Email: hotel.california@orange.fr

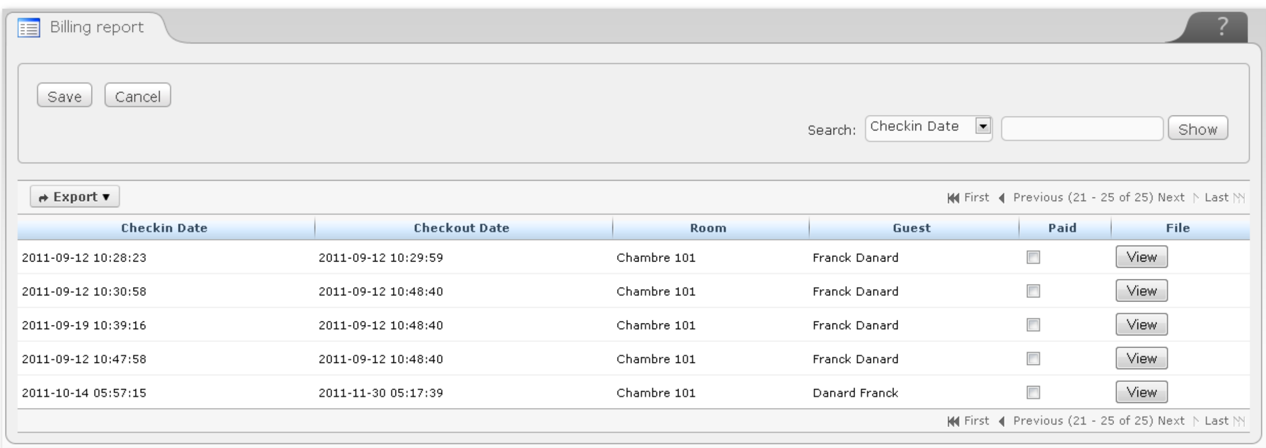
Danard Franck

Sale

Service	Q.T.	PU HT	VAT	Price
Nights with room Double	2	100.00 €	39.20 €	239.20 €

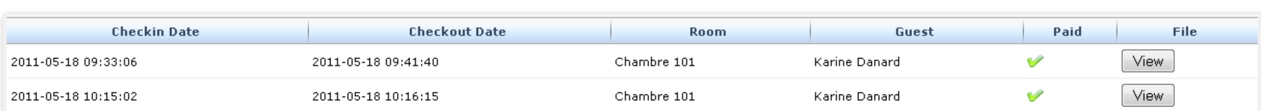
Total	
H.T -	200.00 €
VAT -	39.20 €
Total -	239.20 €

10. 8 Billing Report



Checkin Date	Checkout Date	Room	Guest	Paid	File
2011-09-12 10:28:23	2011-09-12 10:29:59	Chambre 101	Franck Danard		View
2011-09-12 10:30:58	2011-09-12 10:48:40	Chambre 101	Franck Danard		View
2011-09-19 10:39:16	2011-09-12 10:48:40	Chambre 101	Franck Danard		View
2011-09-12 10:47:58	2011-09-12 10:48:40	Chambre 101	Franck Danard		View
2011-10-14 05:57:15	2011-11-30 05:17:39	Chambre 101	Danard Franck		View

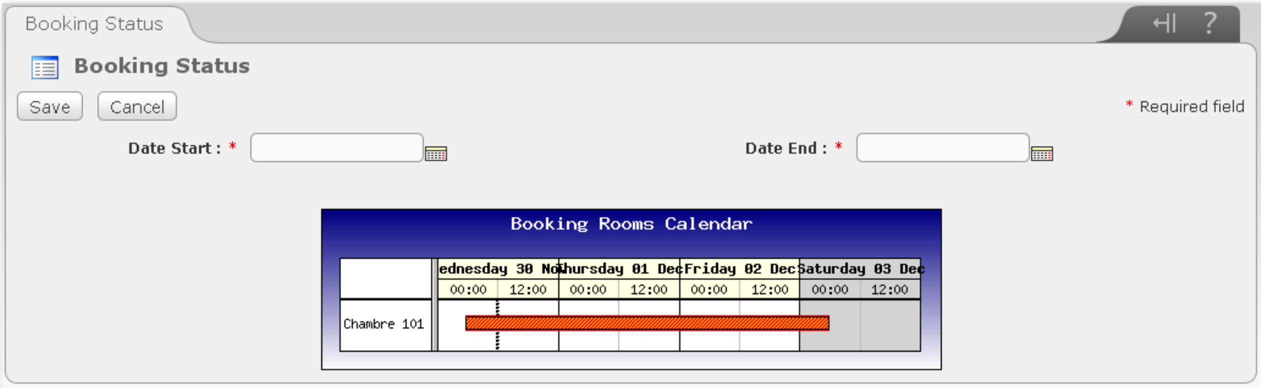
When a guest will paid its billing, then you could check the paid box.



Checkin Date	Checkout Date	Room	Guest	Paid	File
2011-05-18 09:33:06	2011-05-18 09:41:40	Chambre 101	Karine Danard	✔	View
2011-05-18 10:15:02	2011-05-18 10:16:15	Chambre 101	Karine Danard	✔	View

10. 9 Booking

- **Booking Status**

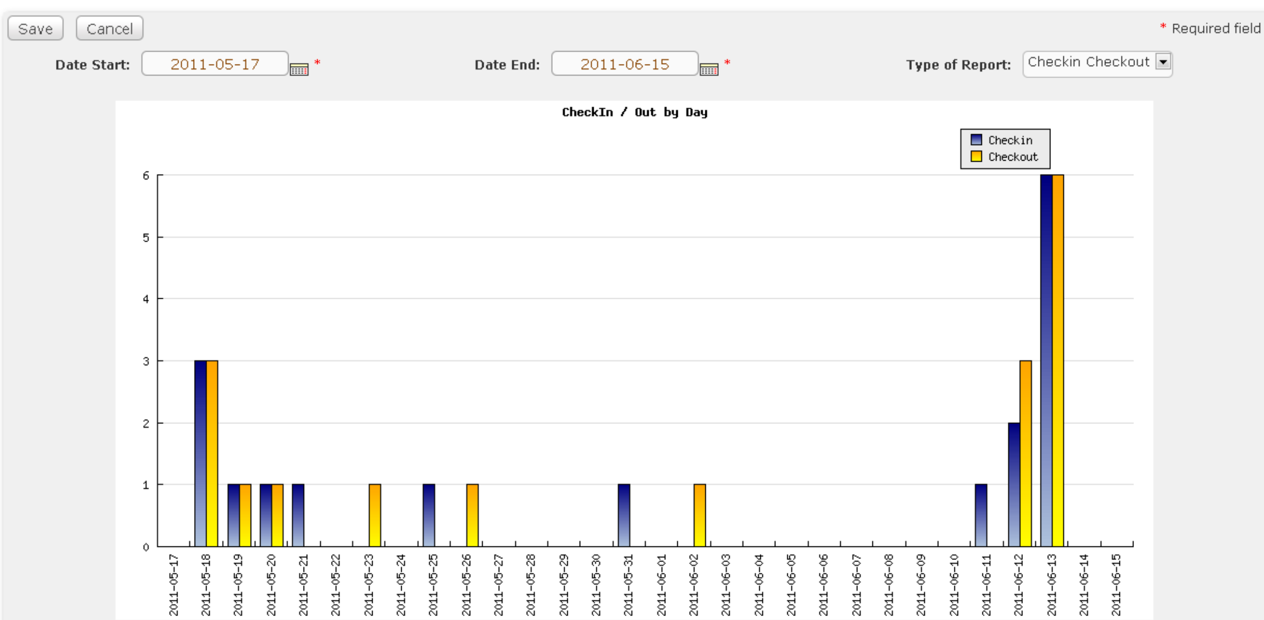


Here, you have all booking which currently entered into RoomX. You can do a view between 2 dates.

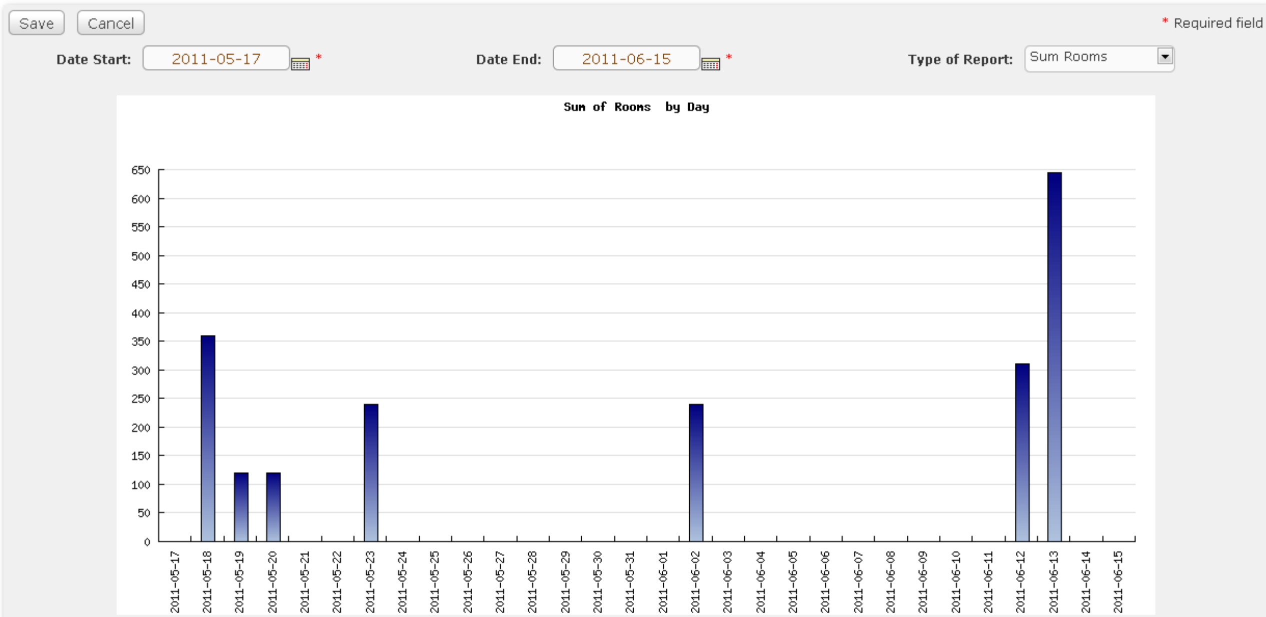
- **Booking List**

To make a checking on a booked room, check the checking box, and if you want to cancel a booking, check the canceled box.

- **Company report**



You can realize some company report, like how many checking and checkout by day between two dates.



Sum Rooms, mini-bar, calls, and billings.

(*The details will coming soon with the next versions*)

10. 10 Home

On the Home page, you can see some information directly on this page.

How many available rooms (*rooms free*), or not (*rooms busy*), if you have some booking today, the total rooms, if your Hotel is full or potentially full (*caused by the booking*).

If there's a booking today, just click on the button to going to booking menu directly.

11. Asterisk AGI Programming

11.1 What is AGI?

The Asterisk Gateway Interface, or AGI, provides a standard interface by which external programs may control the Asterisk dialplan. Usually, AGI scripts are used to do advanced logic, communicate with relational databases (such as PostgreSQL or MySQL), and access other external resources. Turning over control of the dialplan to an external AGI script enables Asterisk to easily perform tasks that would otherwise be difficult or impossible.

Asterisk **AGI** enables an IVR developer to develop IVR structures that are sometimes, bordering on the absurd, as applications tend to become more and more complex by using AGI. However, there are some scenarios where common dialplan practices are no longer applicable, and the use of an external logic is a must.

11.2 Calling an AGI Script from the Dialplan

In order to work properly, your AGI script must be executable. To use an AGI script

inside your dialplan simply call the `AGI()` application, with the name of the AGI script.

You can do it by adding your code to extensions_custom.conf file in `/etc/asterisk`

as the argument, like this:

```
include => test


[test]

exten => 123,1,Answer()

exten => 123,2,AGI(agi-test.agi)
```

AGI scripts often reside in the AGI directory (usually located

in */var/lib/asterisk/agi-bin*), but you can specify the complete path to the AGI script.

11.3 AGI, EAGI, DeadAGI and FastAGI

In addition to the `AGI()` application, there are several other AGI applications suited to

different circumstances. While they won't be covered in this chapter, they should be

quite simple to figure out once you understand the basics of AGI scripting.

The **`EAGI()`** (enhanced AGI) application acts just like `AGI()` but allows your AGI script

to read the inbound audio stream on file descriptor number three.

The **`DeadAGI()`** application is also just like `AGI()`, but it works correctly on a channel

that is dead (i.e., a channel that has been hung up). As this implies, the regular `AGI()`

application doesn't work on dead channels.

The **`FastAGI()`** application allows the AGI script to be called across the network, so that

multiple Asterisk servers can call AGI scripts from a central location.

In this chapter, we'll first cover the sample *agi-test.agi* script that comes with Asterisk

(which was written in Perl), then write a weather report AGI program in PHP, and finish up by writing an AGI program in Python to play a math game.

* To get a list of available AGI commands, type `show agi` at the Asterisk command-line interface. You can also

refer to Appendix C for an AGI command reference.

11.4 AGI scripting frameworks

As with any other open source project, the number of frameworks built for the development of AGI scripts is amazing. Considering the fact that the AGI language consists of less than thirty different methods, the existence of over thirty different scripting frameworks is amazing.

The following list contains some of the more popular frameworks for AGI scripting:

Language	Framework	URL
PERL	Asterisk PERL Library	http://asterisk.gnuinter.net/
PHP	PHP-AGI	http://sourceforge.net/projects/phpagi/
Python	py-Asterisk	http://py-asterisk.berlios.de/py-asterisk.php
C	libagiNow	http://www.open-tk.de/libagiNow/
.NET	MONO-TONE	http://gundy.org/asterisk
JAVA	Asterisk-java	http://www. asterisk-java.org

11.5 The ten rules of AGI development

Rule #1: An AGI script should terminate as fast as possible

First-time AGI developers tend to develop their entire application within an AGI script. As you develop your entire application within an AGI script, you may gain the power of the scripting language, but will incur a cost of performance. Always make sure that the AGI scripts that you develop terminate their execution as fast as possible, returning to the dialplan as fast as possible. This concept dictates that each AGI script being run should behave quickly as an atomic unit—hence the name "Atomic AGI".

Rule #2: Blocking applications have no place in AGI

As a direct continuation to rule #1, you should never execute a blocking application from within an AGI script. Initiating a blocking application from within an AGI script will make your Asterisk environment explode slowly. Why is that? Because for every blocking application that you run from within the AGI script, you will have both your AGI script and the blocking application running for the duration of the block. Imagine that you were to initiate the Dial application from within an AGI script, and the call created would last over thirty minutes. For those thirty minutes, your AGI script is still active. This isn't much of a problem when dealing with small-scale systems. But when trying to run 50 concurrent scripts, be prepared for failure.

Blocking applications include the following: Dial, MeetMe, MusicOnHold, Playback (when dealing with long playbacks), Monitor, ChanSpy, and other applications that have an unknown execution duration.

Rule #3: Asterisk channels are stateful—use them

An Asterisk channel, once operational, is like a big bucket of information. Channel variables can be used to carry information from your AGI script to the dialplan and back. The variables remain as part of the channel until the channel is terminated, when memory is then freed.

Using this "bucket" enables you to carry variables and information obtained via an AGI script and then pass these to an application in the dialplan. For example, imagine that you are developing a pre-paid platform. A decision on the call timeout is taken via an AGI script. However, the actual dialing of the call is performed from the dialplan itself.

Rule #4: AGI scripts should manipulate data—no more

Most developers tend to think of AGI scripting as a functional unit, meaning that they enclose multiple functionalities into the AGI script. While AGI is fully capable of performing telephony functionality, it is best to leave this functionality to the dialplan.

Utilize your AGI script to set and reset channel variables and communicate with out-of-band information systems. The concept of allowing out-of-band information flow into Asterisk's operational flow of channel, enables new functionality and possibilities. Not all logic should be handled by your AGI script. Sometimes, it is better to use the AGI script as a data conduit, while letting an external information system handle the data manipulation.

Rule #5: VM based languages are bad for AGI scripting

Virtual machine based programming languages are bad for AGI scripting. Putting aside the rules #1 and #2, imagine that your application is built using an AGI script using the Java programming language. If you are familiar with Java, you most probably know that for each program that you execute using Java, a full Java virtual machine is invoked.

While this practice may seem fairly normal for information systems, Asterisk and IVR development vary. Imagine that our system is required to handle a maximum number of 120 concurrent channels, which means that the maximum number of concurrent AGI scripts will be 120. According to this concept, our Java environment will be fully invoked for each of these 120 instances. In other words, too many resources are being used just for the VM.

If you really feel that you want to develop AGI scripts using Java, FastAGI is the way to go for you.

Rule #6: Binary-compiled AGI is not always the answer

While evaluating rules #1, #2 and #5, we can't but reach an almost immediate conclusion that we need to have our AGI script binary compiled. While in terms of efficiency and performance, a binary compiled AGI may provide better performance, the time required to develop it may be longer. In some cases, scripting languages such as PHP, PERL, and Python may provide near-similar performance at a lesser cost.

Also, when using binary compiled AGI scripts, you are always in charge of the memory allocation and cleanup. Even the most experienced developer can commit errors while dealing with memory allocation, so binary compiled AGI need not be the solution always.

If your system truly requires the performance edge of a binary compiled AGI, we encourage you to develop a prototype using a scripting language. Once you have your prototype working, start developing your binary version.

Rule #7: Balance your scripts with dialplan logic

While evaluating rules #1, #2 and #4, we must keep in mind that developing IVR systems with Asterisk resembles a high-wire balancing act. The fine art of balancing your dialplan with AGI scripts proves to be a powerful tool, especially when developing complex IVR systems.

Many developers tend to externalize functionality from the dialplan into AGI, while the same functionality can be achieved by writing dialplan macros or dialplan contexts. Using Asterisk's branching commands (*goto*, *gotoif*, *exec*,, *execif*, *gosub* and *gosubif*) can easily remove redundant AGI code, passing the control from the AGI back to the dialplan.

Rule #8: Balance your scripts with web services

When evaluating rule #4, one may ask: "What is an out-of-band information system?" We shall explain now. Most Asterisk developers tend to develop their systems with the data information system—either embedded into their Asterisk server or communicating with an information system installed on the same server with the Asterisk server.

While, for small systems, this proves to be both efficient and economic, when developing a high-end system that requires scalability and redundancy, this methodology proves to be counter-productive. One of the methodologies (although many others exist) for interconnecting Asterisk with an out-of-band information system is web services. Communication to the web service is performed via AGI; the web-service protocol—you can use your favorite one.

The choice of protocol isn't that important, as almost any protocol type used for web services would do. Be it SOAP, WSDL, XML-RPC, WDDX or any other, take your pick, and the performance and scalability should be similar in any of these.

Rule #9: Syslog is your friend—use it

Every developer knows that using log files for debugging and monitoring purposes is a must. Be it for using a binary compiled AGI or a scripting language based AGI, make sure to utilize the logging facility. Trying to debug an AGI application from within the Asterisk console, though possible, can prove to be a tedious task. Sending log entries to a well-formatted log can save you much time and headache.

Scripting languages, such as PHP and PERL, do not offer a direct debugging facility, making the debugging of such AGI scripts even harder. Using log files as a debugging point for your AGI script will prove very useful when developing highly complex systems.

In order to make your syslog more readable, assign a self-created unique ID to each of your calls. When writing to your log, make sure that the unique ID appears in each log entry, so that you can trace a specific session flow through Asterisk. Remember, an Asterisk channel is *stateful. The unique ID will remain as part of the channel until it is removed from the system.*

Rule #10: The Internet is for Asterisk

As bad as the following may sound, if you have a problem or an idea, remember that someone else had almost definitely come across it before you did. I don't want to discourage you, but actually, I want you to make use of the multitude of Asterisk resources available on the Internet.

The amount of information relating to Asterisk and platform development that has been accumulated by search engines is staggering. Over the course of the past two years, the amount of information available has multiplied two times (at least), making it the best source to find answers to your questions.

Asterisk user forums exist today in almost every country around the world; in some countries, there is more than one forum. These forums provide fast answers and professional guidance, allowing you to concentrate on your development, instead of concentrating on obtaining information.

11.6 AGI Commands

* **`answer:`** Asserts answer

* **`asyncagi break:`** Break Async AGI loop (since Asterisk 1.6)

* **`channel status:`** Returns status of the connected channel

* **`control stream file:`** Send the given file, allowing playback to be controlled by the given digits, if any. (since Asterisk 1.2)

* **`database del:`** Removes database key/value

* **`database deltree:`** Removes database keytree/value

* **`database get:`** Gets database value

* **`database put:`** Adds/updates database value

* **`exec:`** Executes a given Application. (Applications are the functions you use to create a dial plan in extensions.conf).

* **`get data:`** Gets data on a channel

* **`get full variable:`** Gets a channel variable, but understands complex variable names and built-in variables. (since Asterisk 1.2)

* **`get option:`** Behaves similar to STREAM FILE but used with a timeout option. (since Asterisk 1.2)

* **`get variable:`** Gets a channel variable

* **`hangup:`** Hangup the current channel

* **`noop:`** Does nothing

* **`receive char:`** Receives one character from channels supporting it

* **`receive text:`** Receives text from channels supporting it

* **`record file:`** Records to a given file

* **`say alpha:`** Says a given character string (since Asterisk 1.2)

* **`say date:`** Say a date (since Asterisk 1.2)

* **`say datetime:`** Say a formatted date and time (since Asterisk 1.2)

* **`say digits:`** Says a given digit string

* **`say number:`** Says a given number

* **`say phonetic:`** Say the given character string.

* **`say time:`** Say a time

* **`send image:`** Sends images to channels supporting it

* **`send text:`** Sends text to channels supporting it

* **`set autohangup:`** Autohangup channel in some time

* **`set callerid:`** Sets callerid for the current channel

* **`set context:`** Sets channel context

* **`set extension:`** Changes channel extension

* **`set music:`** Enable/Disable Music on hold generator, example "SET MUSIC ON default"

* **`set priority:`** Prioritizes the channel

* **`set variable:`** Sets a channel variable

* **`speech activate grammar:`** Activates a grammar (since Asterisk 1.6)

* **`speech create:`** Creates a speech object (since Asterisk 1.6)

* **`speech deactivate grammar:`** Deactivates a grammar (since Asterisk 1.6)

* **`speech destroy:`** Destroys a speech object (since Asterisk 1.6)

* **`speech load grammar:`** Loads a grammar (since Asterisk 1.6)

* **`speech recognize:`** Recognizes speech (since Asterisk 1.6)

* **`speech set:`** Sets a speech engine setting (since Asterisk 1.6)

* **`speech unload grammar:`** Unloads a grammar (since Asterisk 1.6)

* **`stream file:`** Sends audio file on channel

* **`tdd mode:`** Activates TDD mode on channels supporting it, to enable communication with TDDs.

* **`verbose:`** Logs a message to the asterisk verbose log

* **`wait for digit:`** Waits for a digit to be pressed

Note: When Asterisk starts an AGI script, it feeds the channel variables to the script on standard input. The variable names are prefixed with "agi_" and are separated from their values by a colon and a space.

Though the actual channel variables may be in the upper case, the names passed to an AGI script are all lower case. Global variables are not passed to the AGI script in this manner.

* **agi_request** - The filename of your script
* **agi_channel** - The originating channel (your phone)
* **agi_language** - The language code (e.g. "en")
* **agi_type** - The originating channel type (e.g. "SIP" or "ZAP")
* **agi_uniqueid** - A unique ID for the call
* **agi_version** - The version of Asterisk (since Asterisk 1.6)
* **agi_callerid** - The caller ID number (or "unknown")
* **agi_calleridname** - The caller ID name (or "unknown")
* **agi_callingpres** - The presentation for the callerid in a ZAP channel
* **agi_callingani2** - The number which is defined in ANI2 see Asterisk Detailed Variable List (only for PRI Channels)
* **agi_callington** - The type of number used in PRI Channels see Asterisk Detailed Variable List
* **agi_callingtns** - An optional 4 digit number (Transit Network Selector) used in PRI Channels see Asterisk Detailed Variable List
* **agi_dnid** - The dialed number id (or "unknown")
* **agi_rdnis** - The referring DNIS number (or "unknown")
* **agi_context** - Origin context in extensions.conf
* **agi_extension** - The called number
* **agi_priority** - The priority it was executed as in the dial plan
* **agi_enhanced** - The flag value is 1.0 if started as an EAGI script, 0.0 otherwise
* **agi_accountcode** - Account code of the origin channel
* **agi_threadid** - Thread ID of the AGI script (since Asterisk 1.6)

Using the "get variable" AGI command to get values above.

11.7 simple PHPAGI example

at first make extension 888 in extension_custom.conf like this

```
include => test


[test]

exten => 85,1,Answer

exten => 85,n,Wait(2)

exten => 85,n,AGI(test-php.php)

exten = 85,n,hangup()
```

secondly make your test.php file in this path /var/lib/asterisk/agi-bin and don’t forget to give it execution permission.

```
#!/usr/bin/php -q

<?

set_time_limit(30);

require('include/phpagi.php');

error_reporting(E_ALL);

$agi = new AGI();

$agi->answer();

$agi->stream_file("demo-congrats","#");

do

{

  $agi->stream_file("enter-some-digits","#");

  $result = $agi->get_data('beep', 3000, 20);

  $keys = $result['result'];

  $agi->stream_file("you-entered","#");

  $agi->say_digits($keys);

} while($keys != '111');
```

```
$agi->hangup();

?>
```

Now reload asterisk and call 888 from your extension.

11.8 Interact with MySQL inside PHP-AGI

The following AGI retrieves asterisk extensions from Asterisk database in MySQL and displays them as verbose.

```
#!/usr/bin/php5

<?php

set_time_limit(30);

require('phpagi.php');

error_reporting(E_ALL);


$agi = new AGI();


mysql_connect('localhost','root','SmartPBX');


$agi -> verbose('connection set');


mysql_select_db('asterisk');


$sql = "select extension, name from users";

$agi -> verbose($sql);


$result = mysql_query($sql);
```

```
while($row = mysql_fetch_assoc($result))

{

$rows =  "Ext # :{$row['extension']} " . " *** | *** " .
"Name : {$row['name']}";

$agi->verbose($rows);

}

mysql_close();

?>
```

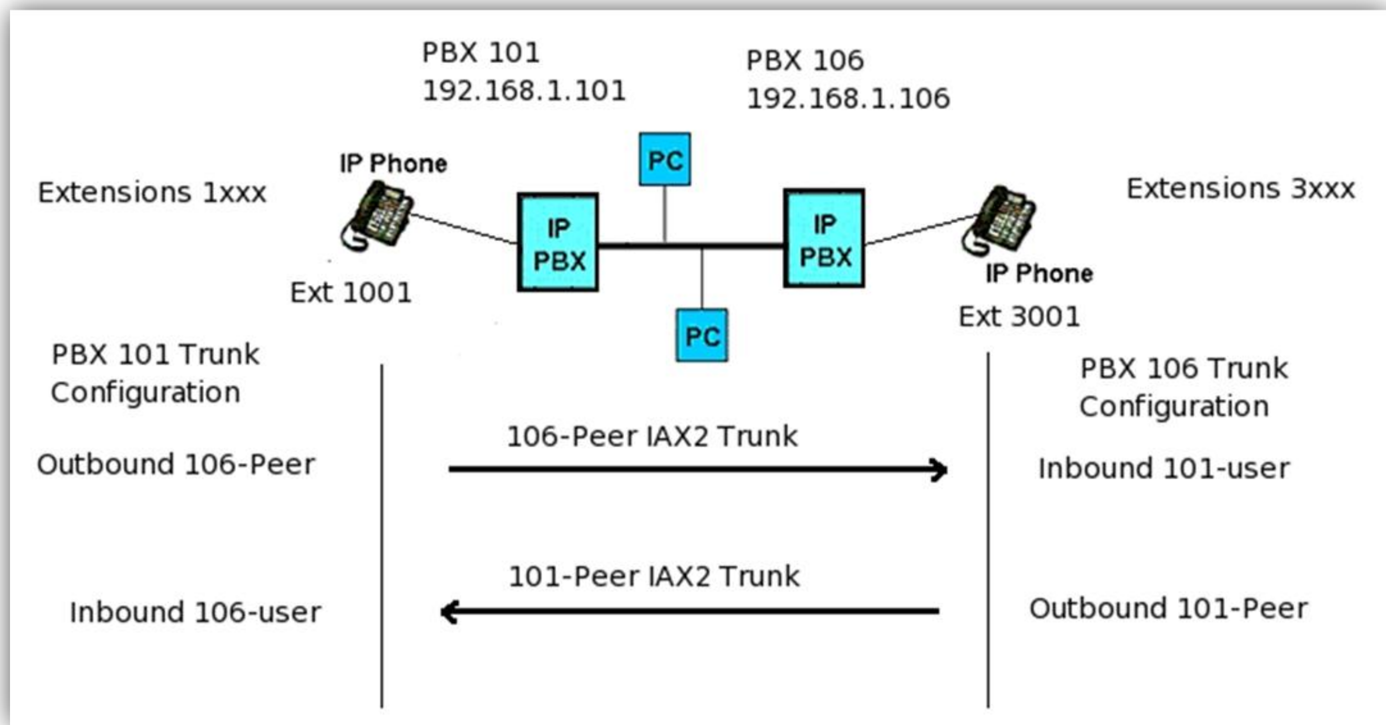
12. Tips and tricks

12.1 connecting 2 servers of Elastix together

It is possible to connect two servers with different ways, the easiest and most reliable way is connecting with IAX protocol. The following will express how to define this relationship. The first step in the trunk of IAX2 is drawing a picture from whatever you need. Here we have a simple connection between two PBX, for convenience name them 101 and 106 (according the IP of two PBX).

All the extensions of PBX 101 start with 1xxx and all the extensions of PBX 106 start with 3xxx.

This is good for creating outbound route.



Trunks of IAX2 are named according to their destinations and are shown with arrows.

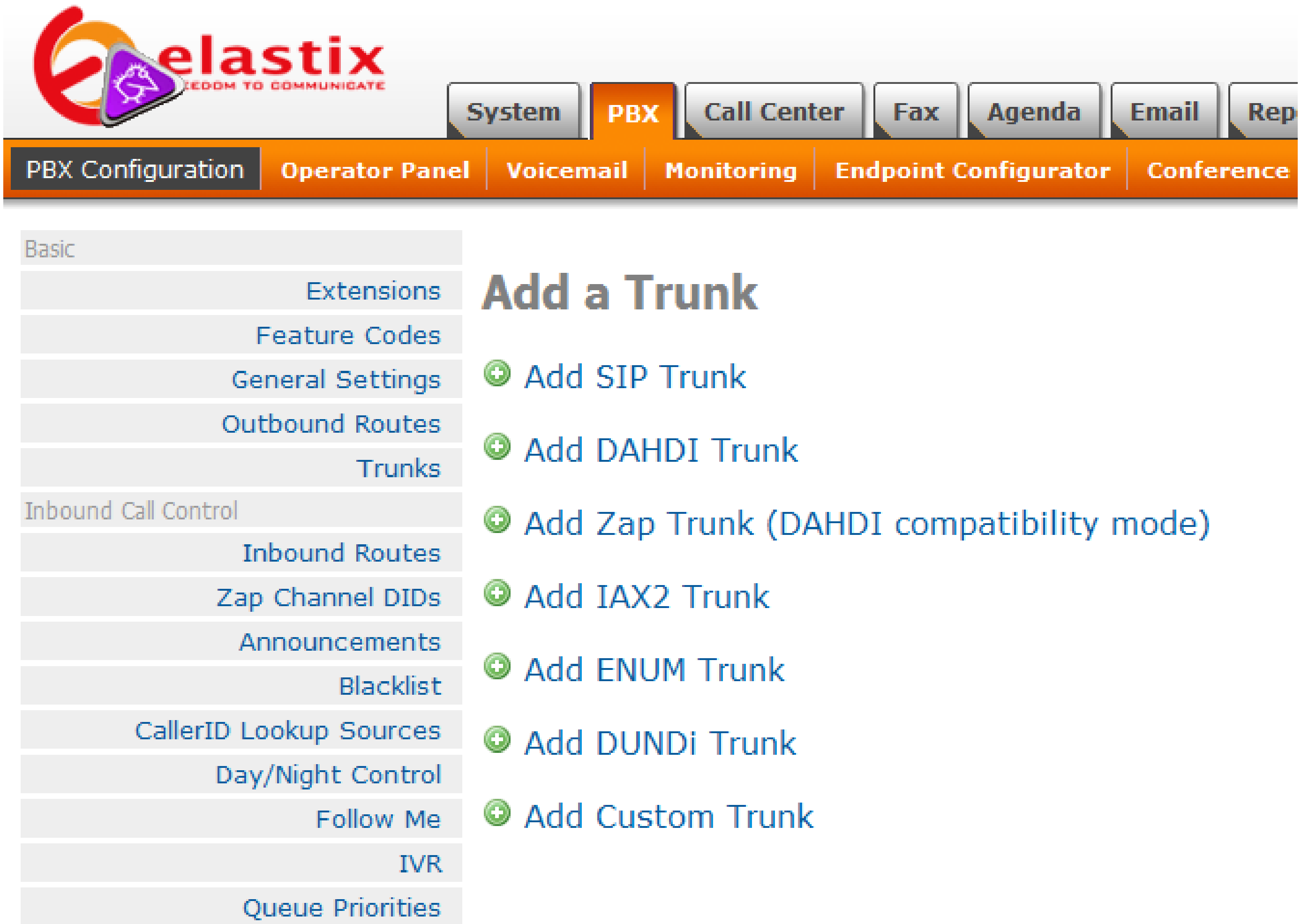
There is information for configuration in two sides of PBXs, as it is shown in PBX 101 it is needed to have outbound trunk which is peer-106 and user-106. Similarly, in PBX 106 some information for configuration is needed for having an outbound trunk with peer-101 and user-101.

We start by having a trunk when both PBX have IAX2 trunk, we define outbound route.

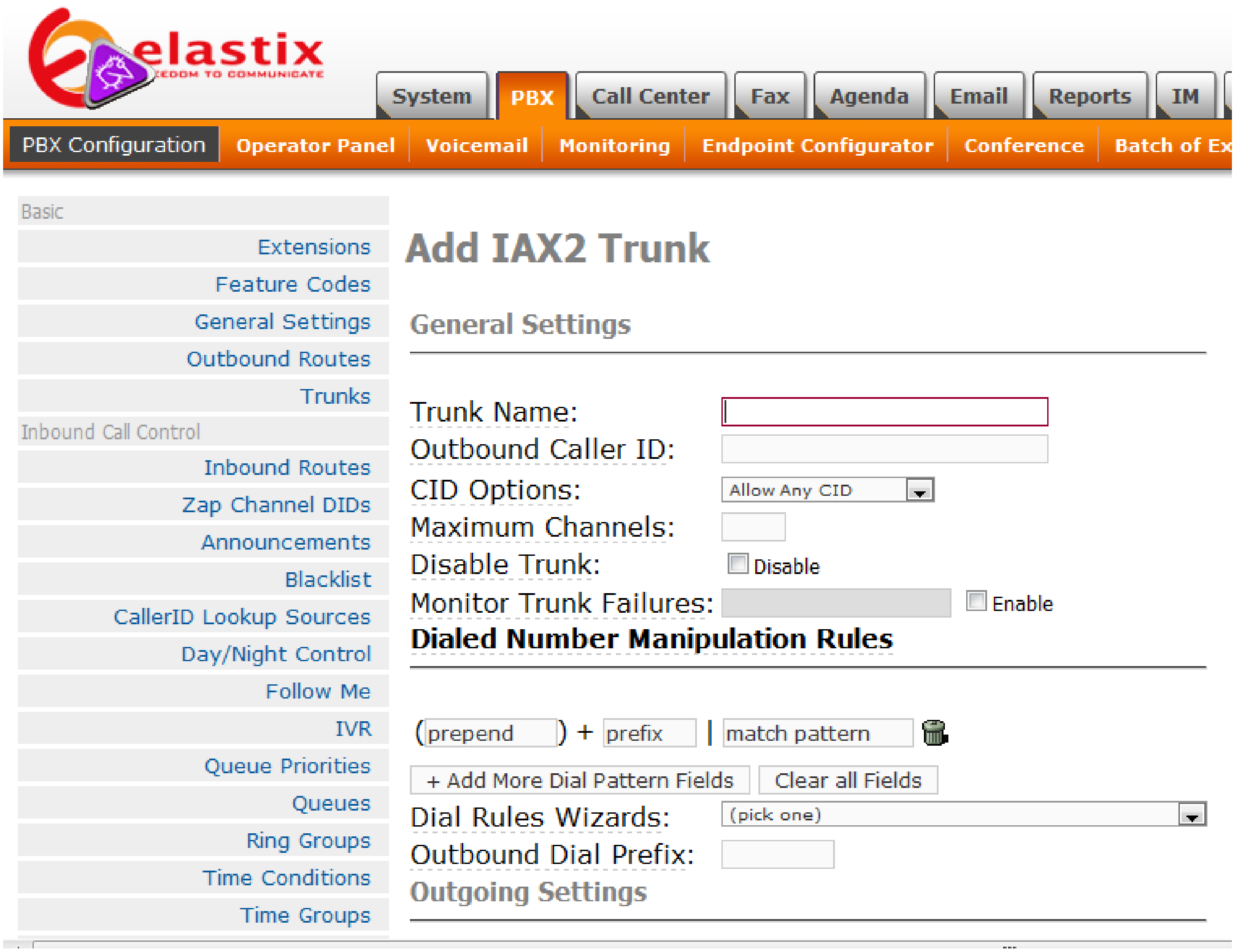
- Setting trunks of IAX2

IAX2 trunk in PBX101 & PBX106

A. Select add trunk in main menu of FreePBX



B. Select add LAX2 Trunk



C. Don't change anything, in Outgoing Settings for configuration of PBX 106 and PBX 101 do as follow:

ELASTIX-SERVER1(192.168.1.101)

Trunk Name: 106-IAX

PEER Details:

```
host=192.168.1.106
username=101-IAXuser
secret=101-IAXpassword
type=peer
qualify=yes
```

Incoming Settings

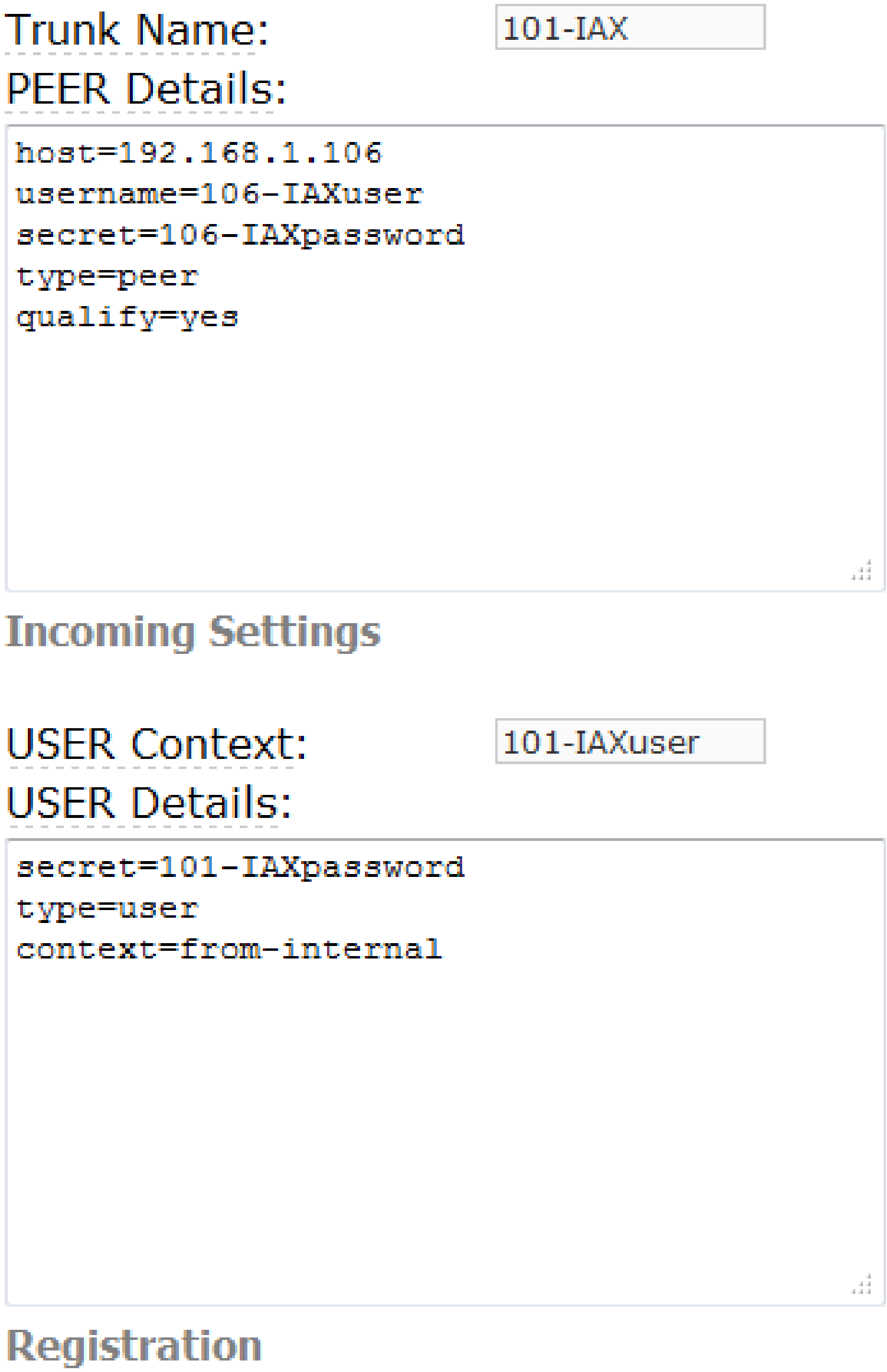
USER Context: 106-IAXuser

USER Details:

```
secret=106-IAXpassword
type=user
context=from-internal
```

Registration

ELASTIX-SERVER1(192.168.1.106)



D. Outgoing setting will be configured by basic information that is destination of IAX user.

Configure outgoing settings and incoming settings in PBX 101.

Outgoing Settings:

***Trunk Name: 106-IAX it can be any name.

*192.168.1.106:host_   it is the name of domain or address of destination trunk.

*username=101-IAXuser it is LAX2 user which is made in destination PBX.

*secret=101-IAXpassword password used for confirming the trunk connection in destination PBX.

*type=peer kind of LAX2 connection. This connection is from one PBX to another PBX.

*qualify=yes it register connection with the destination.

Incoming Settings:

***user context: 106-IAXuser it can be any name.

*secret=101IAXpassword   password used for confirming the trunk connection in destination PBX.

*type=user kind of IAX2 connection. Peer is confirmed by user account.

For configuring the outgoing settings and incoming settings in PBX106 follow the configuration in image.

E. Click on submit, update and reload it. And check it with webmin or other management softwares. File of / etc/Asterisk/iax-additional.conf in PBX101 should be configured as follow:

```
; do not edit this file, this is an auto-generated file by freepbx
; all modifications must be done from the web gui


[106-peer]
host=192.168.1.106
username=101-user
secret=1234
type=peer
qualify=yes
trunk=yes

[106-user]
secret=1234
type=user
context=from-trunk
```

And file of /etc/Asterisk/iax-additional.conf in PBX106 as below:

```
; do not edit this file, this is an auto-generated file by freepbx
; all modifications must be done from the web gui


[101-peer]
host=192.168.1.101
username=106-user
secret=1234
type=peer
qualify=yes
trunk=yes

[101-user]
secret=1234
type=user
context=from-trunk

[3002]
```

F. Go to the next level and test the trunks of IAX2

- Testing the trunks of LAX2:

Go to Asterisk console for testing the trunks (CLI), you can do it with several ways.

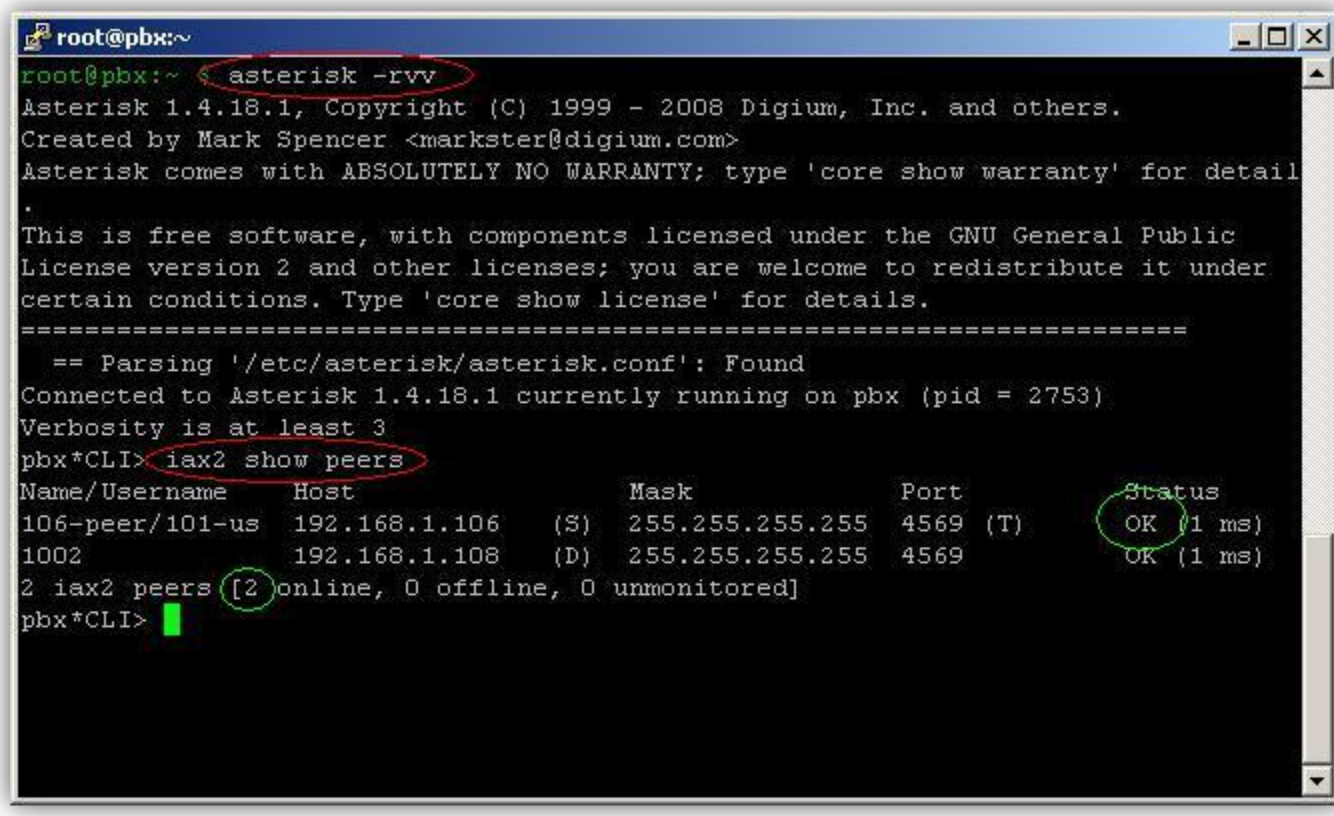
- ✓ Through console of PBX
- ✓ Through doing SSH by using Putty SSh
- ✓ Through FreePBX/Webmin SSH

Enter the following command in Linux.

root@pbx~$ Asterisk –rvv

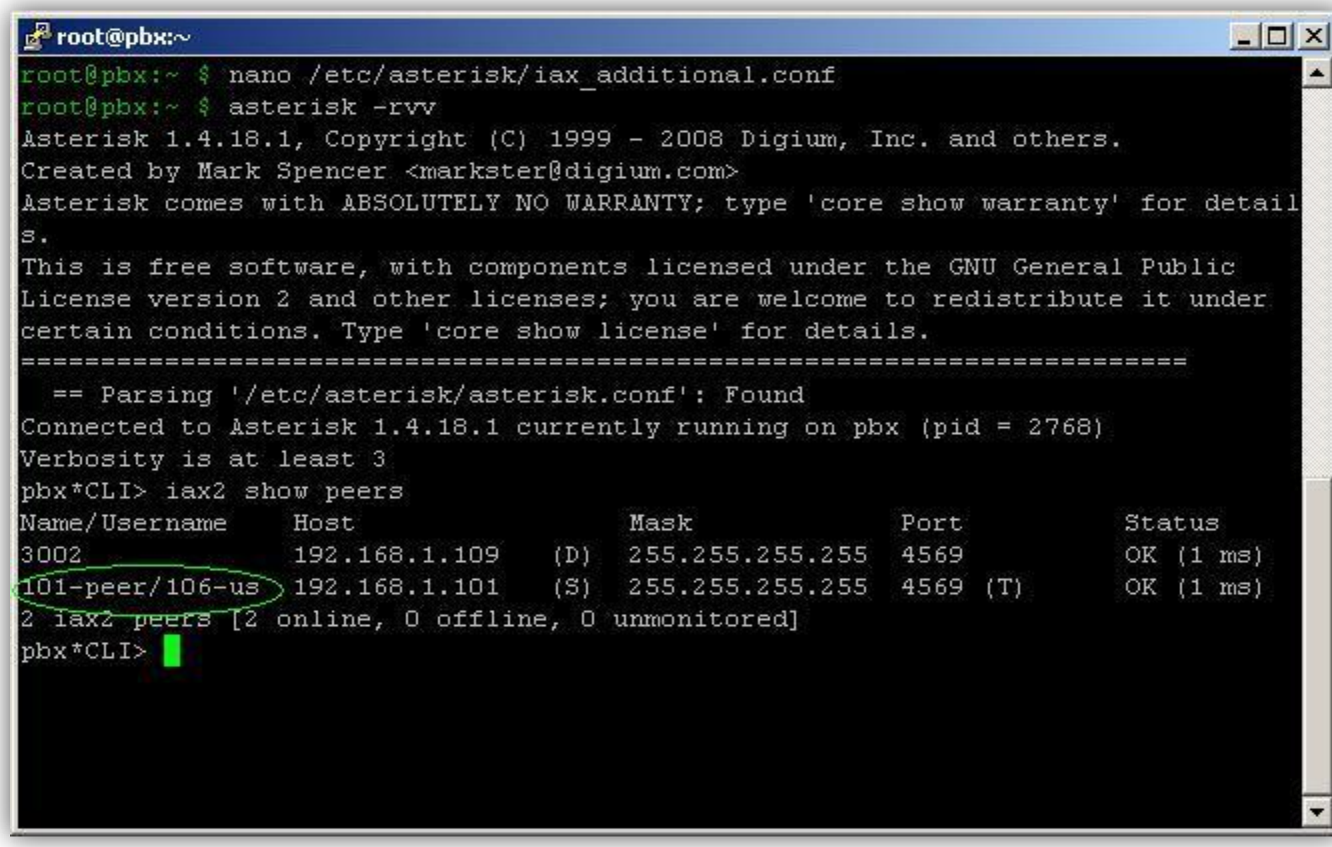
(One r & two v)

In Asterisk click on command of iax2 show peers.



Above image shows the connections of PBX 101 IAX peer. Pay attention to its important parts:

- ✓ Above image shows that peer-106 is connected by user-101
- ✓ Status should be ok, change it if it was something else.
- ✓ Client of 1002 and IAX2 are connected and are 2 high peers.



```
root@pbx:~
root@pbx:~ $ nano /etc/asterisk/iax_additional.conf
root@pbx:~ $ asterisk -rvv
Asterisk 1.4.18.1, Copyright (C) 1999 - 2008 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for detail
s.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=========================================================================
  == Parsing '/etc/asterisk/asterisk.conf': Found
Connected to Asterisk 1.4.18.1 currently running on pbx (pid = 2768)
Verbosity is at least 3
pbx*CLI> iax2 show peers
Name/Username    Host                 Mask             Port          Status
3002             192.168.1.109   (D)  255.255.255.255  4569          OK (1 ms)
101-peer/106-us  192.168.1.101   (S)  255.255.255.255  4569 (T)      OK (1 ms)
2 iax2 peers [2 online, 0 offline, 0 unmonitored]
pbx*CLI>
```

Above image shows the connection of PBX 106, IAX2 peer. The important parts are:

Above Image shows that peer-101 is connected by user-106

- ✓ Status is ok, check the setting if it was something else.
- ✓ Client of 3002 of IAX is connected.
- ✓ And 2 peers are online.

Next stage is definition of outbound route for PBX 101 & PBX 106.

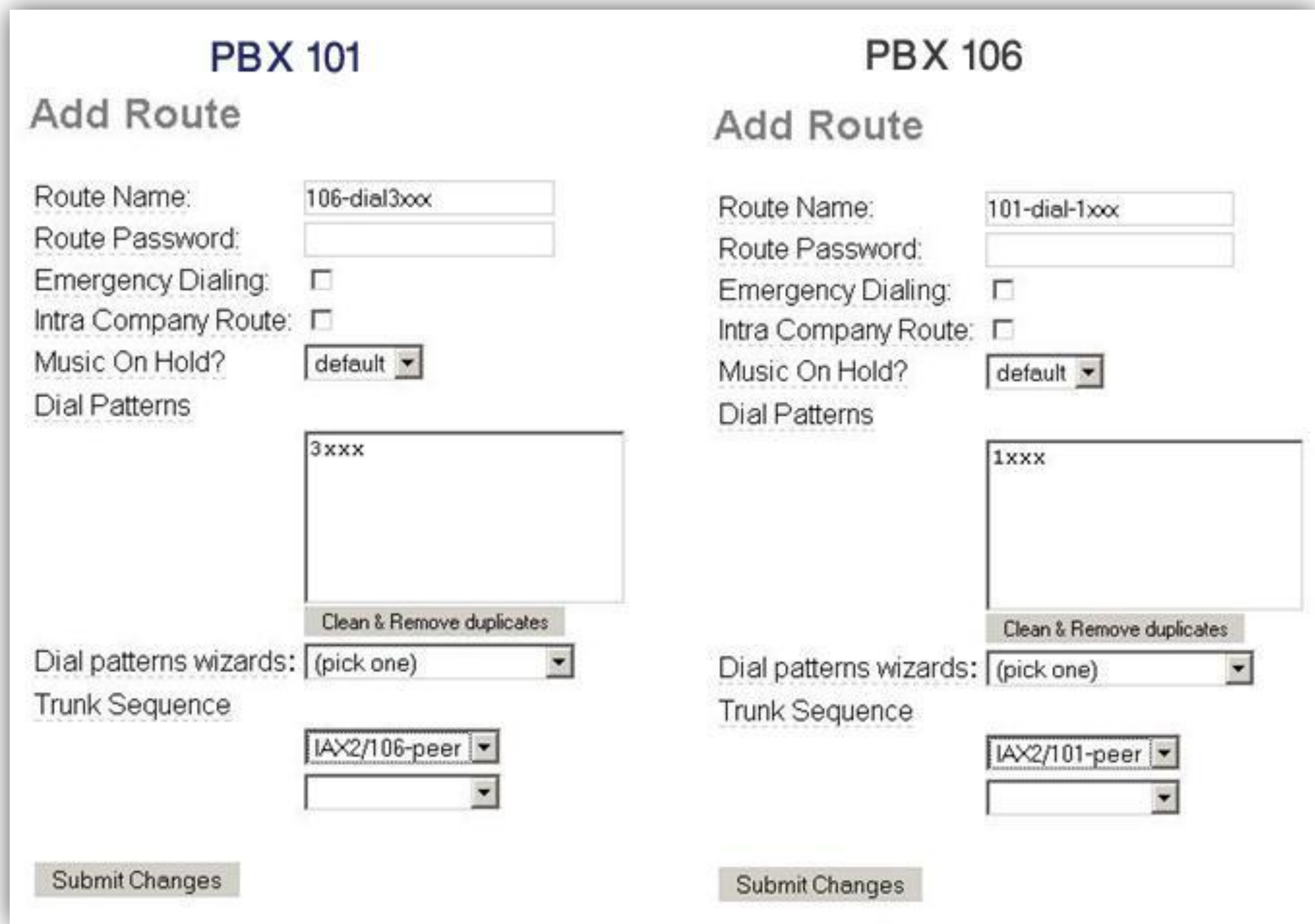
- Setting outbound routes

There are 2 options for setting the outbound routs: Direct call to the extension or entering number 7. Both will be described.

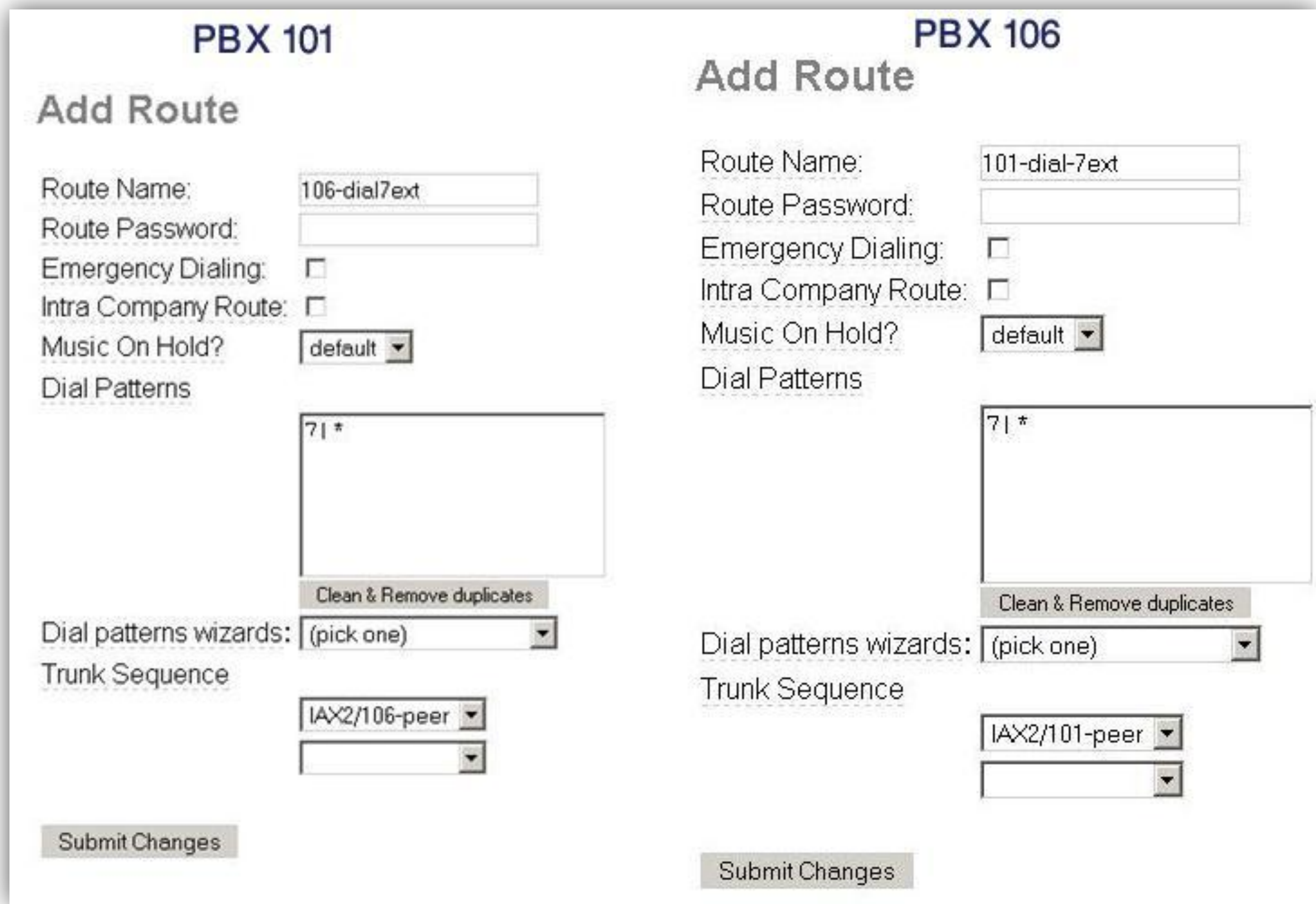
Outbound routs in PBX 101

There are 2 ways for outbound routes, select one of them and start with outbound route in the menu.

First_ calling directly to the extension, PBX101&PBX106



Rules of outbound allow PBX 101 call the extensions of PBX 106 directly. For example for calling to extension 3001, dial 3001,PBX 106.and also rules of outbound allows pbx 106 to call directly to extensions of PBX101. For example for calling to extension 1001, dial 1001,pbx 101.



Configure these in PBX 101:

*route name: 106-dial7ext_ it can be anything

**|7_ shows that everything after 7 will be forward to destination trunk.

*trunk sequence: IAX2/106-peer_ this trunk is made for going to the PBX 106.

Configure these in PBX 106:

*route name: 101-dial7ext: can be anything.

**|7_ shows that everything after 7 will be forwarded to destination trunk.

*trunk sequence: IAX2/101-peer_ this trunk is made for going to the PBX 101

You should send IAX or POST to peers of each PBX from any SIP extension.

You can check status of online phones, trunks through FreePBX window.

In making trunk, there is no limitation in number of channels that can use trunk. In above window, 6 channels were connected in 3 connections through a trunk of IAX2. Three IP phones, two IAX2 s100i POSTS attached to adaptors of IAX2 and a FXS pots phone.

12.2 Video call on Elastix

For having video call on Elastix you should do following configuration. Go to this link. Open the file of "sip_general_additional.conf" for edit. If there was no file, make it: "vi /etc/Asterisk/sip_general_additional.conf"

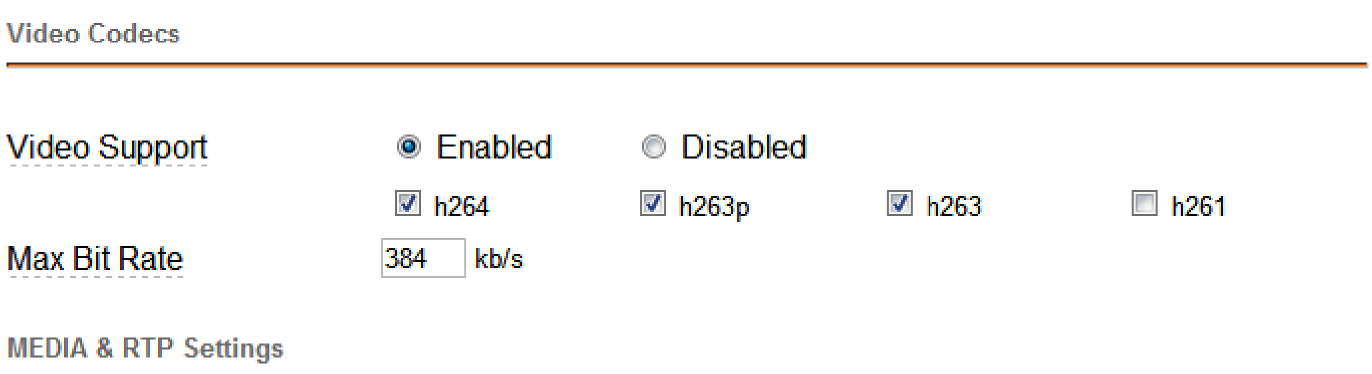
Add this configuration:

```
Videosupport=yes

Maxcallbitrate=384

Allow=h261

Allow=h263

Allow=h263p

Allow=h264
```

And reload it with this command:

Asterisk -rx "module reload"

in freePBX version 2.7 you can set it easily from graphical environment. So go to Elastix menu, PBX configuration and click on sub menu of Unembedded FreePBX to enter freepbx program. Then go to Tools, sub menu of Asterisk SIP setting or Asterisk IXA setting and enable the video support.



Some of softphones that supports video are listed:

Bear in mind that you should activate video support on any softphone you use.

software	url	platform
Ekiga	http://www.ekiga.org	**Linux**
Adore Video	http://www.adoresoftphone.com/softphones/softphone-video.html	**Windows**
Eyebeam	http://www.counterpath.com/index.php?menu=Products&smenu=eyeBeam	**Windows**
Bria 2.0	http://www.counterpath.com/index.php?menu=Products&smenu=bria	**Windows**

12.3 Limiting the conversation time

A way for limiting and categorizing the extensions is using custom-context module which will be explained later. But it's using has trouble because of the complexity. Now if you want to limit time of conversation for all the user you can use this trick. Go to PBX, PBX configuration, General setting and add L(3600000:300000) in Asterisk outbound dial command options.

Your configuration will be like this:

trL (3600000:300000)

The above command is L (x:y). x is the time allowed for conversation in milliseconds and call will be terminated in this time and y is remaining time of the call for announcing the warning for disconnecting it. In fact this example means that a call will be terminated in 60 minutes and 5minute remaining to the end of the call, a warning announced. For more information see the following reference.

http://www.voip-info.org/wiki/view/Asterisk+cmd+Dial

12.4 put your Asterisk server behind NAT

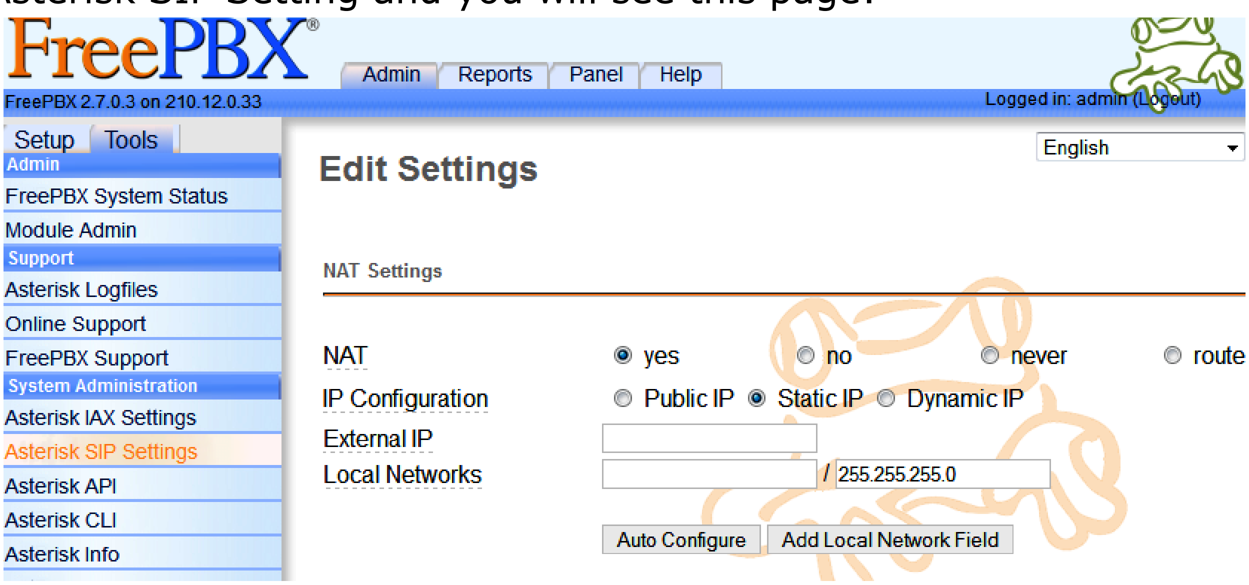
- **Manually:** it is possible that you want to enter your extensions on internet and be registered on your server. Maybe you don't have

valid IP and put your server behind NAT and you should have access to internet through that sever. In this case you should have some configuration on your Asterisk to operate properly behind NAT and with valid IP, extensions registered on it and contact establish correctly.
Put these items in “sip_nat.conf” in “etc/Asterisk” branch:

nat=yes
externip=<your fixed external IP> or
externhost=<myAsterisk.freedns.com>
localnet=192.168.1.0/255.255.255.0
externrefresh=10
Extern ip is your valid ip and is map to you. If you use ADSL, it is your modem IP. In addition, you should forward these ports to the ip server on your ADSL modem or router.

tcp: 5060
udp: 10000-20000

- **Graphical:** these can be done through SIP setting in freepbx. These features is set on freepbx from version 2.7 so go to the pbx configuration from Elastix menu and click on unembedded freepbx to enter the freepbx program. Then go to Tools and sub menu of Asterisk SIP Setting and you will see this page:



All the parameters that were written in this file can be easily entered here. Don't forget there are always an easy way!!!!

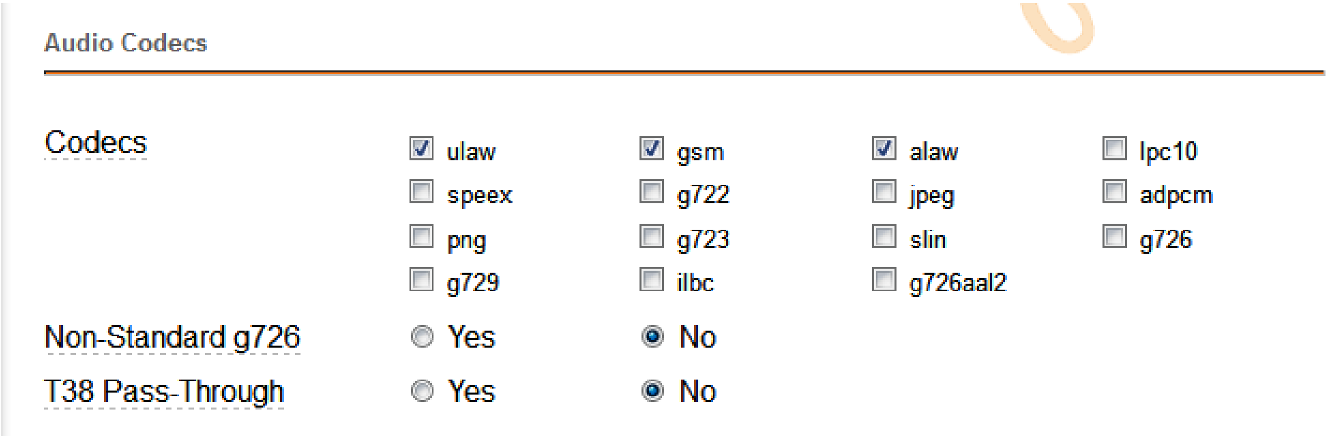
12.5 Installing Codecs of g729 & g723

Every call that is established use a protocol to talk and a compactor codec for compressed audio packets. It will use less bandwidth if the packages become more compressed by codec but you will have more processing load on the processor and the quality of voice will be less.

```
cd /usr/src

 Service Asterisk restart

wget http://Asterisk.hosting.lv/bin/codec g723-ast 14- icc-glibc-pentium4.so

mv codec_g723-ast14-icc-glibc-pentium4.so/usr/lib/Asterisk/modules

chmod +x /usr/lib/Asterisk/modules/codec_g723-ast14-icc-glibc-pentium4.so

wget http://Asterisk.hosting.lv/bin/codec g729-ast 14- icc-glibc-pentium4.so

mv codec_g729-ast14-icc-glibc-pentium4.so/usr/lib/Asterisk/modules

chmod +x /usr/lib/Asterisk/modules/codec_g729-ast14-icc-glibc-pentium4.so

Asterisk -rx "module load codec_g729-ast14-icc-glibc-pentium4.so"

Asterisk -rx "module load codec_g723-ast14-icc-glibc-pentium4.so"

Service Asterisk restart

Sleep 2

Asterisk -rx "core show codecs
```

After installing the packages, you should allow the VoIP protocols use these codecs. Add "Allow = g729" & "Allow = g723" to "/etc/Asterisk/sip_custom.conf "&" /etc/Asterisk/sip_custom.conf".

You can add these from graphical environment. For this go to Elastix menu, PBX configuration and click on Unembeded FreePBX and entered to FreePBX program. Then go to Tools, sub menu of Asterisk sip setting or Asterisk IXA setting. You will see following page:



12.6 Asterisk command-line interface (CLI)

Asterisk has an environment that can be changed by some commands or be a suitable device for controlling Asterisk. For accessing to this environment which is called CLI, you should enter this command after entering to Linux.

Asterisk-rvvv

“r” means performing the environment and “v” means view details of the system. The more the number of “v”, the more you can see details from log of Asterisk, but for normal items, 3 “v” is enough.

In this environment you can see whatever Asterisk does like a log. In addition, you can configure by using commands or view more details.

For seeing the list, enter the “Help” on CLI environment.

This allows you to run a command as if it was typed into the asterisk CLI. *Examples:*

* **sip show peers**
o This displays all the known SIP devices, and their state, according to Asterisk
* **show channels**

o Show any channels that are in use at the moment
* **soft hangup Zap/1**
o Hangs up the Zap/1 channel

General commands

abort halt: Cancel a running halt
add extension: Add new extension into context
add ignorepat: Add new ignore pattern
add indication: Add the given indication to the country
debug channel: Enable debugging on a channel
don't include: Remove a specified include from context
help: Display help list, or specific help on a command
include context: Include context in other context
load: Load a dynamic module by name logger reload: Reopen log files. Use after rotating the log files.
no debug channel: Disable debugging on a channel
pri debug span: Enables PRI debugging on a span
pri intense debug span: Enables REALLY INTENSE PRI debugging
pri no debug span: Disables PRI debugging on a span
remove extension: Remove a specified extension
remove ignorepat: Remove ignore pattern from context
remove indication: Remove the given indication from the country
save dialplan: Overwrites your current extensions.conf file with an exported version based on the current state of the dialplan. A backup copy of your old extensions.conf is not saved. The initial values of global variables defined in the [globals] category retain their previous initial values; the current values of global variables are not written into the new extensions.conf. (:exclaim:) Using "save dialplan" will result in losing any comments in your current extensions.conf.
set verbose: Set level of verboseness
show agents: Show status of agents
show applications: Shows registered applications

show application: Describe a specific application
show channel: Display information on a specific channel
show channels: Display information on channels
show codecs: Display information on codecs
show conferences: Show status of conferences
show dialplan: Show dialplan
show hints: Show registered hints
show image formats: Displays image formats
show indications: Show a list of all country/indications
show locals: Show status of local channels
show manager command: Show manager commands
show manager connect: Show connected manager users
show parkedcalls: Lists parked calls
show queues: Show status of queues
show switches: Show alternative switches
show translation: Display translation matrix
soft hangup: Request a hangup on a given channel
show voicemail users: List defined voicemail boxes
show voicemail zones: List zone message formats

Server management

restart gracefully: Restart Asterisk gracefully, i.e. stop receiving new calls and restart at empty call volume
restart now: Restart Asterisk immediately
restart when convenient: Restart Asterisk at empty call volume

Note: Restart is more like a reload, not a real restart it just runs the reload routines (thus open ports are not closed). Often you don't need to really restart asterisk, instead just e.g. 'unload chan_sip.so' and 'load chan_sip.so'.
reload: Reload configuration
stop gracefully: Gracefully shut down Asterisk, i.e. stop receiving new calls and shut down at empty call volume
stop now: Shut down Asterisk immediately

stop when convenient: Shut down Asterisk at empty call volume
extensions reload: Reload extensions and only extensions
unload: Unload a dynamic module by name
show modules: List modules and info about them
show uptime: Show uptime information
show version: Display Asterisk version info

AGI commands

show agi: Show AGI commands or specific help
dump agihtml: Dumps a list of agi command in html format

Database handling commands

database del: Removes database key/value
database deltree: Removes database keytree/values
database get: Gets database value
database put: Adds/updates database value
database show: Shows database contents

IAX Channel commands

iax2 debug: Enable IAX debugging
iax2 no debug: Disable IAX debugging
iax2 set jitter: Sets IAX jitter buffer
iax2 show cache: Display IAX cached dialplan
iax2 show channels: Show active IAX channels
iax2 show peers: Show defined IAX peers
iax2 show registry: Show IAX registration status
iax2 show stats: Display IAX statistics
iax2 show users: Show defined IAX users
iax2 trunk debug: Request IAX trunk debug
iax debug: Enable IAX debugging
iax no debug: Disable IAX debugging
iax set jitter: Sets IAX jitter buffer

iax show cache: Display IAX cached dialplan
iax show channels: Show active IAX channels
iax show peers: Show defined IAX peers
iax show registry: Show IAX registration status
iax show stats: Display IAX statistics
iax show users: Show defined IAX users
init keys: Initialize RSA key passcodes
show keys: Displays RSA key information

H323 channel commands

h.323 debug: Enable chan_h323 debug
h.323 gk cycle: Manually re-register with the Gatekeper
h.323 hangup: Manually try to hang up a call
h.323 no debug: Disable chan_h323 debug
h.323 no trace: Disable H.323 Stack Tracing
h.323 show codecs: Show enabled codecs
h.323 show tokens: Manually try to hang up a call
h.323 trace: Enable H.323 Stack Tracing

SIP channel commands

sip debug: Enable SIP debugging
sip no debug: Disable SIP debugging
sip reload: Reload sip.conf (added after 0.7.1 on 2004-01-23)
sip show channels: Show active SIP channels
sip show channel: Show detailed SIP channel info
sip show inuse: List all inuse/limit
sip show peers: Show defined SIP peers (clients that register to your Asterisk server)
sip show registry: Show SIP registration status (when Asterisk registers as a client to a SIP Proxy)
sip show users: Show defined SIP users

Zap channel commands

soft hangup Zap/1: Hangs up the Zap/1 channel
zap destroy channel: Destroy a channel
zap show channels: Show active zapata channels
zap show channel: Show information on a channel

MGCP channel commands

mgcp audit endpoint: Audit specified MGCP endpoint
mgcp debug: Enable MGCP debugging
mgcp no debug: Disable MGCP debugging
mgcp show endpoints: Show defined MGCP endpoints

skinny channel commands

skinny debug: Enable Skinny debugging
skinny no debug: Disable Skinny debugging
skinny show lines: Show defined Skinny lines per device

CAPI channel commands

capi debug: Enable CAPI debugging
capi no debug: Disable CAPI debugging
capi info: Show CAPI info

Sirrix ISDN channel commands

srx reload: Reload channel driver configuration; active calls are not terminated!
srx show ccmsgs: Disable / enable output of incoming callcontrol messages.
srx show chans: Show info about B-Channels
srx show globals: Show info about global settings
srx show groups: Show info about configured groups
srx show layers: Show info about ISDN stack (Layer 1, 2, 3)

srx show sxpvts: Show private info about active channels
srx show timers: Show info about running timers

vISDN ISDN channel commands (Driver which supports Euro ISDN for HFC chipsets)
visdn reload: Reloads vISDN configuration from the `/etc/asterisk/visdn.conf file`
show visdn calls: Shows active calls going through a vISDN channel.
show visdn huntgroups: Shows vISDN huntgroup information configured correctly in visdn.conf.
show visdn interfaces: Shows configured and available vISDN interfaces.
debug visdn generic: Enables generic vISDN debugging
debug visdn q921: Enables q.921 debugging
debug visdn q931: Enables q.931 debugging
no debug visdn generic: Disables generic vISDN debugging
no debug visdn q921: Disables q.921 debugging
no debug visdn q931: Disables q.931 debugging

12.7 Asterisk feature codes

There are some codes on Asterisk as a default. Each of them has a feature and you can dial them as extensions:

- Transfer (during conversation)

70 - Park Extension
*30 - Blacklist a Number
*32 - Blacklist the Last Caller
*31 - Remove a Blacklisted Number
*72 - Call Forward All Activate
*73 - Call Forward All Deactivate
*74 - Call Forward All Prompting Deactivate
*90 - Call Forward Busy Activate
*91 - Call Forward Busy Deactivate
*92 - Call Forward Busy Prompting Deactivate

*52 - Call Forward No Answer/Unavailable Activate
*53 - Call Forward No Answer/Unavailable Deactivate
*70 - Call Waiting Activate
*71 - Call Waiting Deactivate

** - Call Pickup (dialing another extension)

555 – ChanSpy (hearing all the channels)

7777 - Simulate Incoming Call
666 - Simulate Incoming FAX Call
*12 - User Logoff
*11 - User Logon

888 – ZapBarge (hearing urban lines)

*35 - Email completed dictation
*34 - Perfom Dictation
*78 - Do Not Disturb Activate
*79 - Do Not Disturb Deactivate
*422 - Connect to Gabcast
*69 - Call Trace
- Directory
*43 - Echo Test
*65 - Get Your Extension
*60 - Get the Time
*80 - Intercom Prefix
*54 - User Intercom Allow
*55 - User Intercom Disallow
411 - Phonebook dial-by-name directory
*99 - Check Recording
*77 - Save Recording
*75 - Set user speed dial
*0 - Speeddial prefix
*98 - Dial Voicemail
*97 - My Voicemail

12.8 Reading asterisk log files

Perhaps the most important skill required to troubleshoot is reading the log files. Like other Linux services, asterisk writes the logs in `/var/log/asterisk`

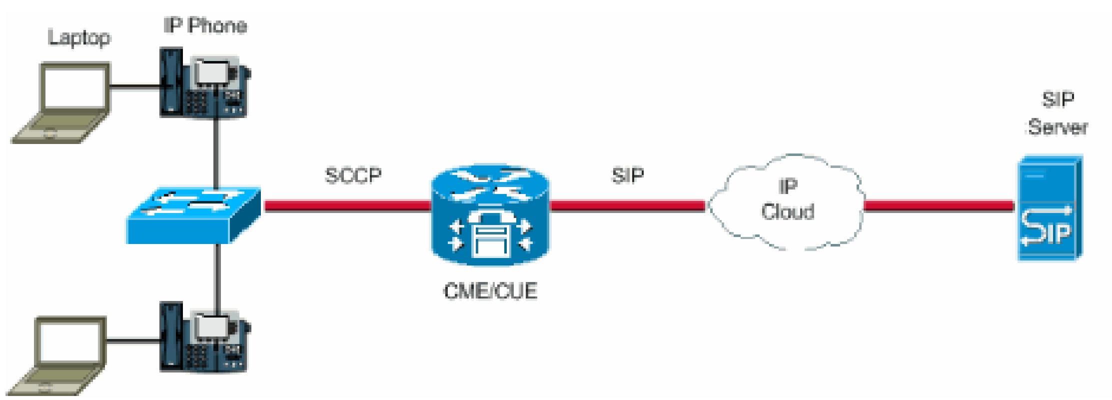
To enable a full log you should edit the /etc/asterisk/logger.conf and uncomment the full line. Afterwards you must run this command to apply changes:

asterisk -rx “logger reload”

Here are examples of tracing in logs:

```
cat /var/log/asterisk/full | grep ERROR

cat /var/log/asterisk/full.1 | grep 9878 | grep “some text”

cat /var/log/asterisk/full|grep “user-callerid: “|grep 9899 >
/var/log/asterisk/9899.log

cat /var/log/asterisk/full.1| grep “[22435]” > 9864.log

tail /var/log/asterisk/full | tee log_exmpl.txt
```

12.9 Asterisk integration with Cisco Call Manager



Why integrate Cisco CallManager and Asterisk?

- Features: Asterisk provides features that CallManager by itself does not.
- Migration: Allow a gradual migration from a closed source PBX to open source PBX.

There are two ways to accomplish this:

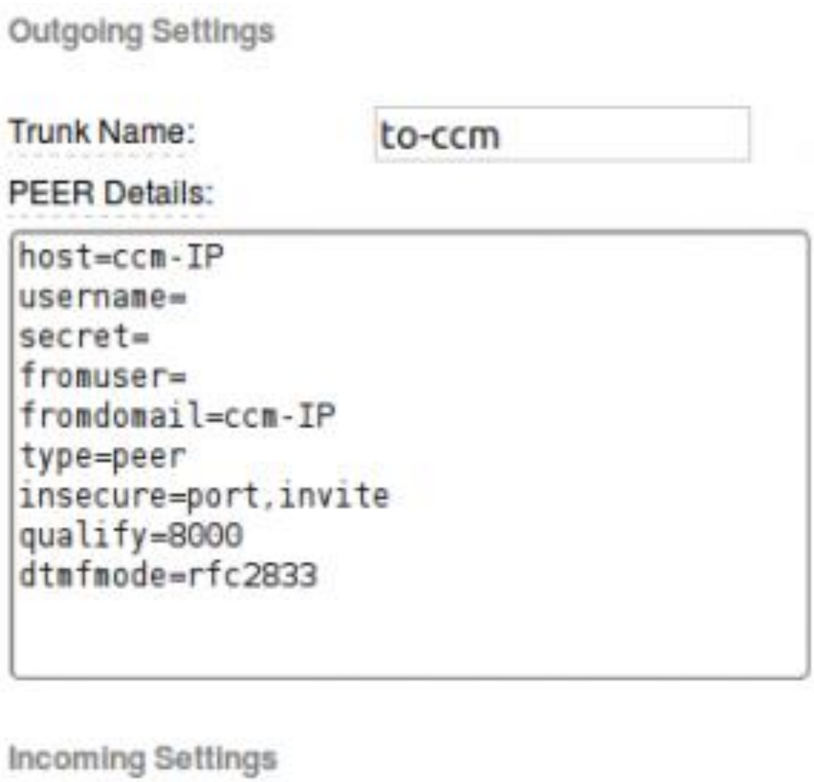
1. Using H.323: In CCM Asterisk appears as a H.323 Gateway.

2. Using SIP (only in CCM 4.X+)

Here is instruction on how to integrate asterisk with Cisco Call Manager using SIP trunk.

Step1) On your freePBX create a new SIP trunk and set the following (remain the rest as is):

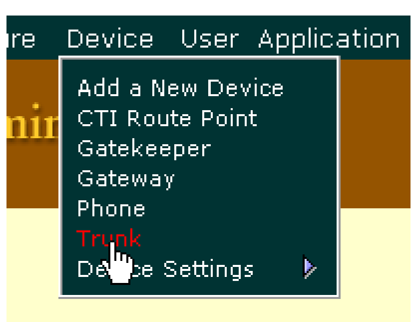
```
host=10.200.214.10
port=5063
fromdomain=10.200.214.10
fromuser=
username=
secret=
type=peer
qualify=yes
canreinvite=no
nat=no
insecure=port,invite
context=from-internal
disallow=all
allow=ulaw
```



Step2) you should define your outgoing calls to cisco

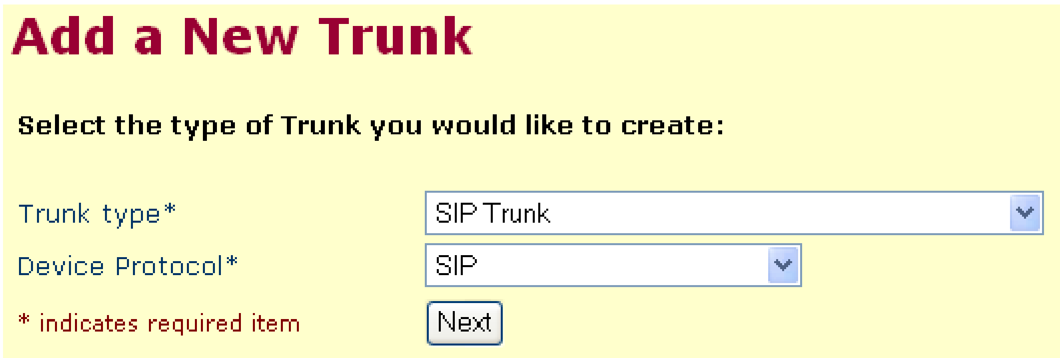
Step3)

1. Open up the CallManager Administration web page.
2. Since a SIP trunk requires MTP, make sure you have one:
    1. Service -> Media Resource -> Media Termination Point
    2. Normally your CallManager server should appear there if you do an empty query
    3. if not, go to the CallManager Serviceabilty web page, and activate the Cisco IP Voice Media Streaming App service
3. Select Device->Trunk from the menu.



Select the "Add a New Trunk" link from the upper right hand corner of the "Find and List Trunks" page.

Select "SIP Trunk" as the "Trunk type" and "SIP" as the "Device Protocol". Click on the "Next" butto



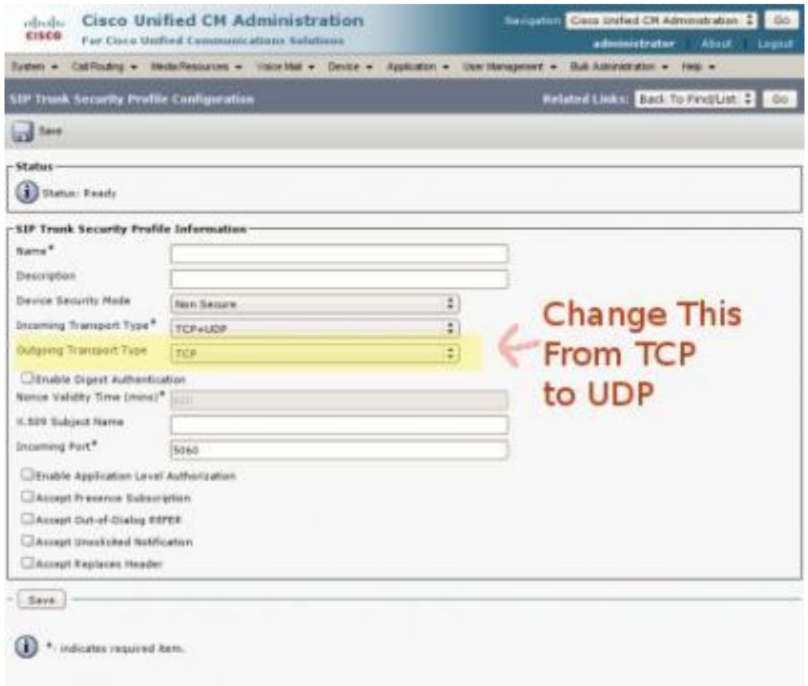
1. Enter a name in the "Device Name". Valid characters are letters, numbers, dashes, dots (periods), and underscores. The device name is only used internally in Call Manager so it can be anything you want.
2. Enter a description in the "Description" field.
3. Select a device pool.
4. Enter the IP address of your Asterisk server in the "Destination Address" field.
5. Select "UDP" as the "Outgoing Transport Type".
6. Modify any other settings as needed for your (CiscoCallManager|CallManager) installation.
7. Click on the "Insert" button.
8. Add route patterns in CallManager that send calls to Asterisk using the SIP trunk that you just created.

Cisco Call Manager 6.1

It' very similar to 4.1, but you must change the UDP protocol of sip in this menu:

System > Security Profile > SIP Trunk Security Profile

Outgoing Transport Type: UDP



Source: http://www.voip-info.org/wiki/view/Asterisk+Cisco+CallManager+Integration

12.10 Customized chanspy

Here is a customized chanspy. It first asks for the password and then the extension number you want to monitor.

Add this custom extension in /etc/asterisk/extension_custom.conf

```
[from-internal-custom]
exten => 1234,1,Playback(demo-congrats)     ; extensions can
dial 1234
exten => 1234,2,Hangup()
exten => h,1,Hangup()
include => agentlogin
include => conferences
include => calendar-event
include => weather-wakeup
```

```
include => chanspy

[chanspy]

exten = 556,1,Authenticate(1234)

exten = 556,2,Read(SPYNUM,extension)

exten = 556,3,ChanSpy(SIP/${SPYNUM},q)

; end of [app-chanspy]
```

12.11 Tips and tricks of reading log files

Perhaps the most important skill required to troubleshoot is reading the log files. Like other Linux services, asterisk writes the logs in /var/log/asterisk

To enable a full log you should edit the /etc/asterisk/logger.conf and uncomment the full line. Afterwards you must run this command to apply changes:

asterisk -rx "logger reload"

Here are examples of tracing in logs:

```
cat /var/log/asterisk/full | grep ERROR

cat /var/log/asterisk/full.1 | grep 9878 | grep "some text"

cat /var/log/asterisk/full|grep "user-callerid: "|grep 9899 >
/var/log/asterisk/9899.log

cat /var/log/asterisk/full.1| grep "[22435]" > 9864.log

tail /var/log/asterisk/full | tee log_exmpl.txt
```

13. Troubleshooting and Maintenance

3.1 Heavy Asterisk Log

Over time Asterisk log can become so heavy and space consuming, not to mention the time you have to spend reading through tons of lines of log files. The way to organize the log files is to rotate them using **logrotate** facility.

Asterisk has 5 different main log files:

1. /var/logs/asterisk/messages
2. /var/logs/asterisk/queue_log
3. /var/logs/asterisk/event_log
4. /var/logs/asterisk/debug
5. /var/logs/asterisk/full

Here are some parameters description:

- **missingok**: If the log file is missing, go on to the next log file without issuing an error message.
- **copytruncate**: Truncate the original log file to zero size in place after creating a copy, instead of moving the old log file and optionally creating a new one
- **rotate 7**: Log files are rotated 7 times before being removed or mailed to the address specified in a mail directive. If count is 0, old versions are removed rather then rotated.
- **compress**: Old versions of log files are compressed with gzip to save disk space.
- **size:** Rotate only if the log file reaches the specified limit
- **notifempty**: Do not rotate the log if it is empty
- **sharedscripts**
  postrotate
  /etc/init.d/lighttpd reload
  endscript: The lines between postrotate and endscript (both of which must appear on lines by themselves) are executed after the log file is rotated. These directives may only appear inside a log file definition. In our case we are reloading lighttpd. Other opting could be send HUP single using kill command.

There are different ways you can configure the rotation:

- Sample 1

```
/var/log/asterisk/debug /var/log/asterisk/event_log
/var/log/asterisk
/messages {
weekly
missingok
rotate 9
size 2000k
copytruncate
endscript
}
```

- Sample 2

```
/var/log/asterisk/cdr-csv/Master.csv /var/log/asterisk/debug
/var/
log/asterisk/event_log /var/log/asterisk/messages {
weekly
missingok
rotate 4
sharedscripts
postrotate
/usr/sbin/invoke-rc.d asterisk logger-reload
endscript
}
```

- Sample 3

```
# cat /etc/logrotate.d/asterisk

/var/log/asterisk/queue_log {
 missingok
 rotate 7
 notifempty
 daily
 create 0640 asterisk asterisk
 postrotate
 /usr/sbin/asterisk -rx 'logger reload' > /dev/null 2>&1
 endscript
}

/var/log/asterisk/event_log {
 missingok
 rotate 7
```

```
 notifempty
 daily
 create 0640 asterisk asterisk
 postrotate
 /usr/sbin/asterisk -rx 'logger reload' > /dev/null 2>&1
 endscript
}

/var/log/asterisk/full {
 missingok
 rotate 7
 notifempty
 daily
 create 0640 asterisk asterisk
 postrotate
 /usr/sbin/asterisk -rx 'logger reload' > /dev/null 2>&1
 endscript
}

/var/log/asterisk/messages {
 missingok
 rotate 7
 notifempty
 daily
 create 0640 asterisk asterisk
 postrotate
 /usr/sbin/asterisk -rx 'logger reload' > /dev/null 2>&1
 endscript
}
```

3.2 How to tackle Jitter issue

In an environment with old LAN cabling, chances are so high that call get interrupted, words drop every now and then, and at the worst case, the sound quality gets terribly unbearable. This is mainly caused by the issue commonly referred to as Jitter.

In voice over IP (VoIP), jitter is the variation in the time between packets arriving, caused by network congestion, timing drift, or route changes. A jitter buffer can be used to handle jitter.
But what exactly does jitter buffer do?

It trades a slight bit of latency to compensate for jitter, allowing a sort amount of time for late packets to be added into the stream that would otherwise have been dropped.

To fix the jitter issue, we must first get to know where we have jitter, that is, SIP <-> SIP or SIP <-> PSTN

In case of SIP <-> SIP do the following:

1) edit /etc/asterisk/sip_general_custom.conf
2) add the following:

```
jbenable=yes

jbforce=yes

jbmaxsize=20
```

3) reload SIP in asterisk:

```
ippbx*CLI> sip reload
```

Here are more parameters involved with jitter buffer:

```
;Enable Jitter Settings
jitterbuffers=4
jbenable=yes
jbforce=no
jbimpl=fixed
;jbimpl=adaptive
jbmaxsize=500
jbresyncthreshold=100
jblog=yes
;tos=0x18
```

3.3 Echo on POTS lines

Using **fxotune** will help echo canceller to work better.

Do this before setting up echo canceller, it means to disable echo canceller options in /etc/dahdi/system.conf and

/etc/asterisk/chan_dahdi.conf, and to stop asterisk but dahdi still running.

follow the following step by step:
1. Stop asterisk: /etc/init.d/amportal stop
2. Stop dahdi: /etc/init.d/dahdi stop
3. Remove echo related options to disable echo canceller:
3.1. edit /etc/dahdi/system.conf and set echocanceller=none for all channels

by default it's like this: echocanceller=oslec,1
change to this:

```
fxsks=1
echocanceller=none
fxsks=2
echocanceller=none
…
```

3.2. edit /etc/asterisk/chan_dahdi.conf and set the echo parameters like this:

;echotraining=800 ; disabled when using OSLEC
echocancel=yes ; yes=128. you can use any power of 2 such as 128,256,512,1024
echocancelwhenbridged=no

4. Start dahdi: /etc/init.d/dahdi start
5. Plug the line to FXO
6. To get echoes information on FXO lines run # **fxotune -i 4** (use 4, its any number, but 4 is on all over documents)
7. Once done (2-3 minutes per line/per FXO port), take a look for /etc/fxotune.conf and read the content, it should have some sort of configurations or options
8. To apply tuning run: # **fxotune -s** (this will be automatically executed by dahdi init, so you dont need to do this everytime)
9. Enable echo canceller options (do not enable on dahdi if you're using hardware echo canceller, only echocancel=yes)

10. Run asterisk, normal execution, reboot the server or just run /etc/init.d/amportal start

3.4 Essential Linux commands

Here is a list of some must-know commands in linux with examples:

- **grep** Command

Option -v, will display all the lines except the match. In the example below,
it displays all the records from /etc/password that doesn't match John.
Note: There are several lines in the /etc/password that doesn't contain the
word John. Only the first line of the output is shown below.

```
# grep -v John /etc/passwd
```

jbourne:x:1084:1084:Jason Bourne:/home/jbourne:/bin/bash

Option -c shows the number of lines that the pattern matches.

```
# grep -c John /etc/passwd
```

2

You can also get the total number of lines that did not match the specific pattern by passing option -cv.

```
# grep -cv John /etc/passwd
```

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Use option -r (recursive) to search all subdirectories for a text matching a specific pattern. In the example below, it will search for the text "John" by ignoring the case inside all the subdirectories under /home/users.
This will display the output in the format of "filename: line that matching the
pattern". You can also pass the option -l, which will display only the

name of
the file that matches the pattern.

```
# grep -ri john /home/users
```

/home/users/subdir1/letter.txt:John, Thanks for your

contribution.

```
/home/users/name_list.txt:John Smith
```

www.thegeekstuff.com

/home/users/name_list.txt:John Doe

```
# grep -ril john /root
```

/home/users/subdir1/letter.txt

/home/users/name_list.txt

- **find** Command

The following command looks for all the files under /etc directory with mail
in the filename.

```
# find /etc -name "*mail*"
```

The following command will list all the files in the system greater than 100MB.

```
# find / -type f -size +100M
```

The following command will list all the files that were modified more than 60
days ago under the current directory.

```
# find . -mtime +60
```

The following command will list all the files that were modified in the last two days under the current directory.

```
# find . –mtime -2
```

How to delete all the archive files with extension *.tar.gz and
greater than 100MB?
Please be careful while executing the following command as you don't
want
to delete the files by mistake. The best practice is to execute the same
command with ls –l to make sure you know which files will get deleted
when
you execute the command with rm.

```
# find / -type f -name *.tar.gz -size +100M -exec ls -l {} ;

# find / -type f -name *.tar.gz -size +100M -exec rm -f {} ;
```

The following command finds all the files not modified in the last 60 days
under /home/jsmith directory and creates an archive files under /tmp in
the
format of ddmmyyyy_archive.tar.

```
# find /home/jsmith -type f -mtime +60 | xargs tar -cvf

/tmp/`date '+%d%m%Y'_archive.tar`
```

- **xargs** Command

xargs is a very powerful command that takes output of a command and
pass it
as argument of another command. Following are some practical
examples on
how to use xargs effectively.
a. When you are trying to delete too many files using rm, you may get
error
message: /bin/rm Argument list too long – Linux. Use xargs to avoid this
problem.

```
# find ~ -name '*.log' -print0 | xargs -0 rm -f
```

b. Get a list of all the *.conf file under /etc/. There are different ways to get
the same result. Following example is only to demonstrate the use of xargs.
The output of the find command in this example is passed to the ls –l one by
one using xargs.

```
# find /etc -name "*.conf" | xargs ls –l
```

c. If you have a file with list of URLs that you would like to download, you can
use xargs as shown below.

```
# cat url-list.txt | xargs wget –c
```

d. Find out all the jpg images and archive it.

```
# find / -name *.jpg -type f -print | xargs tar -cvzf
images.tar.gz
```

e. Copy all the images to an external hard-drive.

```
# ls *.jpg | xargs -n1 -i cp {} /external-hard-
drive/directory
```

3.5 job scheduling in linux using crontab

I think if you wanna be a Elastix Professional you should know Linux abilities as best, one of these abilities is cron job.

crontab allows tasks to be automatically run in the background at regular intervals. You could also use it to automatically create backups, synchronize files, schedule updates, and much more.

run :

```
sudo crontab -e
```

This will open the crontab using the default editor.

Here is the format to add cronjobs:

```
* * * * * /bin/execute/this/script.sh
```

As you can see there are 5 stars. The stars represent different date parts in the following order:

minute (from 0 to 59)

hour (from 0 to 23)

day of month (from 1 to 31)

month (from 1 to 12)

day of week (from 0 to 6) (0=Sunday)

- **example1**

So if we want to schedule the script to run at 1AM every Friday, we would need the following cronjob:

```
0 1 * * 5 /bin/execute/this/script.sh
```

- **example2**

if we want to schedule the script to Monday till Friday at 1 AM, we would need the following cronjob:

```
0 1 * * 1-5 /bin/execute/this/script.sh
```

- **example3**

Execute 10 past after every hour on the 1st of every month

```
10 * 1 * * /bin/execute/this/script.sh
```

- **example4**

What if you'd want to run something every 10 minutes? Well you could do this:

```
0,10,20,30,40,50 * * * * /bin/execute/this/script.sh
```

But crontab allows you to do this as well:

```
*/10 * * * * /bin/execute/this/script.sh
```

- **example 5**
  reboot the system every day at 3 am

```
0 3 * * * /sbin/reboot
```

- **example 6**
  Using the **@reboot** cron keyword, this will execute the specified command once after the machine got booted every time.

@reboot CMD