IP 01/02/04/08

USER MANUAL

Version History

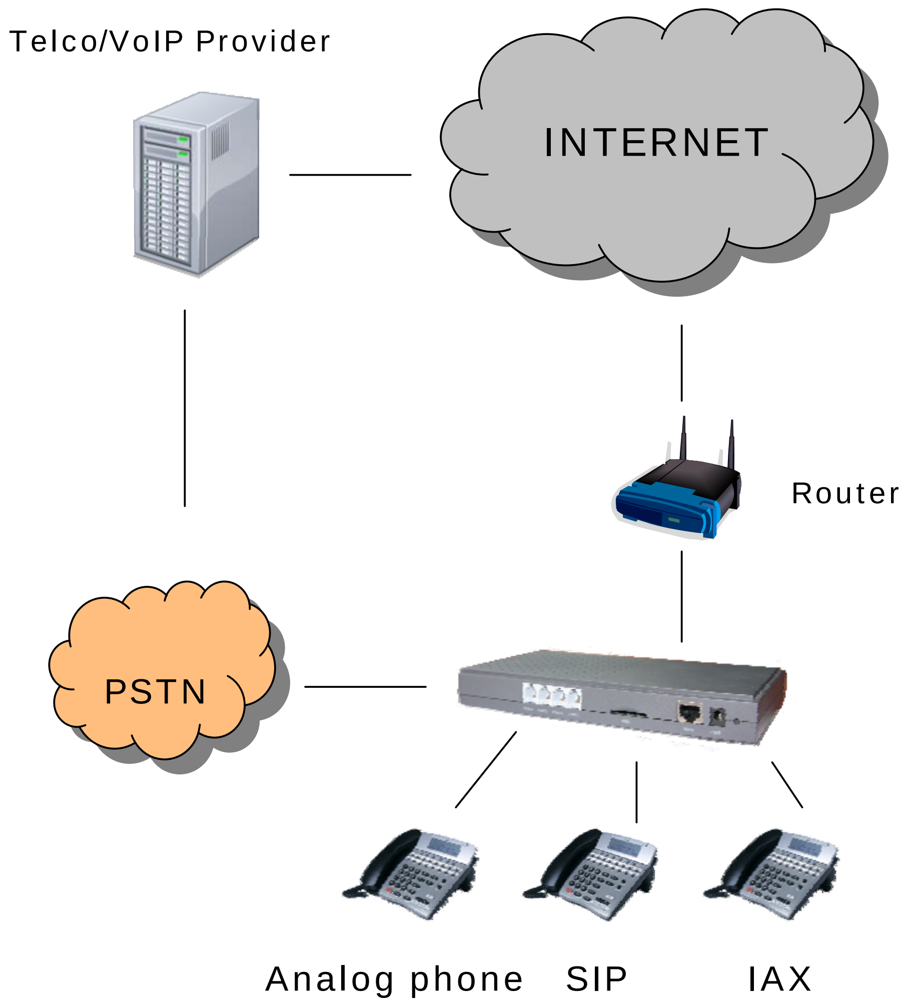
Version	Description of change
0.5	Origin
0.6	Section 4. “How to” guide added
0.7	Section 2.3.1. for interface cables added
0.8	Change all Astfin references to Switchfin

Table of Contents

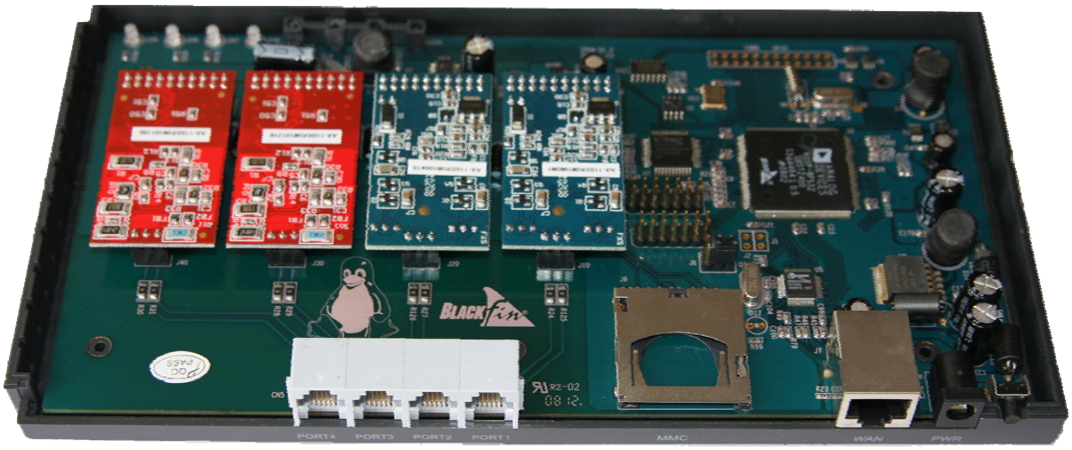
1. General overview

What is IP01/02/04/08 PBX?

IP01/02/04/08 is a Blackfin based, **Asterisk** embedded appliance. It provides one, two, four or eight analog telephony ports respectively. It is an **open source** platform based on **Switchfin**.



2. Technical information



*picture of IP04 with 2 FXO, 2 FXS modules installed

2.1. System: **Switchfin**

Devices manufactured by uCpbx are driven by our own telephony oriented, Open Source uClinux distribution called **Switchfin (http://www.switchfin.org).** Current version of Switchfin provides **Asterisk 1.4.x**, **Zaptel 1.4.x** and **Libpri 1.4.x** together with custom kernel modules to support our hardware. Additional applications such as PPPoE, SMTP forwarder, NTPd and many more are also provided to extend usability of the IP01/02/04/08 under different scenarios.
The IP0x PBX can accept x analog modules in total either FXS or FXO. The IP01/02/04 use port FXO/FXS modules, the IP02/08 support dual port FXO/FXS as well.

- FXS (Foreign eXchange Station) is an interface which drives a telephone or FAX machine. FXS interfaces get phones plugged into them, delivery battery, and provide ringing. FXS interfaces are signalled with FXO signalling.

- FXO (Foreign eXchange Office) is an interface that connects to a phone line. They supply your PBX with access to the public telephone network. FXO interfaces use FXS signalling. FXS interfaces are what allow you to hook telephones to your PBX, and FXO interfaces allow you to connect your PBX to real analog phone lines.

- IP02/04/08 support single GSM trunking module

2.2. Hardware:

- ADSP - BF532 400MHz CPU. DSP core for the media processing.
- 64MB of SDRAM
- 256KB serial flash for the boot-loader
- 256MB NAND flash for voicemail and prompts.
- SD card interface on a dedicated bus.
- Watchdog timer
- High Performance software echo cancellation

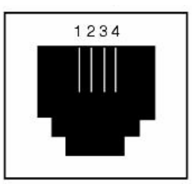
2.3. Interfaces

- FXO – optional
- FXS – optional
- GSM – optional
- 10/100Mbps Ethernet port with high performance PHY (IP02 and IP08 have two Ethernet ports)
- RS232 for console connectivity (115k, 8-N-1)

2.3.1. Interface cables

- For the Ethernet connection you have to use:
- In case you connect to router, switch and etc. Ethernet patch cable
- In case you connect to other PBX device crossover cable
- For the analog ports of your IP0x PBX you have to use RJ11 telephone line cable

The configuration of RJ11 port is



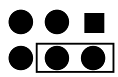
Pin	Description
2	Tip
3	Ring

- Serial Port cable

  For the IP0x PBX you have to use the PCB with RS232 interface. It is included in the kit.

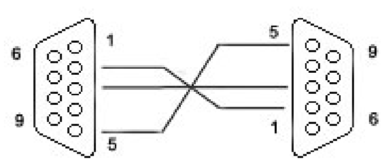
Remove the jumper shown on the picture below.

- **Warning – remember to put the jumper back after you finish using the serial connection. Otherwise your PBX will not boot. Proper place of the jumper is shown below.**



You have to plug in the RS-232 board into the small 6-pin header as shown below.

You need serial patch cable for console access to your PBX from a PC. The schematic of the cable is:



- For more information go to chapter 4.1. "How to access the IP PBX"

2.4. Applications

- VoIP / TDM Gateways
- PBX / IVR functionality
- VoIP Services
- Conferencing
- Custom platforms
- Voice Routing
- Custom Development

2.5. Additional information

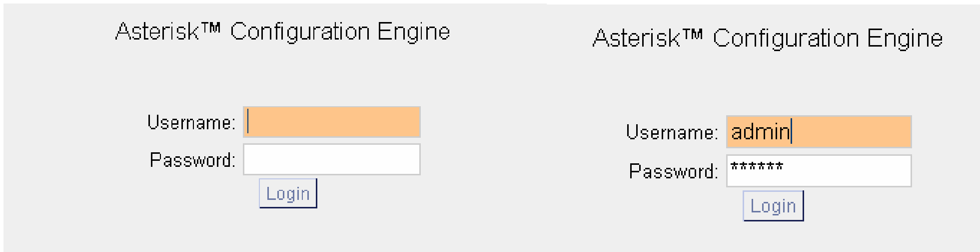
- Power supply 6 - 12VDC
- Current consumption – idle state 150 mA
- Dumensions: 220 x 130 x 40 mm
- 12V, 2A power adapter is included

3. Software and Configuration tips. Working with the GUI

3.1. System Status Menu

After all the interfaces are connected and your IP0x is powered up you can connect to the GUI through your preferred Web browser. By default all IP0x are preconfigured with 192.168.1.100 IP address. Please change the IP address on

the computer you will be using to configure the IP0x to be a part of 192.168.1.x /24 network (for example: 192.168.1.2).
At this point you can connect to the GUI by selecting the following URL: http://192.168.1.100
When the initial page finishes loading you will be prompted to authenticate. Our default user name is **admin** and password is **switchfin**



After successful login, you will see the system status page.

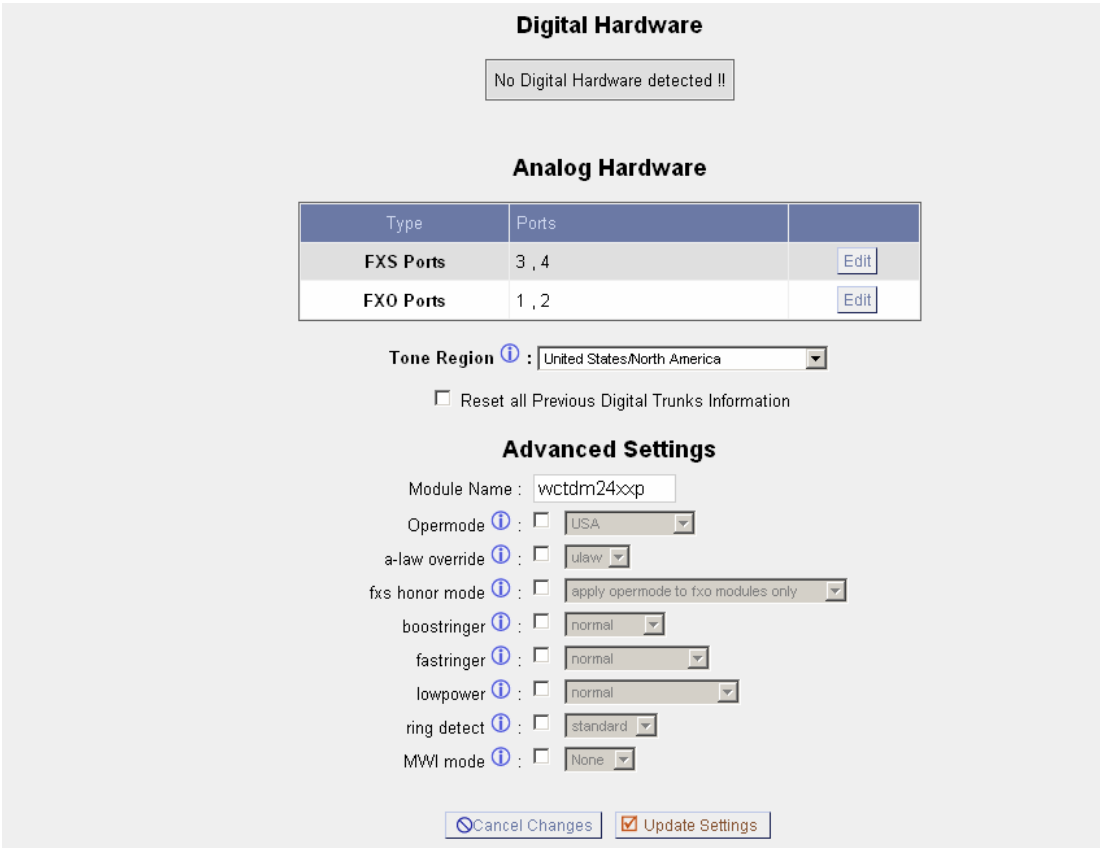
The System Status page presents all vital information about the current state of your IP01/02/04/08.
Here is an example with system status page of a local PBX.
As you can see there is one trunk "Local" in use, the type of it is analog and it is using port 1.
The users with extensions 6204, 6210 and 6250 are agents for a call center.
There is Conference room with public extension 6300 and no participants are connected.
Different color lights indicate the state of the users.
The extension 6200 associated with user name "John Brown", using an analog telephone and currently have one voicemail message. The current state of this user is "Ringing", which indicates that the endpoint is powered and it is ringing (receiving a call).
The extension 6204 associated with user name "Mike Reverouzzi", using an analog telephone and currently have two already checked voicemail messages. The current state of this user is "Ringing", which indicates that the endpoint is powered and it is ringing (receiving a call).
The extension 6210 associated with user name "Sales Department Secretary", using an SIP telephone and currently have no voicemail messages. The current state of this user is "Busy", which indicates that the endpoint is powered and it is on a call.
The extension 6211 associated with user name "Sales Department Head", using a IAX telephone and currently have no voicemail messages. The current state of this user is "UnAvailable", which indicates that the endpoint is powered down or there is some network connectivity problem.
The extension 6250 associated with user name "PR Department Head", using an SIP telephone and currently have four voicemail messages. The current state of this user is "Available". Which indicates that the endpoint is powered and not in use.
The extension 6260 associated with user name "Support Department", using an IAX and currently have no voicemail messages. The current state of this user is "Available". Which indicates that the endpoint is powered and not in use.
The extension 6400 is designated to Ring Group.
The extension 7000 is designated to the English language voice menu.
The extension 6221 is designated to retrieve voicemail messages.
The extension 2676 is designated to corporate directory service which can be used in one or more IVR menus.

- Additional information.

- Power supply – All IP PBXs come with power supply adapter included. It is 12V,2A and it can provide power to all PBX from IP02 to IP08.
- LED lights on the boxes – You have green light on when the PBX is powered up. There is green light that shows network connectivity.
The analog ports can be configured with FXS, FXO or GSM trunking module.

When on the port you have FXO module the light is red. When FXS module is installed the light is green, and when your analog phone is ringing you will have a single blink from green to red light.

3.2. Configure Hardware



- In this setup, you can see the Configure hardware menu of a IP0x PBX The detected analog hardware is two FXS and two FXO loaded.

3.3. Trunks

Click on the button in to main menu

If you want to setup VoIP trunk press the tab "New SIP/IAX Trunk"

***Reminder – You can always use the "i" (info) tooltips for additional information*

- First you have to select the type of your provider as shown SIP or IAX
- Provider name – specify your provider name for reference
- Hostname – here you type the default address of your provider for example support.provider.com
- Username – this is you username also given by the provider
- Fromuser – fill this field as per providers instructions

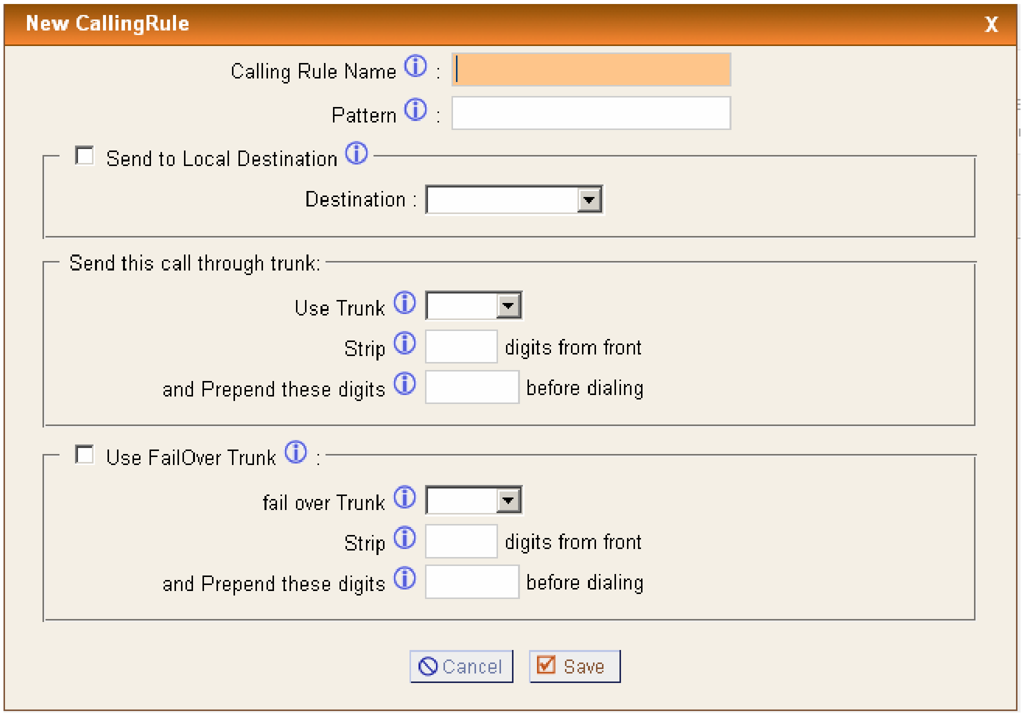
- Fromdomain – fill this field as per providers instructions
- Password – this is you password given by the provider
- Insecure Type – Specifies how to handle connections with peers

*** Warning –** In order for complete adding of new Trunk , after creating the trunk you must reboot your PBX from Reboot button in to the Option Menu. This should be done before doing any other changes in to the GUI.

3.4. Outgoing Calling Rules

Click on the button in to main menu

After loading this page on the system which has not been yet configured, you will need to create a "New Calling Rule" The following menu will appear:



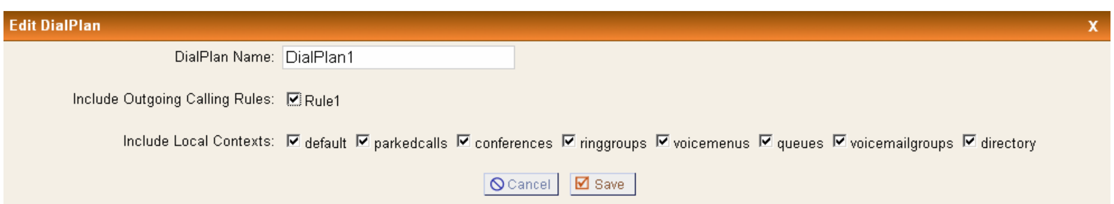
**Reminder – You can always use the "i" (info) tooltips for additional information*

- At first you need to assign unique reference name to the new Outgoing rule
- In the second field, "Pattern" standard telephony regular expression patters needs to be specified:
  X ... Any Digit from 0-9
  Z ... Any Digit from 1-9
  N ... Any Digit from 2-9
  [12345-9] ... Any Digit in the brackets (in this example, 1,2,3,4,5,6,7,8,9
  . ... Wildcard, matches anything remaining; i.e. _9011. Matches anything starting with 9011 (excluding 9011 itself)
  ! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible
  For example, the extension _NXXXXXX would match normal 7 digit numbers, while _1NXXNXXXXXX would represent a three digit area code plus phone number, proceeded by a one.
- "Destination" field should be used only if you want to invoke some local application or if the call should be processed in some special way.

- "Use trunk" indicates which trunk should be used to handle this call (ie: "bri2").
- "Strip" indicates how many proceeding digits should be removed from a dialstring.
- "Failover trunk" indicates if there is an alternative trunk to be used in case if the primary trunk is not available (in use for example).

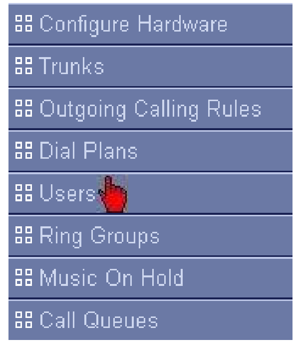
3.5. Dial plans

The "DialPlan" page allows you to define different Classes of Service (Dial Plans) and associate them with one or more Outgoing Calling Rules defined in the previous step. Also, predefined local services/application can be associated with each Class of Service.

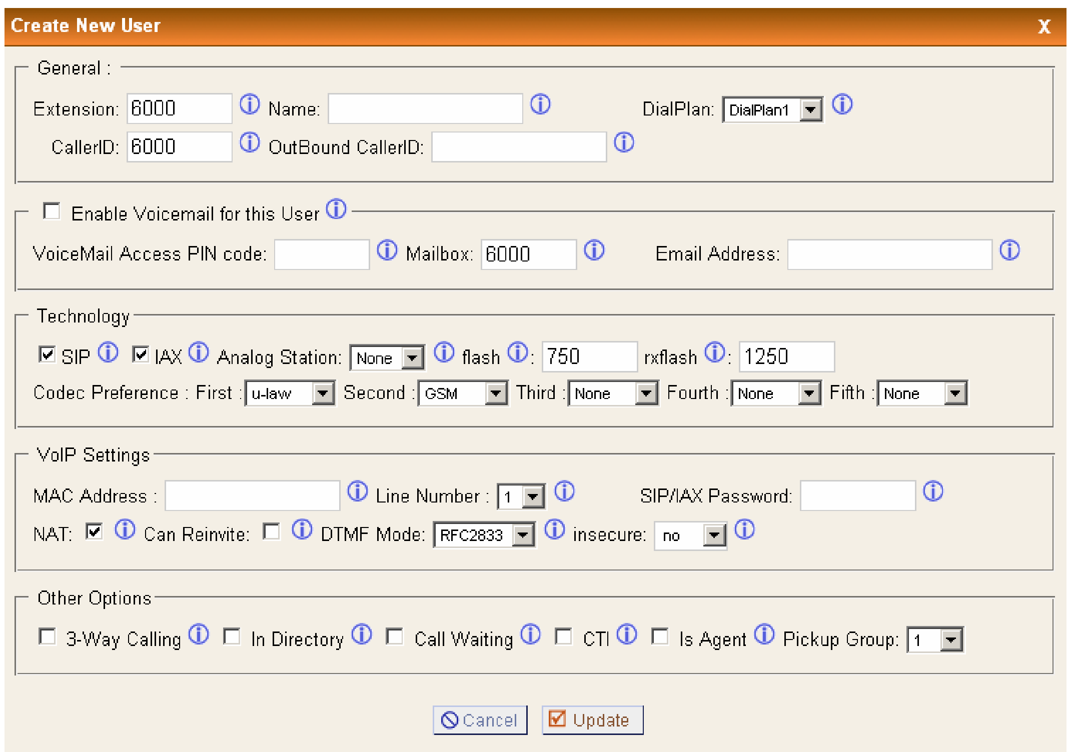


- Please select a unique identifier for your Dial Plan entry.
- Select all services and rules to be available to users using this Dial Plan entry (Class of Service).

3.6. Users



The “Users” page allows creation of user accounts (Extensions) for VoIP accounts. Voicemail options are also configurable through this page.



**Reminder – You can always use the “i” (info) tooltips for additional information*

General

- Extensions – this is the actual number to be dialed to reach the user, it’s an index and it needs to be unique.
- CallerID - specifies the internal caller id number associated with this account. This number will be used to automatically identify the user to the voicemail system. The CallerID does not have to be unique.
- Name – Indicates the Caller ID Name which will be sent to other callers , if the network permits such functionality.

- Outbound CallerID – indicates the public CallerID number to be used for the outbound calls. Depending on the provisioning of your ISDN line and/or feature set supported by your VoIP trunk providers, this number may or may not be presented to the destination endpoint.

Enable VoiceMail for this User

- VoiceMail Access Pin Code – indicates the passcode required to access this voicemail box.
- Mailbox – this field indicates to which mailbox number Voicemail Indicator will subscribe. (visual indicators and stutter tone)
- Email Address – indicates an email address to be used to send voicemail notifications and actual voicemail attachments (depends on voicemail configuration, please refer to the applicable section below).

Technology

- SIP/IAX/Analog Station - indicates the protocol (technology) to be used for this account.
- Flash/RxFlash – Hook flash specific parameters.
- Codec preference – list of available codecs listed in order of preference. Please note that both ends need to agree for connection to be established.

VoIP Settings

- MAC Address – this field is required for Polycom phone provisioning, do not use it at this point.
- Line number – as above
- SIP/IAX password – user password associated with this account
- NAT – check this if the device will be located behind NAT router in respect to our IP0x.
- Can Re-invite – indicates if SIP session can use re-invite to send RTP directly between to endpoints.
- DTMF Mode – DMTF method to be used for VoIP communication. TFC2833 would be the most common choice here.
- Insecure – method of authentication, both ends need to agree.

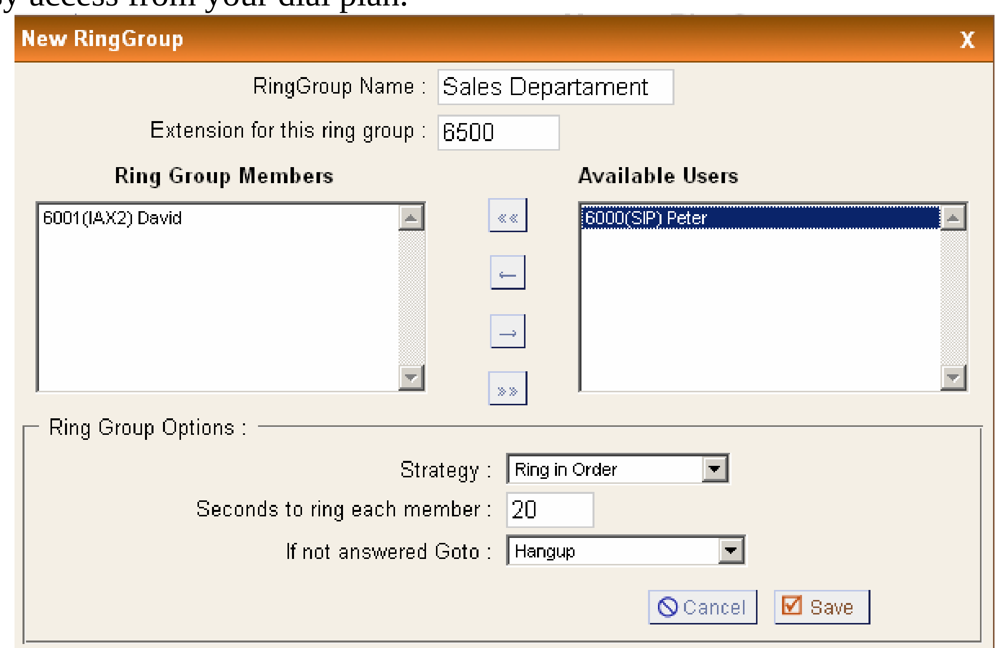
Other Options

- 3 Way Calling – enable/disable 3 way calling functionality, please make sure that your device is capable of handling it before selecting this option.

- In directory - indicates if this user account will be listed in a “directory listing” accessible from one or more IVRs.
- Call Waiting – enable/disable call waiting functionality on the account.
- CTI – Computer Telephony Integration, allows access to 3rd party applications over Asterisk Manager Interface.
- Is Agent – indicates if the user will be available in call queuing application.
- Pickup group – specified the call pickup group, if available

3.7. Ring Groups

This page allows to group several users (extensions) into one Ring Group. Unique extensions number will be associated with each Ring Group to allow easy access from your dial plan.



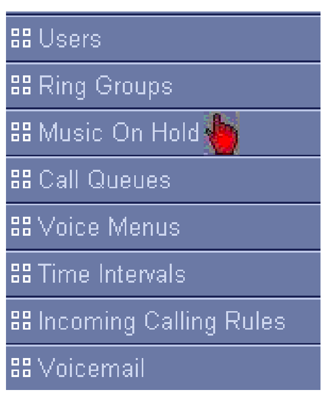
- At first you have to choose the name of the group (you can create several independent groups)
- Type the preferred extension for that group, in the example this is "6500"
- In the left window you can see the users that are members of the specific group, you can add/remove them with button in the middle. The available users are in the right window.

You have several options about the group calls.

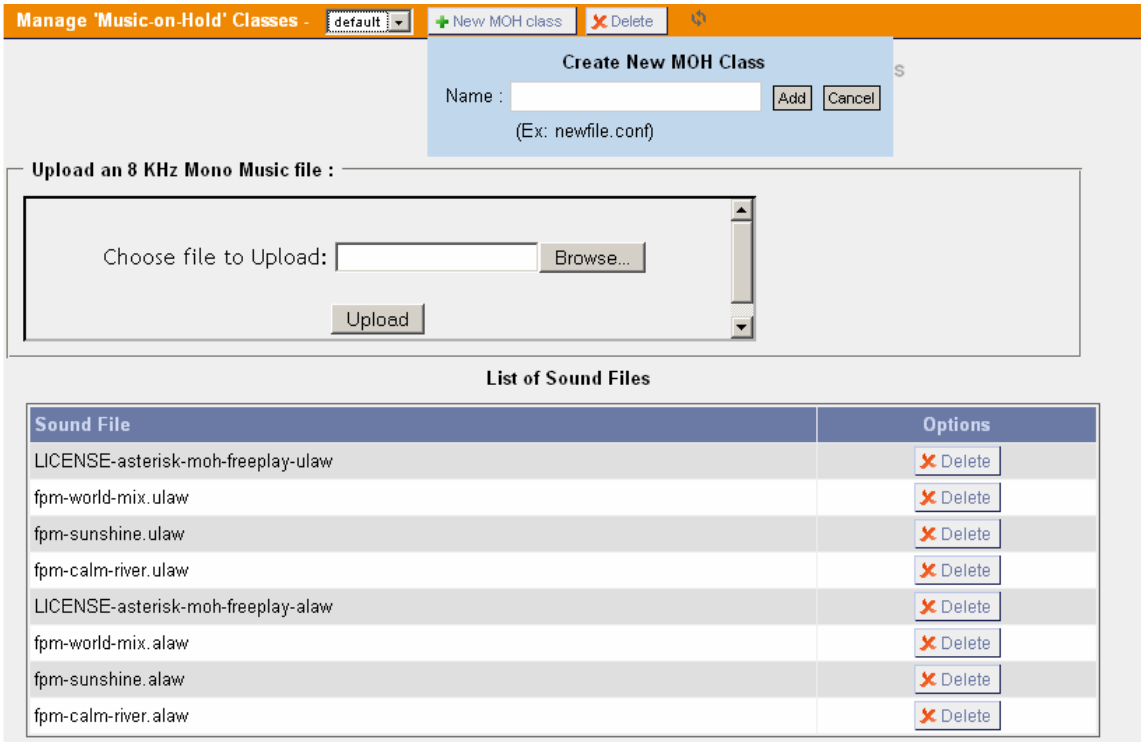
- Strategy, when you dial the extension for the group the phones can ring in order (left window) or in sequence for a predefined period of time. The other option is to ring all the devices in the group together at the same time.

- Handling for non answered calls: Call can be redirected to specific Voicemail, another extension, or another ring group.

3.8. Music on Hold

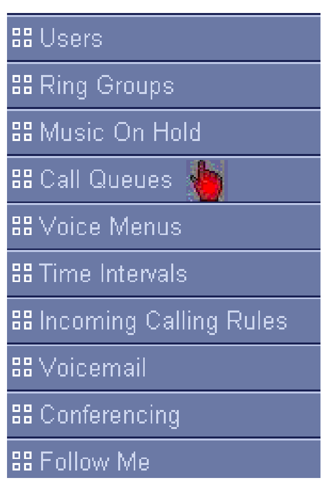


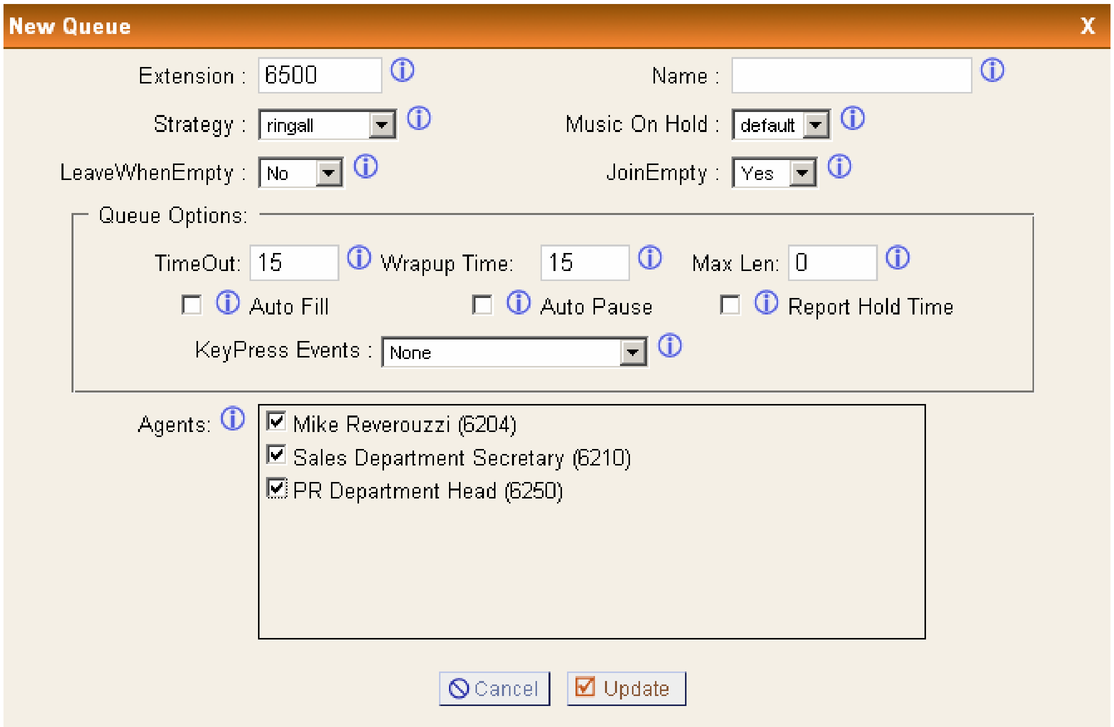
The default Music On Hold (MOH) class is shown below. Please note that you can not upload new MOH files at this point. Direct access to shell is required to scp or ftp new MOH files.



3.9. Call Queues

Call queues allows you to build a call center with one or many trunks.



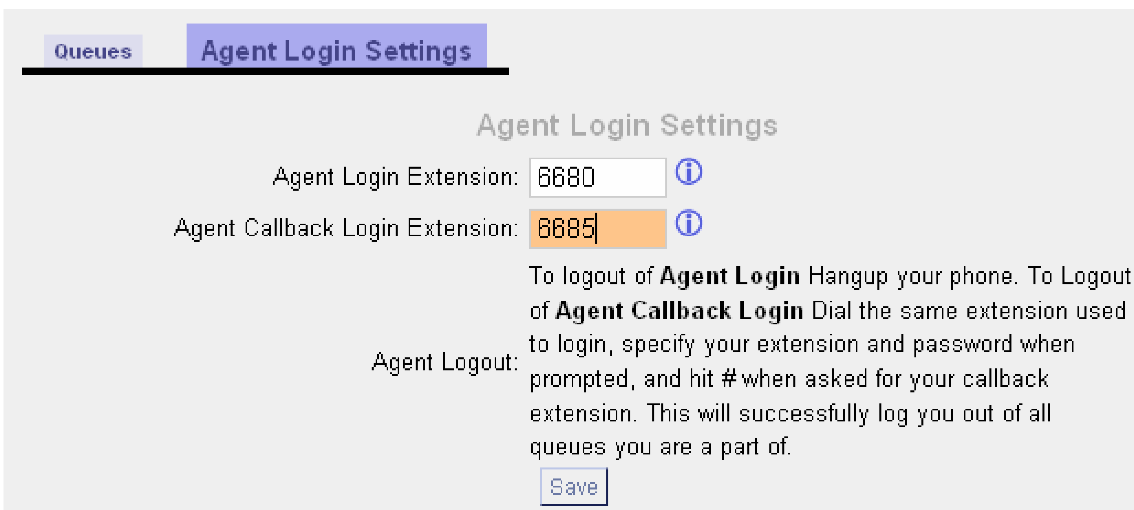


**Reminder – You can always use the "i" (info) tooltips for additional information*

- At first you need to assign unique extension to your new queue
- Then you need to name it, ie: "sales"
- Select the appropriate strategy for your application:
  - **Ringall** – ring all available agents at the same time.
  - **RoundRobin** – Each all agents in sequence.
  - **Leastrecent** – ring agent which was recently called.
  - **Random** – ring random agent.
  - **RRmemory** – RoundRobin with memory, where we left off last ring sequence.
- The appropriate Music on Hold class can be selected from the "Music on hold" menu.
- Leave when empty – this option controls state of users that are in the queue. If **yes** is selected, the callers are pushed out of the queue when no agents are logged in. If **no** is selected, callers will remain in the queue with no agents. If **Strict** is selected, callers are forced out of the queue if no agents are logged in, or if all logged in agents are unavailable.
- Join Empty - If **yes** is selected, callers can join a call queue with no agents or unavailable agents. If **no** is selected, the callers cannot join queue with no agents. If **strict** is selected, the callers cannot join queue with no agents or unavailable agents.
- Queue options –

- **Timeout –** how many seconds will ring an agent's phone before the queue tries to ring the next agent.
- **Wrap-up** time is how many seconds delay has an agent after completing a call, before another call is connect.
- **Max Len** is how many calls can be queued at once. This includes only calls that have not been yet connected.
- **Auto Fill** – when multiple calls are in the queue at the same time, to push them to agents simultaneously.
- **Auto Pause** – this option pauses an agent if they fail to answer a call.
- **Report Hold Time** – this option reports to the agent the hold time of the caller, before he is connected to an agent.
- **Key Press Events** – this setting selects which voice menu to be connected if a user waiting in the queue presses a button.

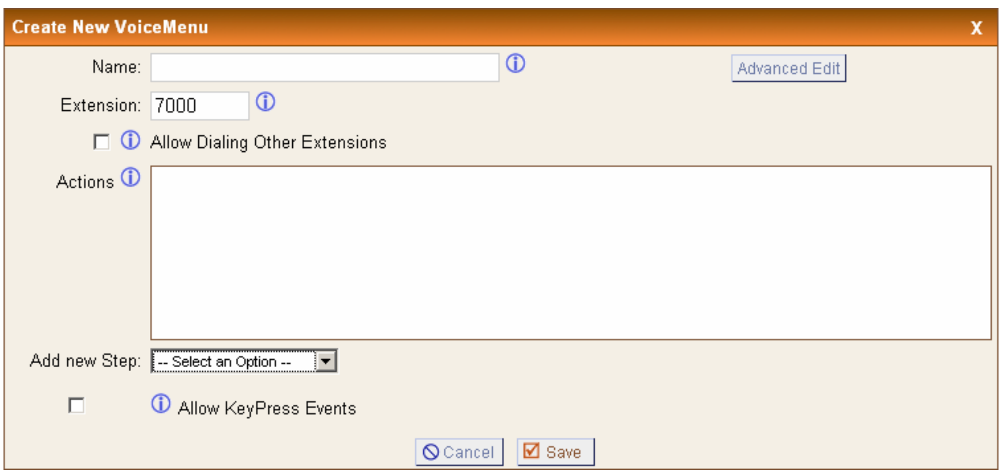
- List of available agents is available at the bottom.
- Agent Login Settings – you have to type the extensions for the agent login and agent callback login
-



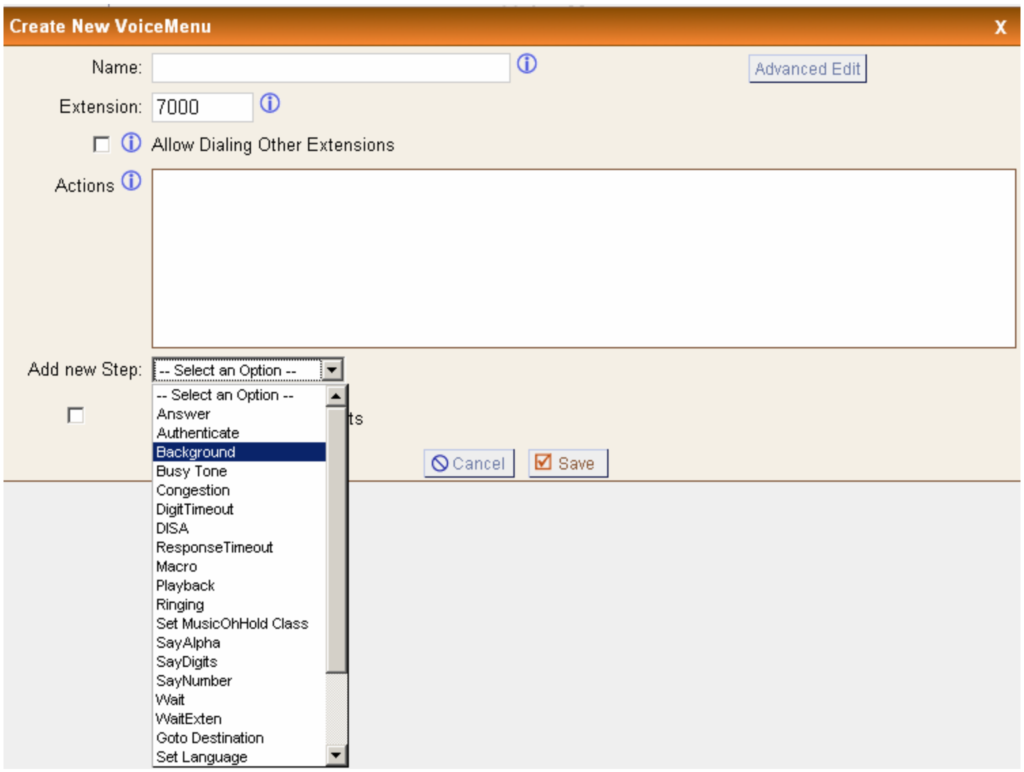
**Reminder – You can always use the "i" (info) tooltips for additional information*

3.10. Voice Menus

This page allows creation of custom voice menus for an IVR system. Please use the **Voice Menu Prompts** page to record your custom prompts/greeting before creating an IVR menu.

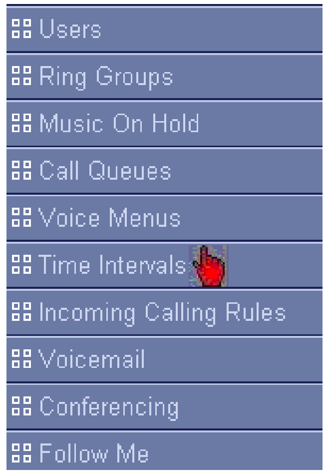


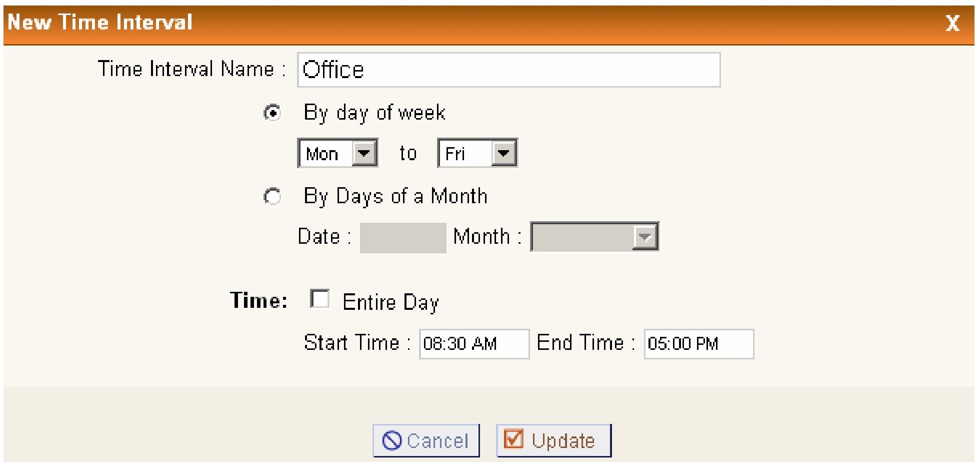
- Name – indicates a unique identifier assigned to your VoiceMenu. For example “Customer Service” or “Sales Office” .
- Extension - specifies an extension number to invoke your Voice Menu.
- Allow Dialing Other Extensions – controls if extensions which were not explicitly listed are also accessible
- Actions - list all defined steps for this Voice Menu.
- Add new step – List of all available options for Voice Menu. You have to choose and add them one by one.



- Allow Key Press Events – defines specific action to be executed upon pressing specific DTMF codes.

3.11. Time Intervals

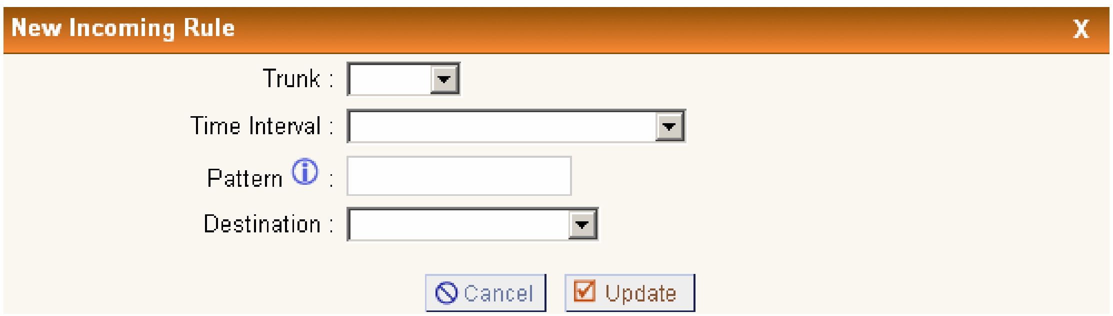




You can define here one more time intervals to allow different call handling during different time of day (week).
This application is very useful, when you want to answer calls only in specific time interval, of the day, week or month.
In the example here the setting has name "Office" and it is set for the regular business hours of the week. You can create several time intervals with different names and later access them under "Incoming Calling Rules" section.

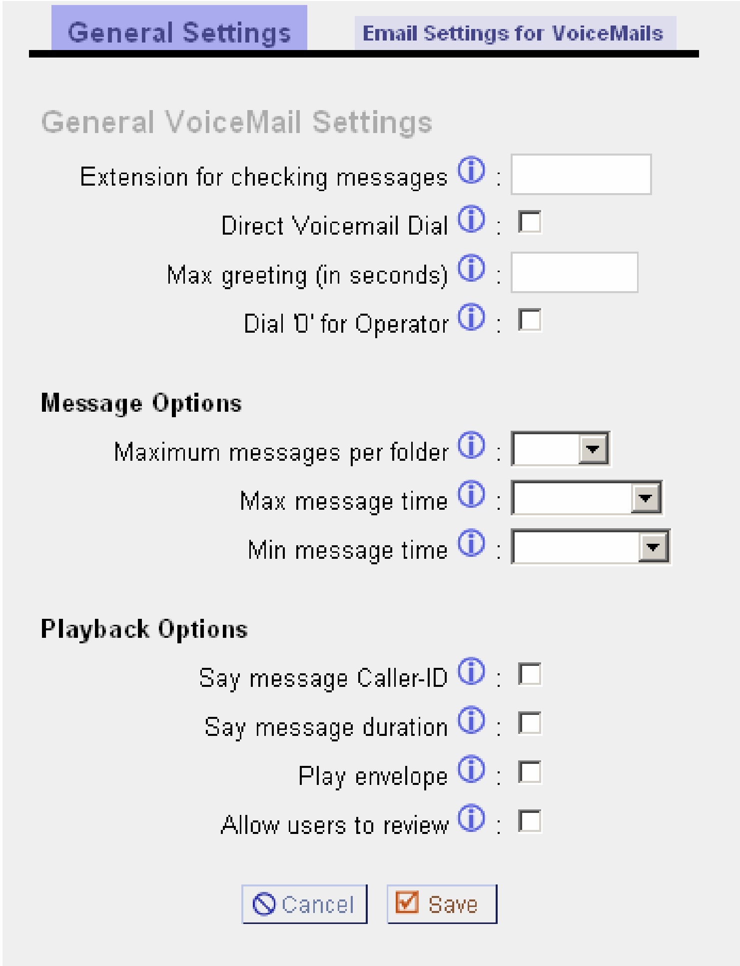
3.12. Incoming Calling Rules

This page defines how to handle incoming calls.



- Trunk – indicates incoming trunk to be handled, ie “bri2”.
- Time Interval – indicates time frame when this rule will apply
- Pattern – indicates NDIS number (dialed) number using the same patterns as listed in the “**<u>Outgoing Calling Rules</u>**” section.
- Destination – specifies an extension, IVR Menu or application which will terminate the incoming calls, providing all the rules above apply.

3.13. Voice mail

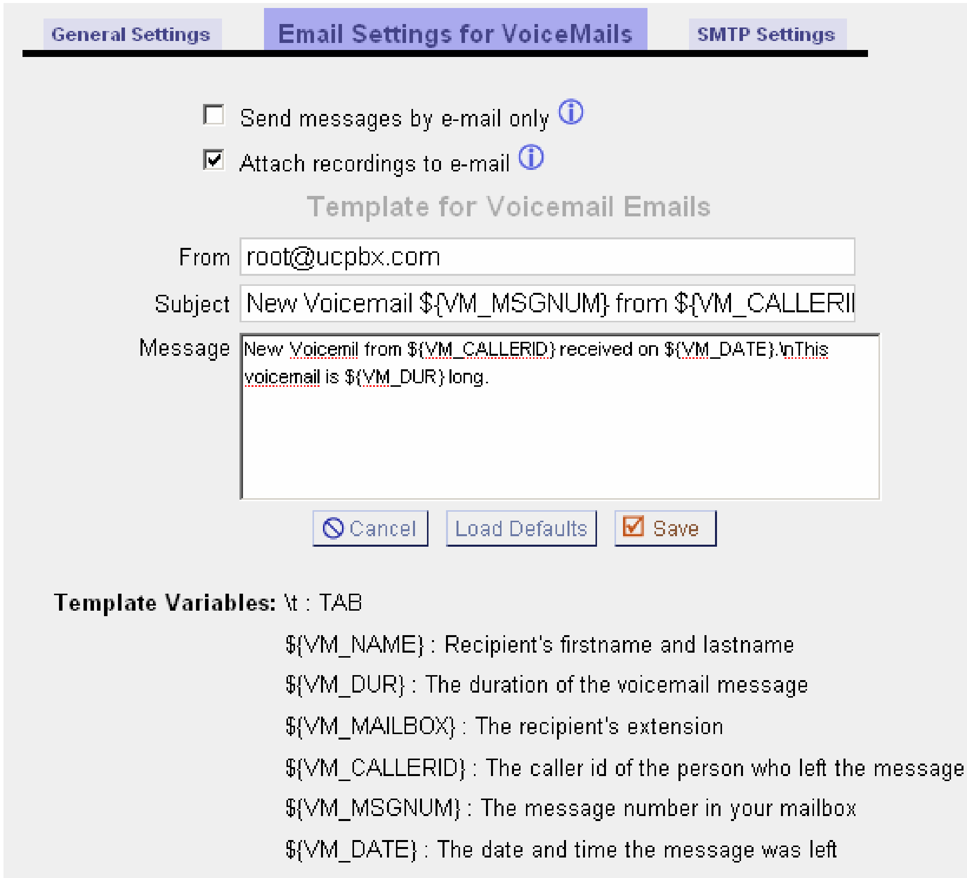


**Reminder – You can always use the "i" (info) tooltips for additional information*

The General Settings Menu:

- Extension for checking messages – you have to print the number, you want to use for checking your Voicemail. For example 111
- Direct Voice mail
- Max Greeting – is the length of the greeting in seconds
- Dial "0" for operator
- Message Options
- Maximum messages per folder – you have choice from 10 to 1000
- Max time – you have to select what is the maximum length of recorded message
- Main time – you have to select what is the minimum length of recorded message
- Playback Options - they are simple and very useful

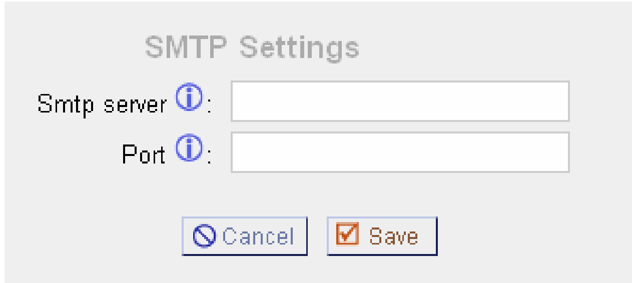
Email Settings for Voicemail:



**Reminder – You can always use the "i" (info) tooltips for additional information*

- Send messages – if this option is set, the voicemails will not be kept on the server. They will be sent directly to the e-mail.
- Attach recordings – this option defines whether or not to attach recordings to voicemail attachments. Note: You need SMTP server set for using this option.
- There are a couple options about the emails. You can set the PBX to send e-mail with information about the caller ID, message number, recipient's names and etc.

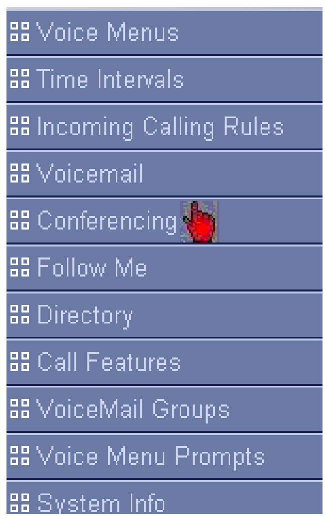
SMTP Settings



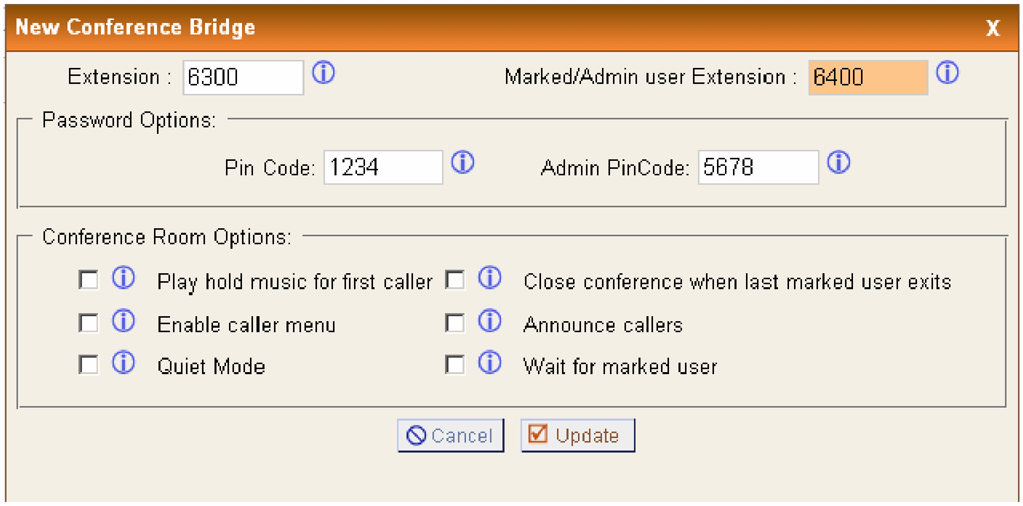
**Reminder – You can always use the "i" (info) tooltips for additional information*

- STMP server – it is the IP/host name of the outgoing mail server that your IP0X will connect and send e-mails with **voicemail** notifications
- Port – The port number on which the SMTP server is running

3.14. Conferencing



After loading this page on the system which has not been yet configured, you will need to create a "New Conferencing Bridge". The following menu will appear:



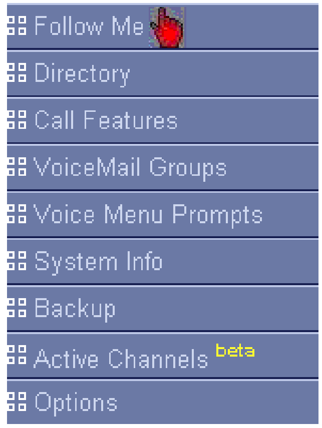
**Reminder – You can always use the "i" (info) tooltips for additional information*

The Conference Bridge can be used for creating a conference call with several participants. The bridge has public extension for all users and admin extension for the administrator (if applicable). Access to the conference call could be protected with PIN codes for both admin and public.

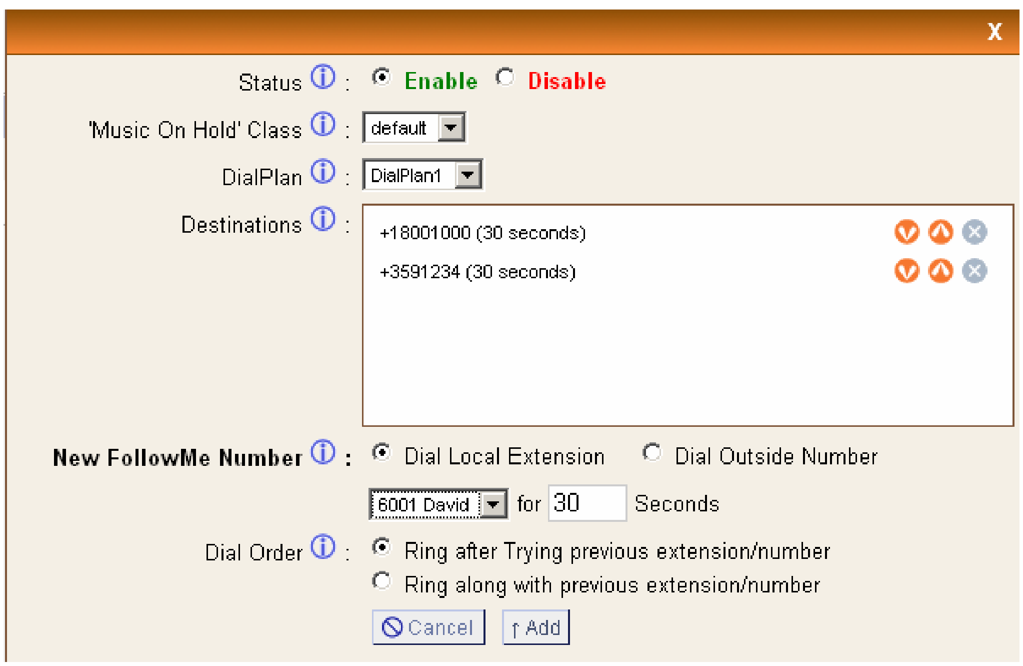
- Extension – public extension for entering the bridge
- Marked/Admin user Extension – administrative extension
- Pin Code – optional Pin codes associated with public and admin extensions.
- Conference room options
- Play hold music – this option causes Asterisk to play Hold Music to the first user in the conference, until another user joins the same conference
- Caller Menu – checking this option allows the user to access the Conference menu by pressing " * " on his numpad
- Quiet Mode – do not play enter/leave sounds
- Close conference when last marked user exits – when the last user logouts from the bridge, close it.
- Announce callers – when checked the participants of the bridge are announced when another participant is joining the bridge.
- Wait for marked users – prevent conference participants from hearing each other, until marked user has joined

Note that the conference can starts after entrance of the admin. If the admin is not available all users hear the default music on hold.

3.15. Follow me



- The follow me option allows to specify sophisticated call handling for selected extensions. Calls can be routed to other applications, local numbers and external numbers. By default "FollowMe" options are disabled for all users.

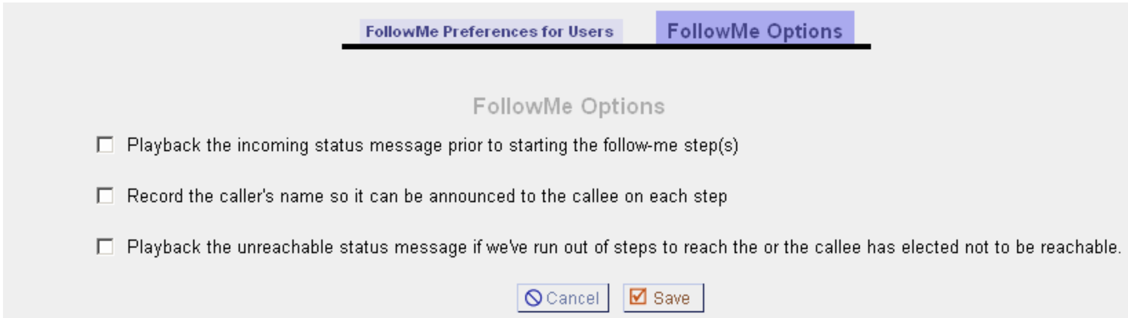


**Reminder – You can always use the "i" (info) tooltips for additional information*

- Status – Select to enable or disable it on per user basis.
- Music on hold class – select the appropriate MOH class.
- DialPlan – Select Class of Service to control access to outbound trunks.
- Destination – list of numbers in priority sequence

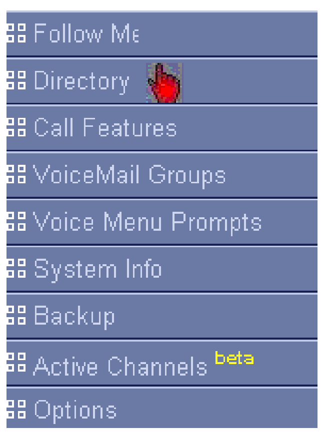
- Add Number – you can add local numbers or outside numbers.
- Dial Order – method of handling the call

For some cases you may find the additional follow me options useful.

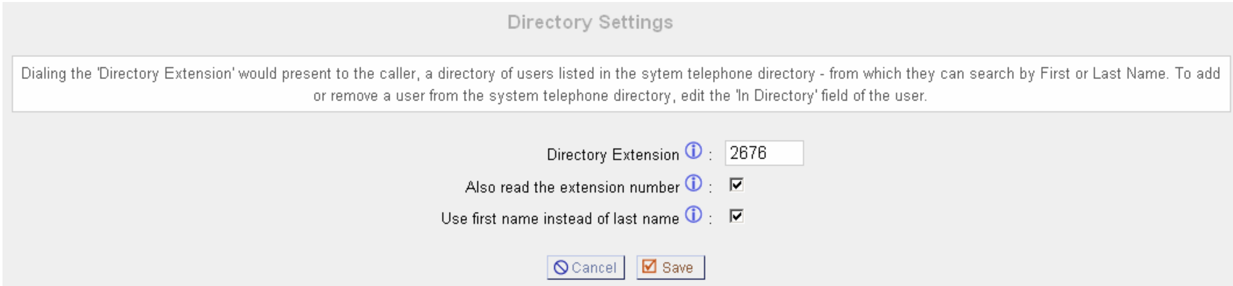


- Playback the incoming status messages prior to starting the follow-me step(s) – by checking this option there will be a massage for the caller, before starting the follow me steps
- Record the caller's name –
- Playback the unreachable status message – if this option is checked there will be status message for the caller when we're out of follow me steps or the callee wants to be not reachable.

3.16. Directory

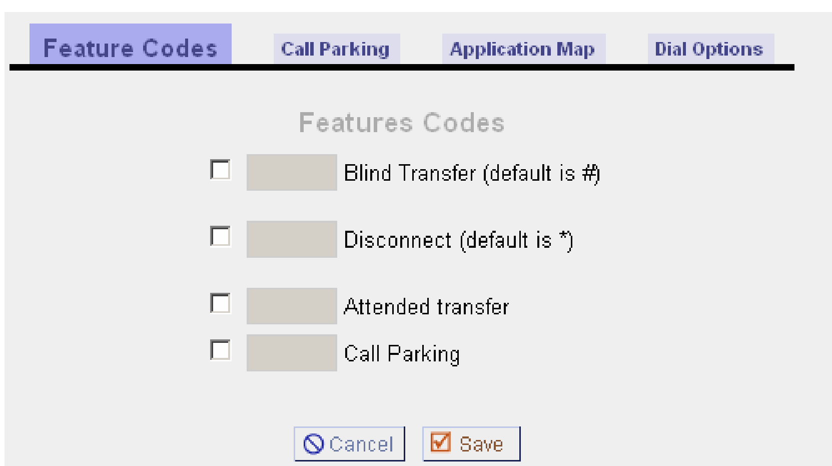


- Directory extension provides searchable list of users for IVR purposes. User will be listed in the Directory, if the field "In Directory" is selected under the "Users" settings window. The directory can be based on the users first or last name.

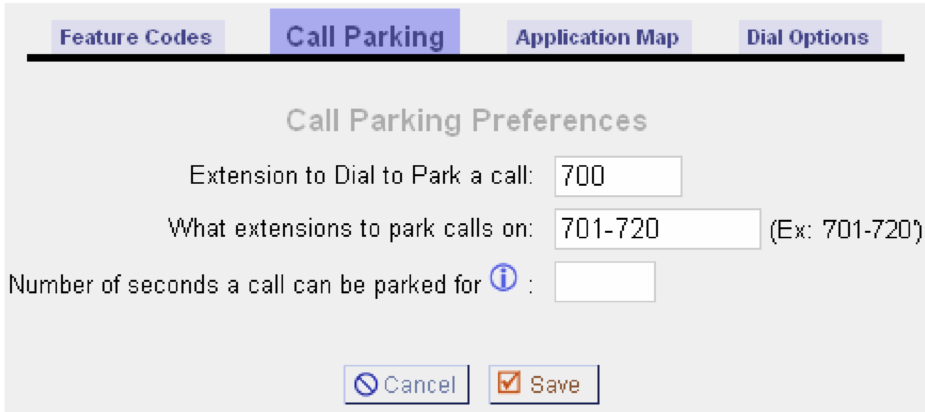


- Directory extension – extension to dial for accessing the name directory
- Also read extension – in addition to the name also read the extension number to the caller before presenting the dialing options
- Use first or last name – when this option is checked the caller is allowed to enter the first name in to the directory instead of using the last name.

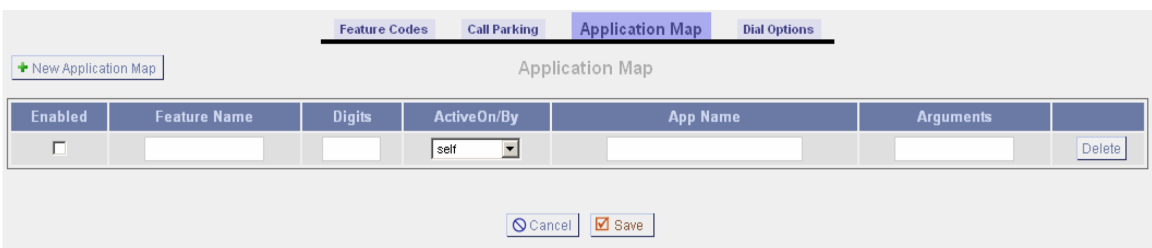
3.17. Call Features



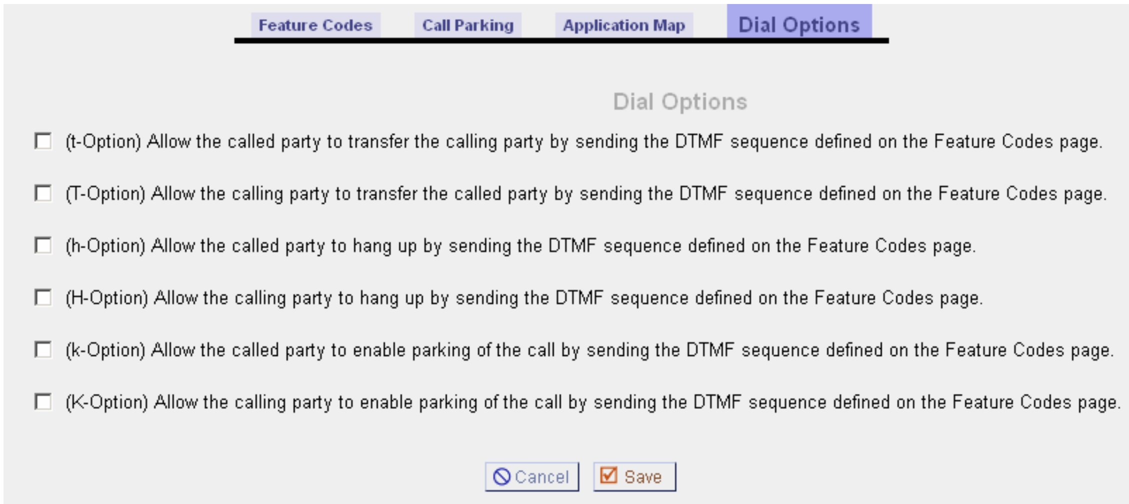
- Feature Codes section specifies DTMF sequence to answer specific services (features).



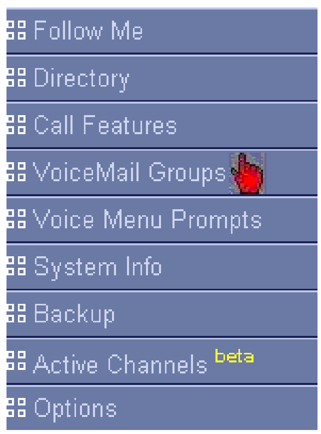
- Call Parking - section allows to configure functionality that allows the user to put a call on hold and pick it up at a different phone. Single call parking extensions can be defined here and a range of call parking "spots" where calls will be parked in sequence.



- Application Map section - allows definition of a key-sequence (DTMF digit sequence), an application and the party on which this application is executed when the sequence is pressed.
- Dial Options section allows to specify how the Feature Codes applies to the calling party and called party.



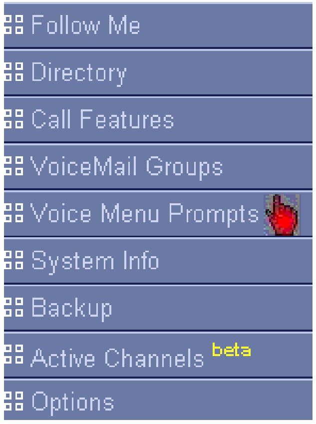
3.18. VoiceMail Groups



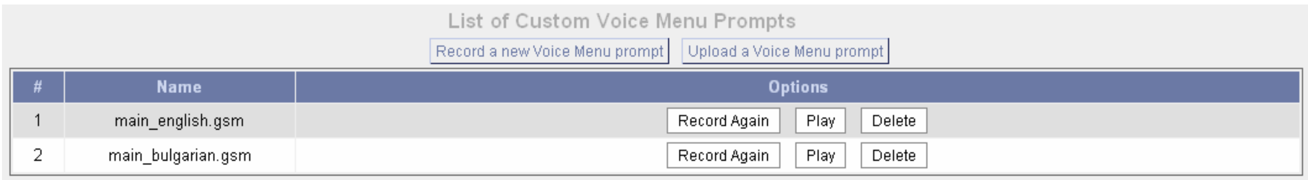
VoiceMail Groups allows to create a virtual mailbox allowing to distribute a message to several mailboxes at once. In the example below, mailboxes 6204, 6210 and 6250 are grouped under virtual mailbox 6600. Any messages recorded for mailbox number 6600 will appear in mailboxes 200, 204 and 207. This is useful to for teams such as **support** or **sales**.

- **Extension** to access the voicemail of the group.
- **Label** used for reference
- **MailBoxes** distribution list

3.19. Voice Menu Prompts



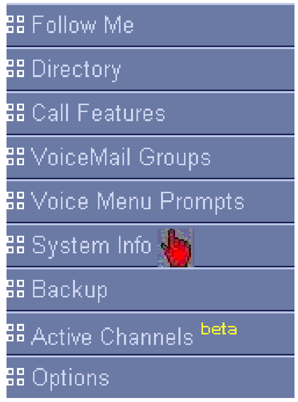
- You can record custom voice menu prompt for IVR purposes. This could be a greeting or instructions in foreign languages. You need to specify a valid extension to perform recording procedure. Please note that at this time we do not support uploads of your custom prompts through the GUI. Shell access and ftp, scp will need to be used.

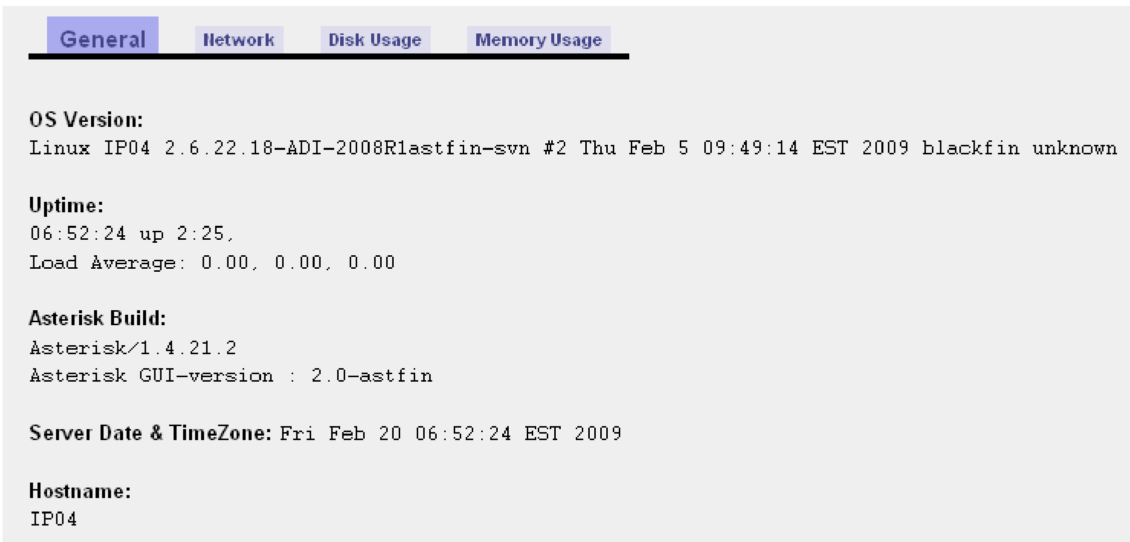

List of Custom Voice Menu Prompts

Record a new Voice Menu prompt | Upload a Voice Menu prompt

#	Name	Options
1	main_english.gsm	Record Again Play Delete
2	main_bulgarian.gsm	Record Again Play Delete

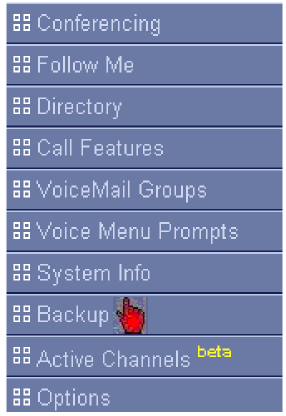
3.20. System Info





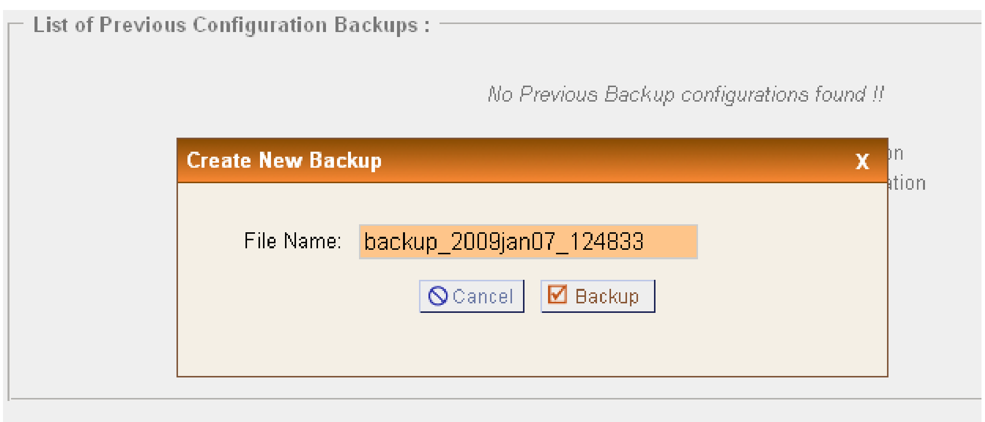
System info page provides information about vital components of your system.

3.21. Back up

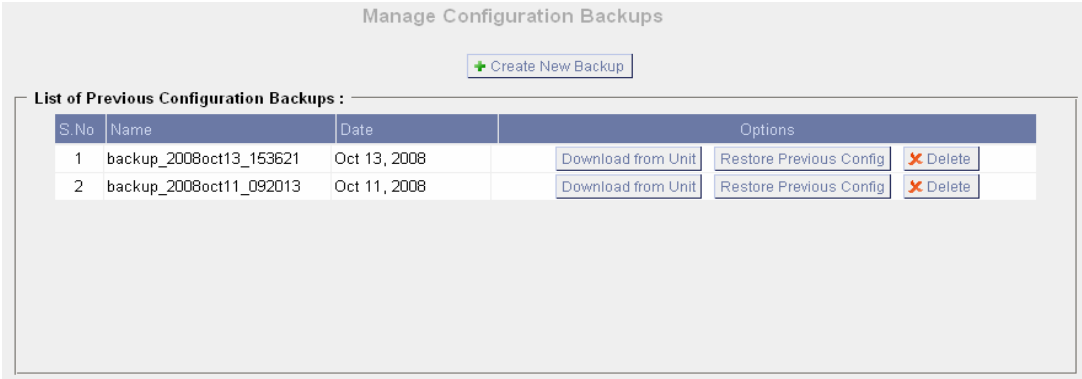


With this option you can make a back up of your system configuration, saving all the settings of the PBX. All the previous back up files are visible in to window.

- If you press **Create New Backup** button all the PBX configurations will be stored in a single backup file stored in the PBX itself. Please note that at this point we will only backup the content of /etc/asterisk. Your voicemail, custom recordings and networking setup will not be saved.



- The backup file will be tagged with the current date and the backup will be displayed in the list of backups as it is shown below



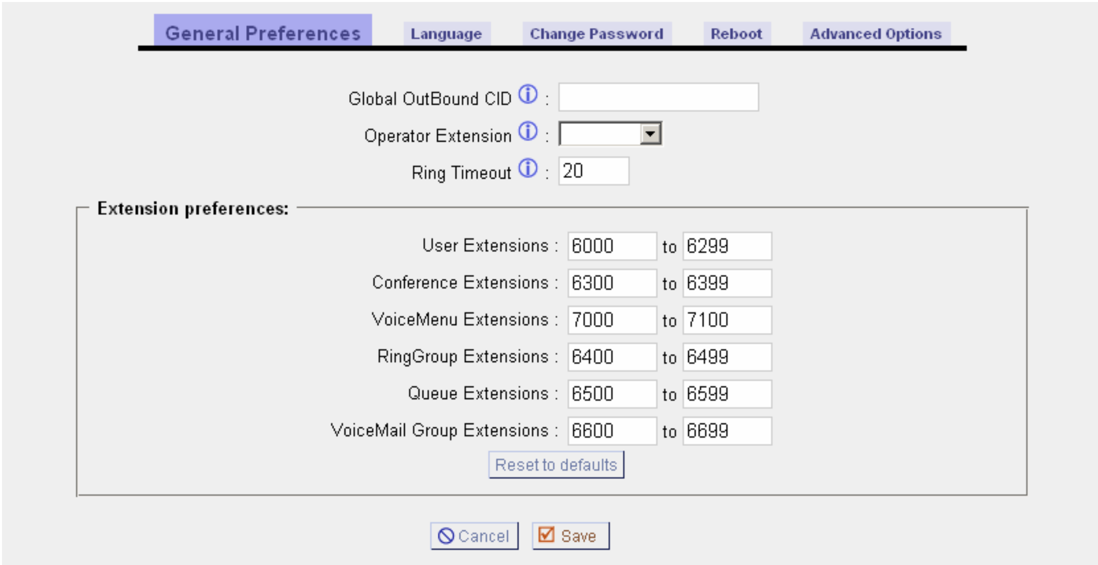
- You can download the backup file from the PBX to your local PC for even safer storage. Do this by pressing **Download from Unit** button

- If you want to restore the configurations which is backed up on the given date just press the corresponding button **Restore Previous Config.**

Please note that we do not provide upload functionality at this point.

3.22. Options

This is the general view of the Options menu



**Reminder – You can always use the "i" (info) tooltips for additional information*

The first tab in the menu is General preferences.

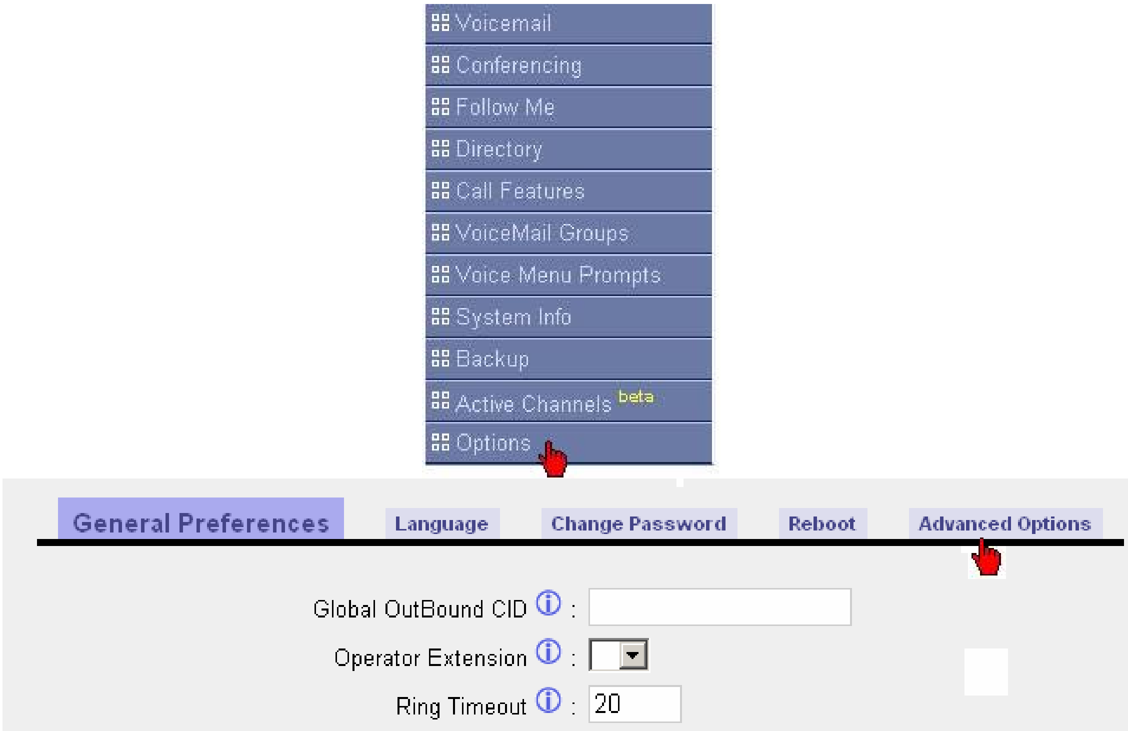
- Global Outbound CID – Caller ID to be used for outbound calls, if no specific callerID is defined in caller's profile.
- Operator Extension – extension that will be dialed when a user presses "0" in VoiceMail menu.
- Ring Timeout – default global timeout.
- Extension Preferences – ranges of extensions for specific features of the IP0x PBX.

- Language – You can select the GUI language here. Only English is supported at this time.
- Password – Change your admin password

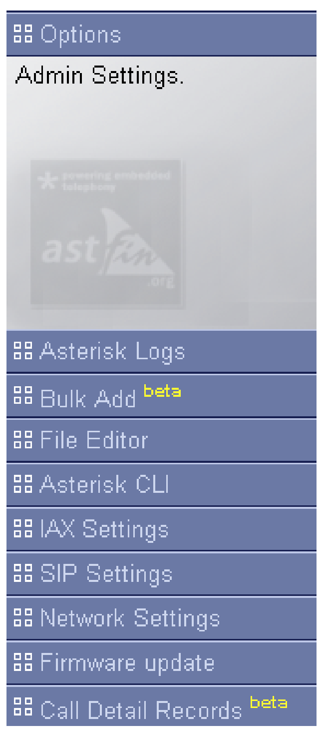
- Reboot – reboot the appliance.

3.23. Advanced Options

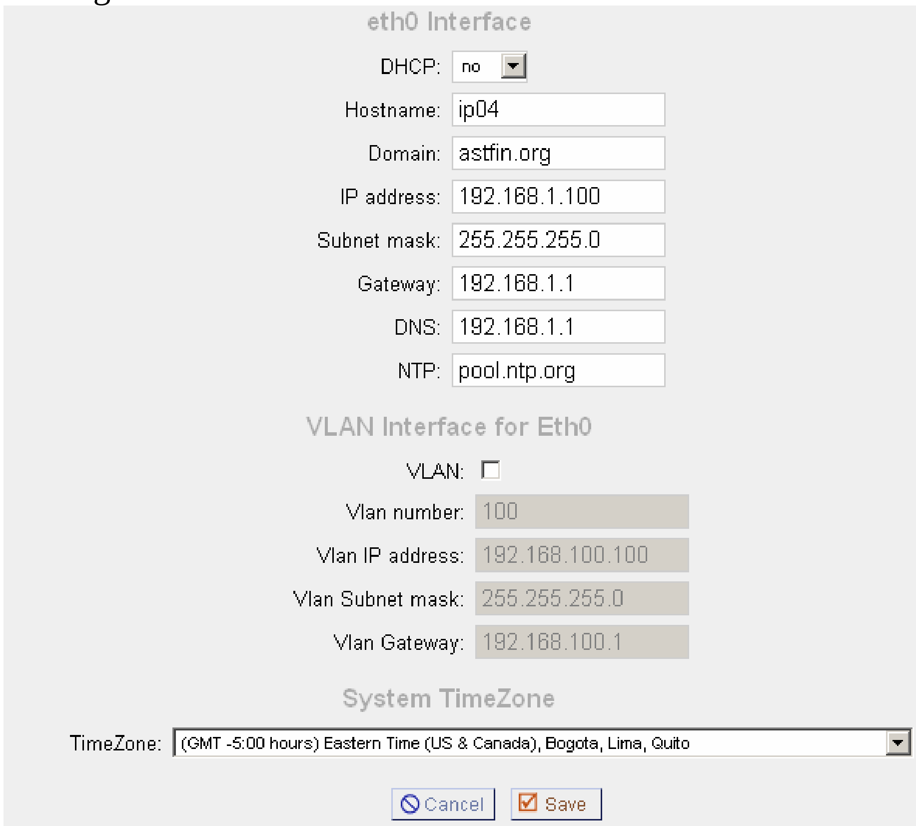
To enable the advanced option, click on the "Options" button, select the "Advanced options" tab and then press the "Show advanced options" button.



You will see the following at the bottom left of your GUI main menu.

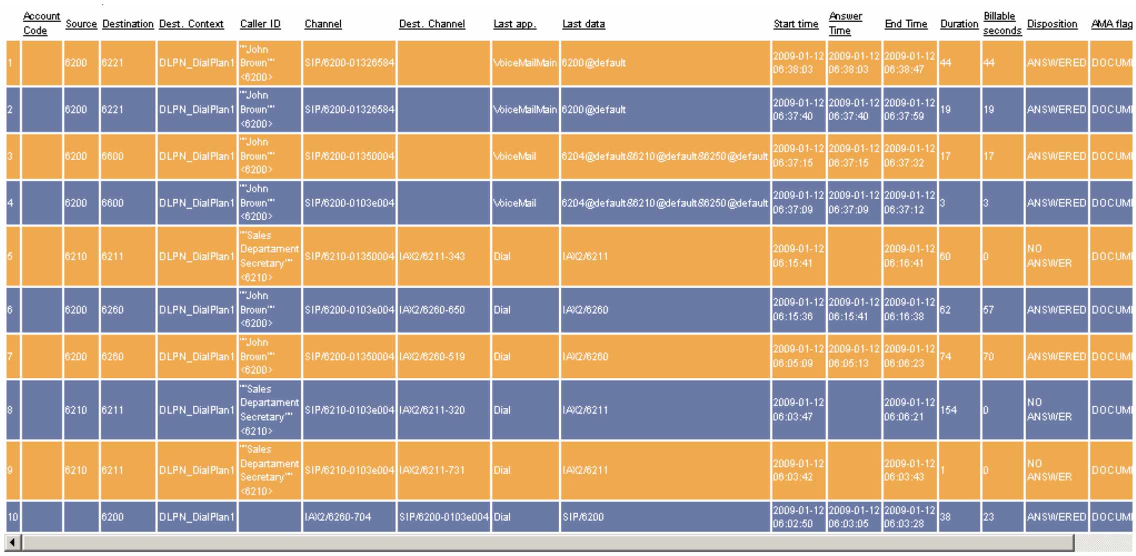


Network settings



- DHCP - selection determines if the PBX will use static IP or it will obtain its IP from the DHCP server of the network that IP0X is connected to.
- Hostname - is the hostname of the PBX. This is the name which will be used in any log and cdr files.
- IP address - is the Internet Protocol (IP) address of the IP0X. Please note that this field is only editable if static IP address is selected (no DHCP client)
- Subnet mask – Defines size of your LAN, please use xxx.xxx.xxx.xxx notification to specify your subnet.
- Gateway – Indicates IP address of the default router on your network.
- DNS – Indicates Domain Name Server, to be used to resolve names to IP addresses.
- IP/host name of your preferred NTP server. If unsure specify pool.ntp.org

Call detail records – provides you with information about all the calls made thru your PBX. From this record we can see the following information: source and destination of the call, start time, duration, disposition etc.

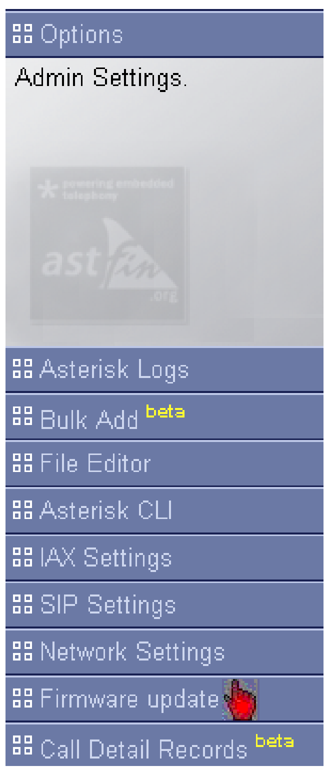


	Account Code	Source	Destination	Dest. Context	Caller ID	Channel	Dest. Channel	Last app.	Last data	Start time	Answer Time	End Time	Duration	Billable seconds	Disposition	AMA flag
1		6200	6221	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01326584		VoiceMailMain	6200@default	2009-01-12 06:38:03	2009-01-12 06:38:03	2009-01-12 06:38:47	44	44	ANSWERED	DOCUM
2		6200	6221	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01326584		VoiceMailMain	6200@default	2009-01-12 06:37:40	2009-01-12 06:37:40	2009-01-12 06:37:59	19	19	ANSWERED	DOCUM
3		6200	6600	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01350004		VoiceMail	6204@default&6210@default&6250@default	2009-01-12 06:37:15	2009-01-12 06:37:15	2009-01-12 06:37:32	17	17	ANSWERED	DOCUM
4		6200	6600	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-0103e004		VoiceMail	6204@default&6210@default&6250@default	2009-01-12 06:37:09	2009-01-12 06:37:09	2009-01-12 06:37:12	3	3	ANSWERED	DOCUM
5		6210	6211	DLPN_DialPlan1	""Sales Departament Secretary"" <6210>	SIP/6210-01350004	IAX2/6211-343	Dial	IAX2/6211	2009-01-12 06:15:41		2009-01-12 06:16:41	60	0	NO ANSWER	DOCUM
6		6200	6260	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-0103e004	IAX2/6260-650	Dial	IAX2/6260	2009-01-12 06:15:36	2009-01-12 06:15:41	2009-01-12 06:16:38	62	57	ANSWERED	DOCUM
7		6200	6260	DLPN_DialPlan1	""John Brown"" <6200>	SIP/6200-01350004	IAX2/6260-519	Dial	IAX2/6260	2009-01-12 06:05:09	2009-01-12 06:05:13	2009-01-12 06:06:23	74	70	ANSWERED	DOCUM
8		6210	6211	DLPN_DialPlan1	""Sales Departament Secretary"" <6210>	SIP/6210-0103e004	IAX2/6211-320	Dial	IAX2/6211	2009-01-12 06:03:47		2009-01-12 06:06:21	154	0	NO ANSWER	DOCUM
9		6210	6211	DLPN_DialPlan1	""Sales Departament Secretary"" <6210>	SIP/6210-0103e004	IAX2/6211-731	Dial	IAX2/6211	2009-01-12 06:03:42		2009-01-12 06:03:43	1	0	NO ANSWER	DOCUM
10			6200	DLPN_DialPlan1		IAX2/6260-704	SIP/6200-0103e004	Dial	SIP/6200	2009-01-12 06:02:50	2009-01-12 06:03:05	2009-01-12 06:03:28	38	23	ANSWERED	DOCUM

3.23.1 Updating your IP0X PBX firmware

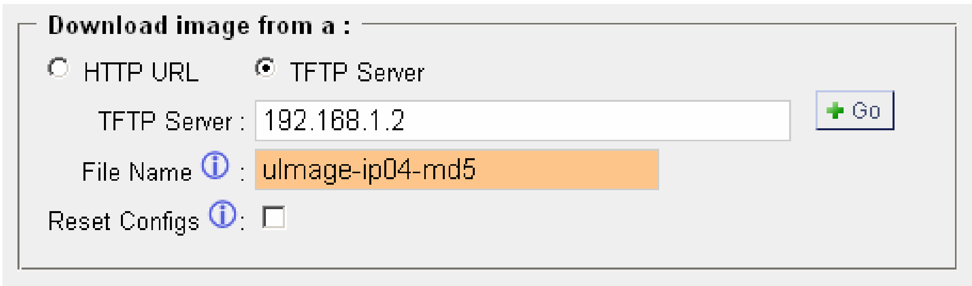
Your IP0X PBX can be updated over the network.

Here you have to select "Firmware update"



- You can use your LAN (preferred) or WAN and either TFTP server or HTTP server to distribute your new uImage. After compiling your new image in Switchfin, you will see that uImage and uImage-md5 files are available under build_ip04/image_04/ folder. Please use uImage-md5 for all updates through the GUI.

- You have to type the following information, and click the **+ Go** button. If you have a local TFTP server, you have to type its IP address and type the name of the uImage in the field **File Name**.



**Reminder – You can always use the "i" (info) tooltips for additional information*

You need to use the image with the included md5 check sum which is automatically generated when you compile **Switchfin**. The image you will find in the normal image directory. Please don't forget to put the image in the TFTP directory of your server. Alternatively, HTTP URL link can be used instead.

Important* – In order to complete the upgrade of the unit you have to **reboot the system.

4. Simple "How To" guide.

4.1 How to access the IP PBX

You have several options to access the PBX.

- Web access

First you can use the most common and easier way – The Graphical User Interface (GUI)

Just after your PBX is connected in your local network, open various web browser. Type the default address of your PBX 192.168.1.100 and after that type the default user: admin and password: switchfin

- SSH access

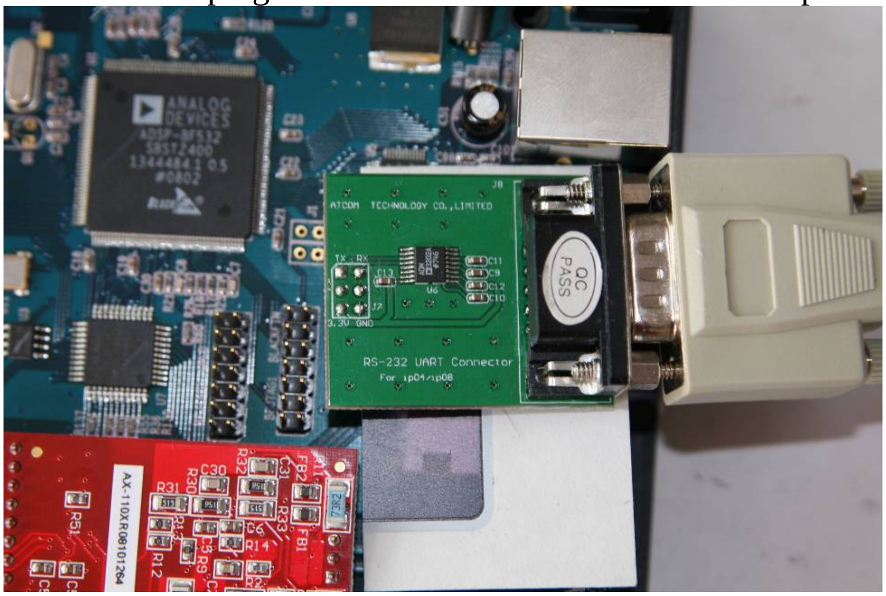
You can use software like Putty
(http://www.chiark.greenend.org.uk/~sgtatham/putty/download.html)
Or SecureCRT
(http://www.vandyke.com/products/securecrt/index.html)
This way you can access directly the Linux of your PBX and use advanced options.
For SSH access use user: root password: uClinux

- Console access

This is advanced way to access your PBX. You have to open the box of your IP PBX.
First remove the jumper

*Warning – remember to put the jumper on exactly the same place on the pin header. Other way your PBX will not boot.

You have to plug in the RS-232 board into the small 6-pin header

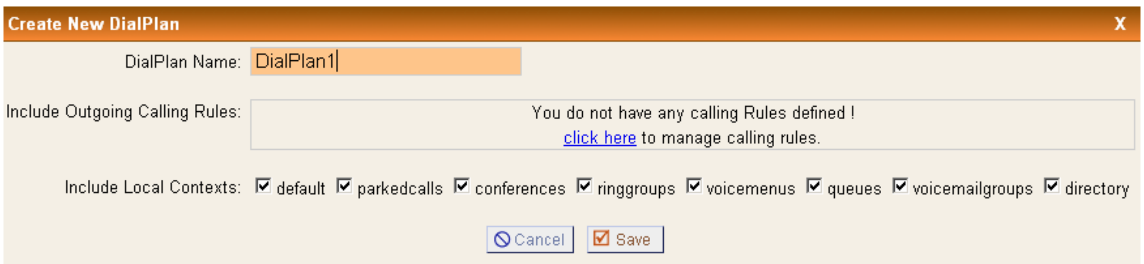


Connect the cable to the serial port of your PC
You can use SecureCRT software again with settings:
-Bit per second to 115200
-Data bits 8
-Stop bits 1

4.2 Make internal calls

At first connect your IP PBX with all interfaces and login.
For making internal calls you need a valid Dial Plan.

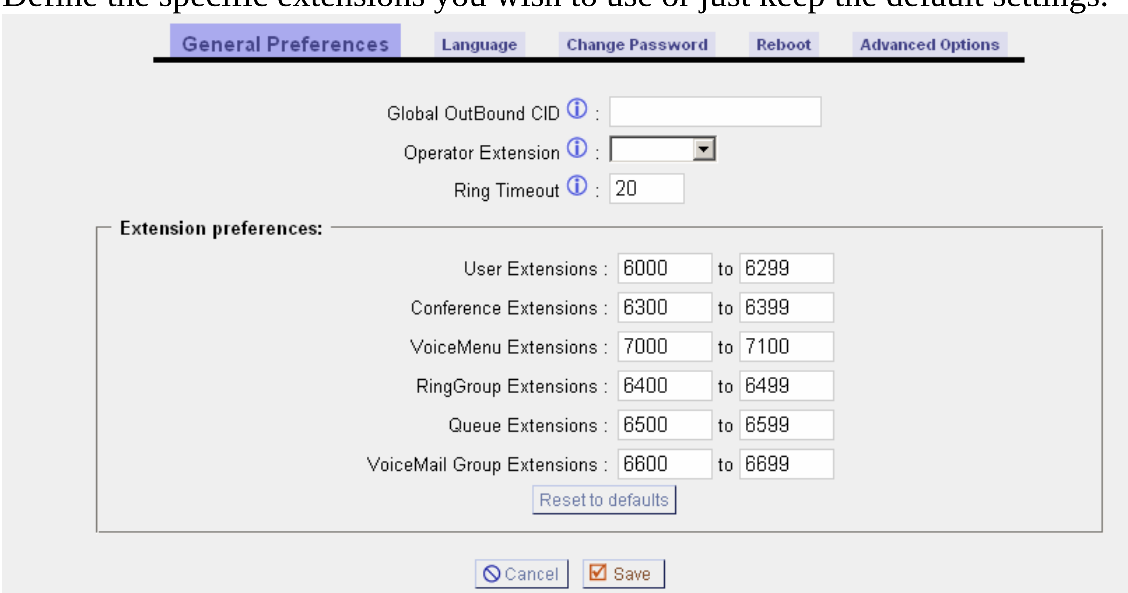
- Go to Dial Plans and select New "Dial Plan"



Select contexts as you want, and click save, this will create a default "DialPlan1" and later on we will put extensions in this plan.

- Define Local Extensions

Go to "Options" and then to General Preferences.
Define the specific extensions you wish to use or just keep the default settings.

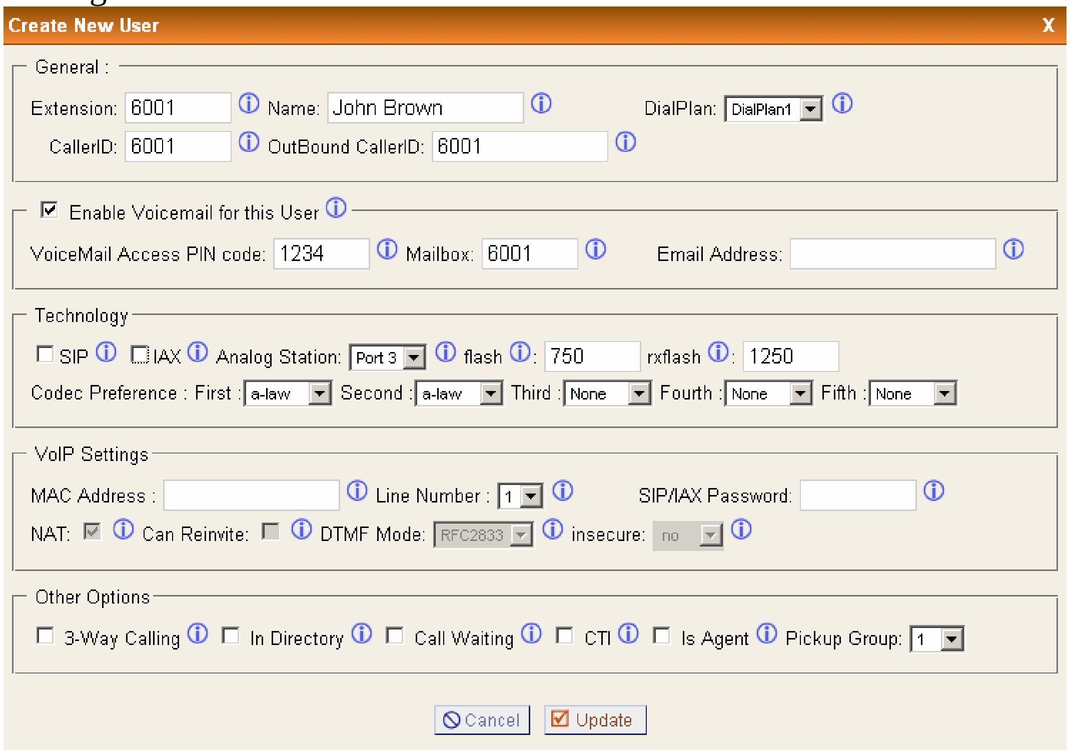


- Next step is to define the user extensions and options

Go to “Users” and then create new user

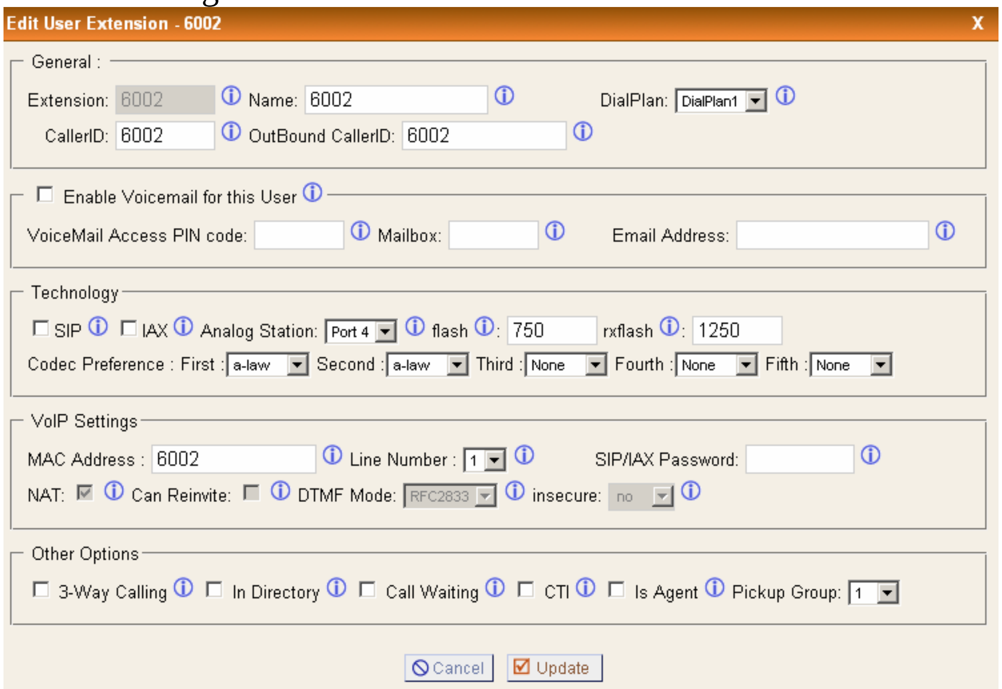
- In this example we are going to create analog, SIP and IAX clients.

Analog user



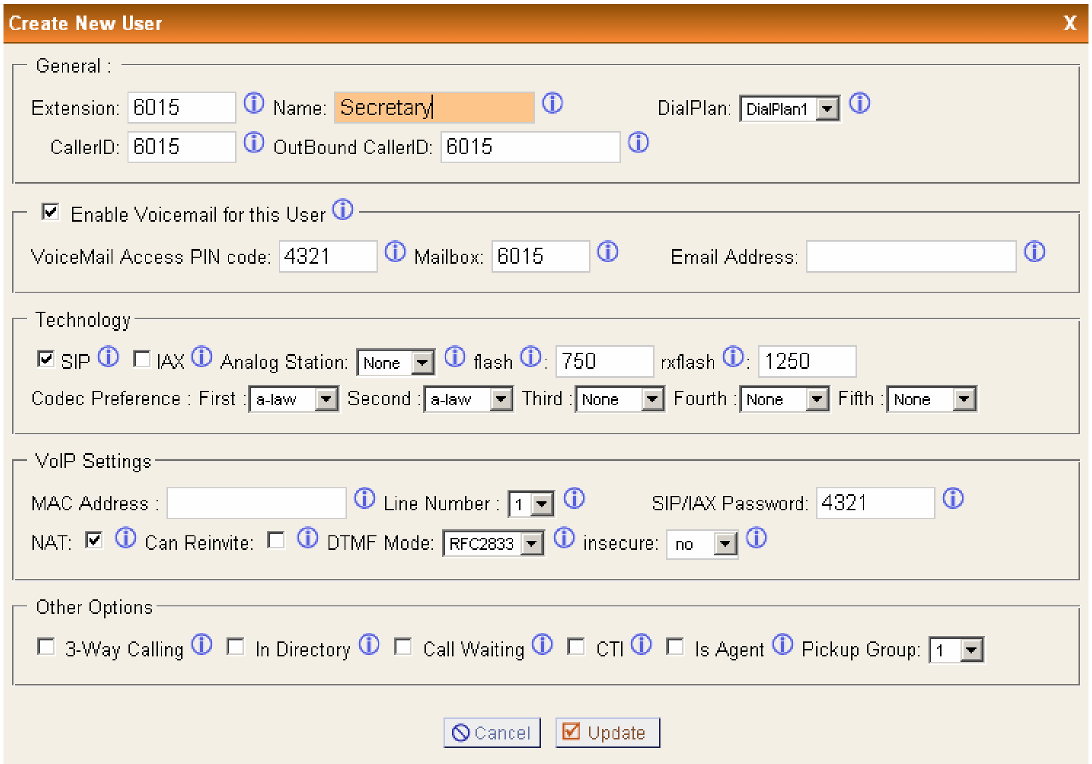
- As you can see, the user extension is 6001 and it is on name John Brown. It is using DialPlan1 and analog port 3 (FXS module) of the PBX. There is also Voicemail box setup for the user, password protected (1234)

Second Analog user



- As you can see, the user extension is 6002 and it is on name 6002. It is using DialPlan1 and analog port 4 (FXS module) of the PBX. There is no Voicemail box setup for that user.

SIP user

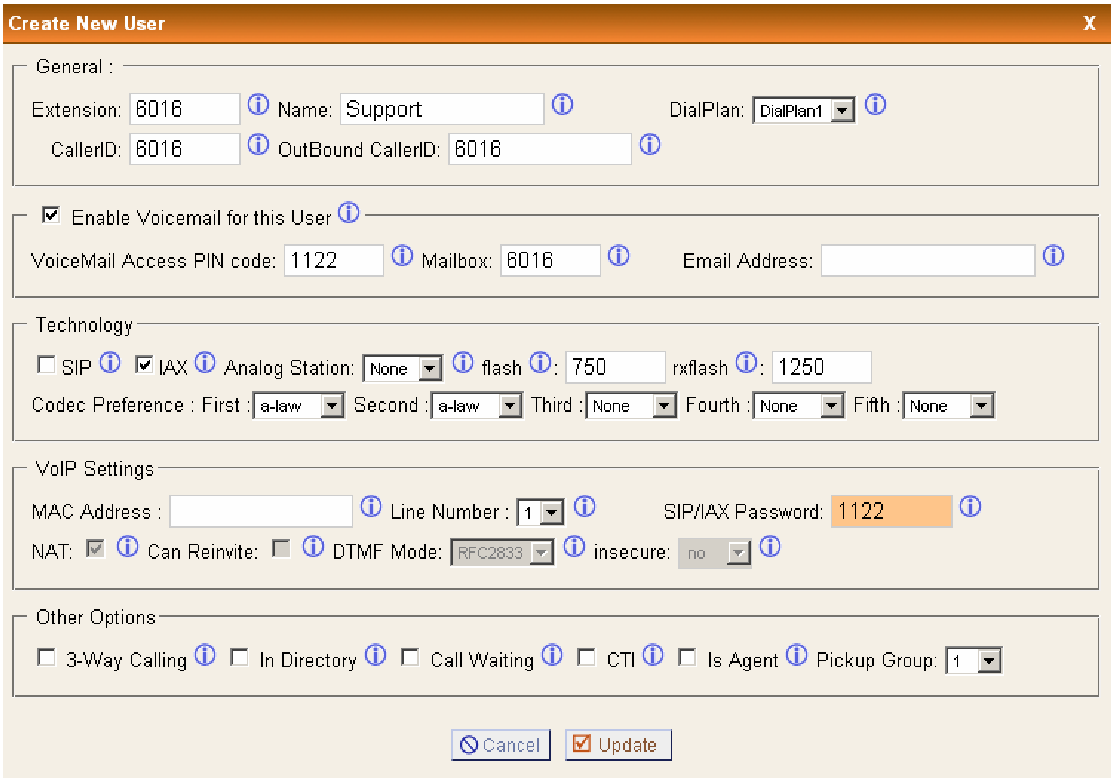


- As you can see, the user extension is 6015 and it is on name Secretary. It is using DialPlan1 and it is a SIP client. There is also Voicemail box setup for the user, password protected (1234)

  We recommend Zoiper http://www.zoiper.com/ as free SIP/IAX client for Microsoft Windows

The settings you have to make are:
SIP account, user: 6015 and the password: 4321. Save the account and register it.

IAX user



- As you can see, the user extension is 6016 and it is on name Support. It is using DialPlan1 and it is a IAX user, the password for the user is 1122 . . There is also Voicemail box setup for the user, password protected (1234)

We recommend Zoiper http://www.zoiper.com/ as free SIP/IAX client for Microsoft Windows

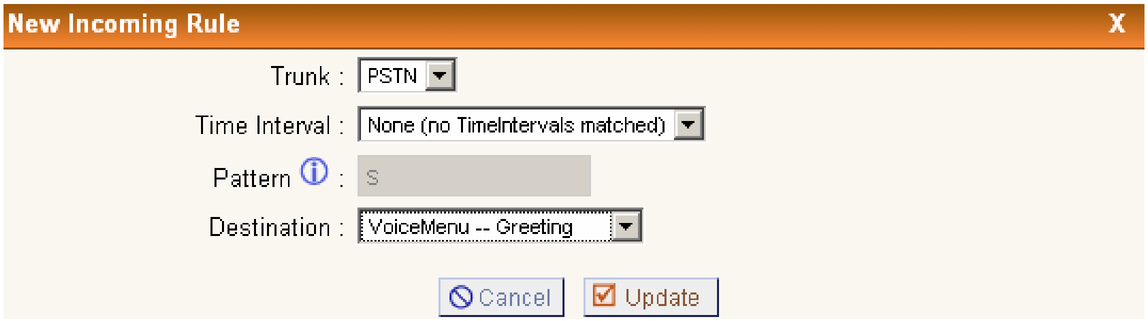
After all users are created and the analog stations are created please click “Apply Changes” tooltip.

Now you can make internal calls in you office, home and etc. with these parameters:

Name	Extension	Type of user	Password	Voicemail
John	6001	Analog station	No	Password:1234
6002	6002	Analog station	No	No
Secretary	6015	SIP	1234	Password:4321
Support	6016	IAX	1122	Password:1122

4.3 Add Incoming calls from users outside your local network

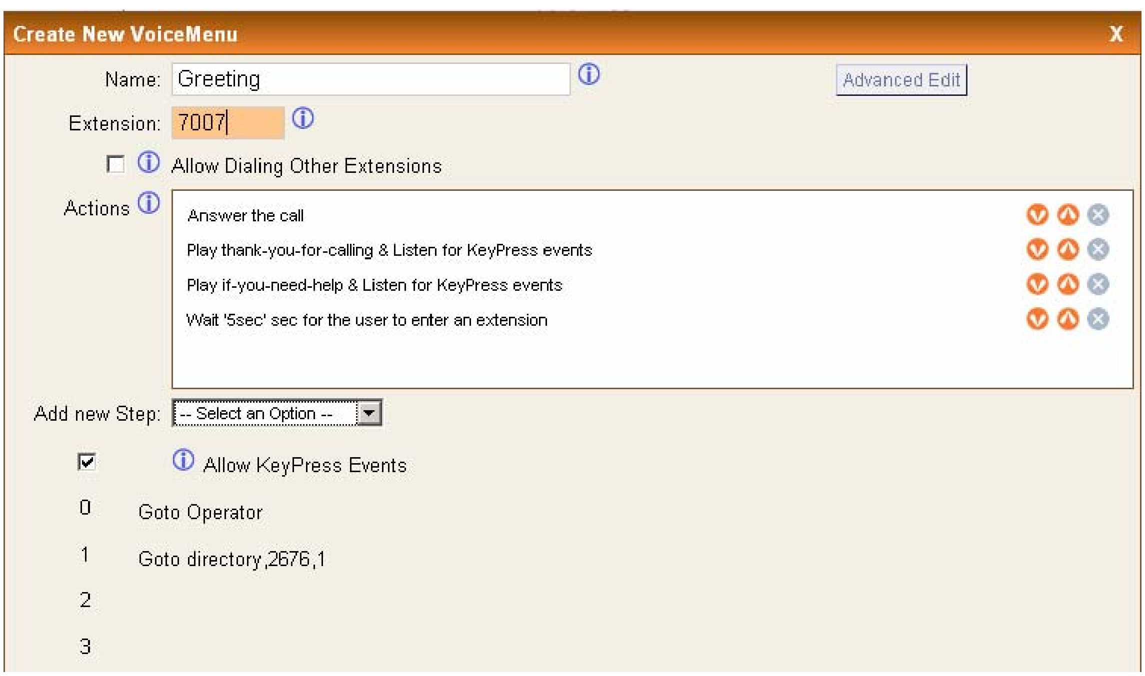
4. 3.1 Incoming calling rules



- This calling rule will send all inbound calls from the PSTN to a voice menu called Greeting

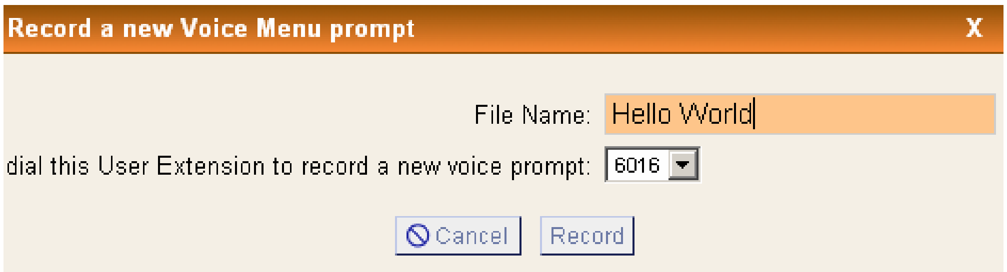
4.3.2 Add IVR – Voice menu (Greeting)

- IVR (Interactive voice response) can be very useful option in many cases. For example you have a small company with PBX, but you have only one PSTN connection and no IVR. Let's say you are looking for specific person and you don't know his internal extension what are you going to do?! This is how a voice menu can help.
To setup a voice menu go to the "Voice Menus" and click on New Voice Menu

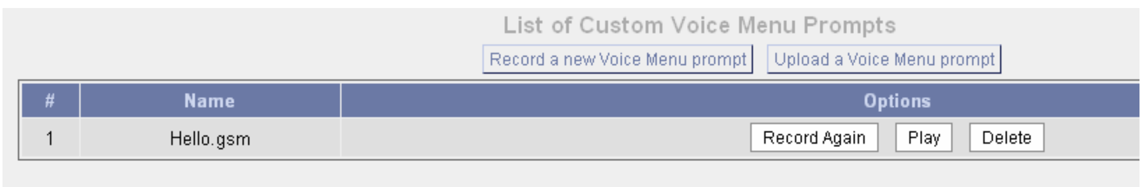


- You have to choose a name for the menu and action steps of it. In the example the menu is "Greeting" and with the setup we did earlier for the Incoming calling rules all the inbound calls will reach the voice menu.

- You can record your voice this way make a custom voice menu prompts
Go to "Voice Menu Prompts" and click record a new voice menu prompt



Click record and the 6016 extension will ring, pick up and follow the prompt recording your voice. After the recording refresh the page and you will see

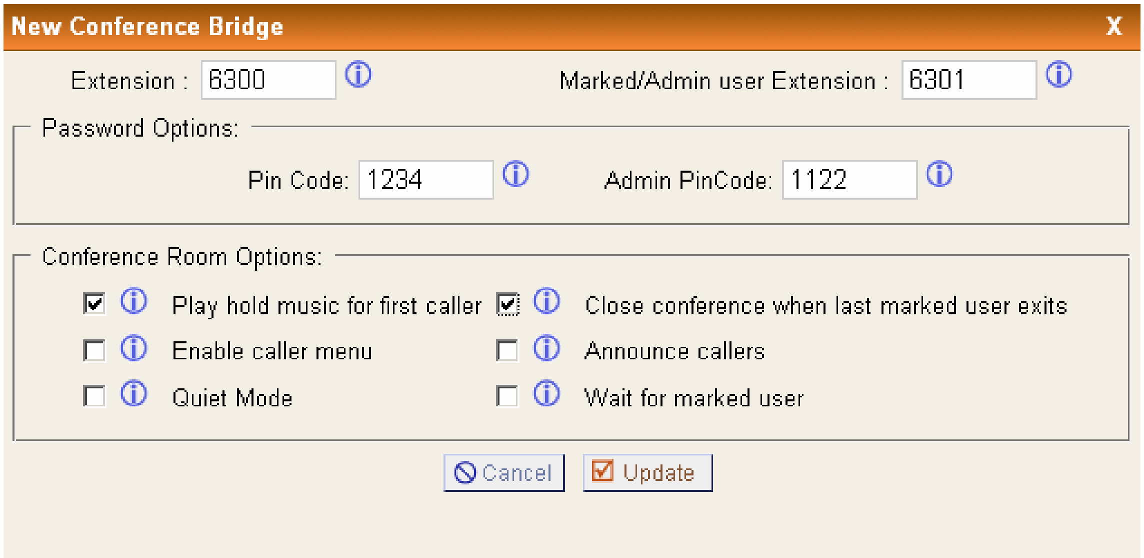


Go to Voice menus again and use the Hello prompt.

4.3.3. Add Conference room

- Popular application for enterprise use in cases like different city offices or branches. When you want to make a conference with other people you just dial a specific extension and wait for the others to join.

Go to "Conferencing" and click New Conference Bridge



This setup will give you password protected extension 6300 for all users and password protected extension 6301 for the conference Admin

4.4. Make Outbound Calls

- To make an outbound call, we need to add trunk/service provider first. There are two types of service providers:
- Analog ports: FXO ports of your IP PBX that connect to your local PSTN
- VoIP: SIP or IAX trunk, connect to remote SIP/IAX VoIP service provider server.

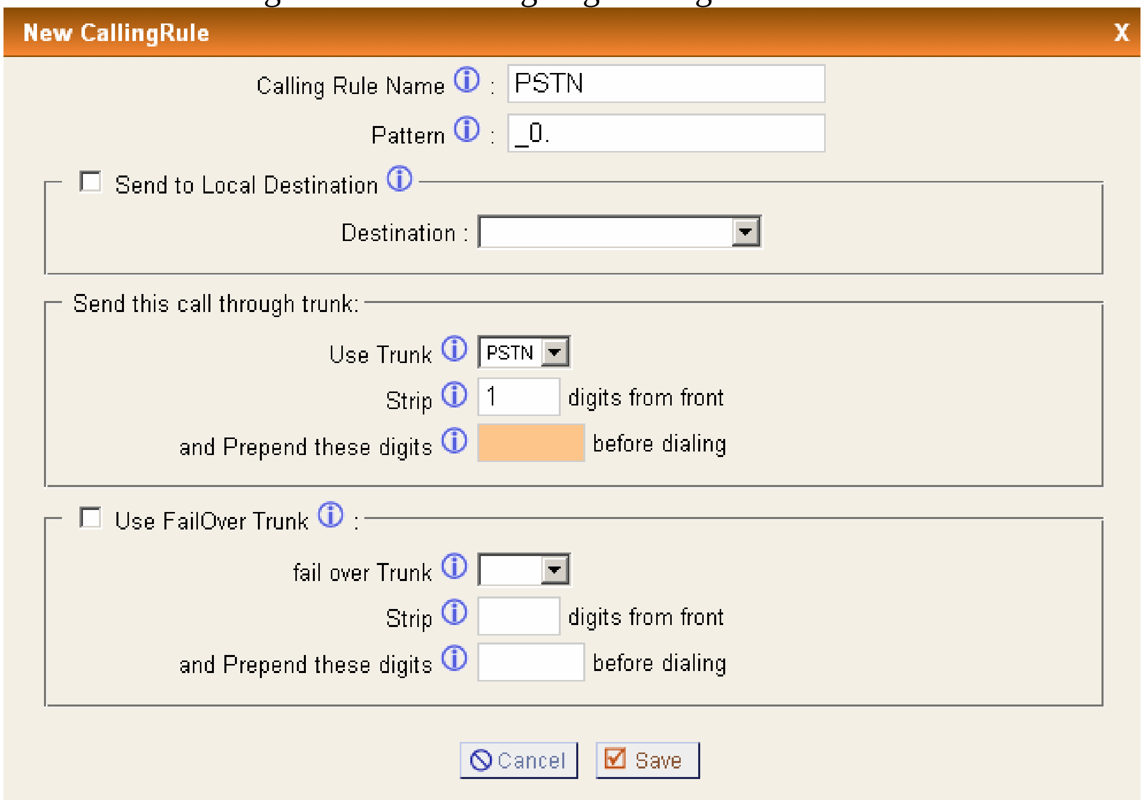
* Here we have to point that the ports of your IP PBX have indicator LED's when a LED is red the port is configured with FXO module, when it is Green it is configured with FXS module.

4.4.1 Make outbound calls using a PSTN trunk

Go to "Trunks" New Analog Trunk

The service provider will be as shown below

Add calling rule – Go to Outgoing Calling rules

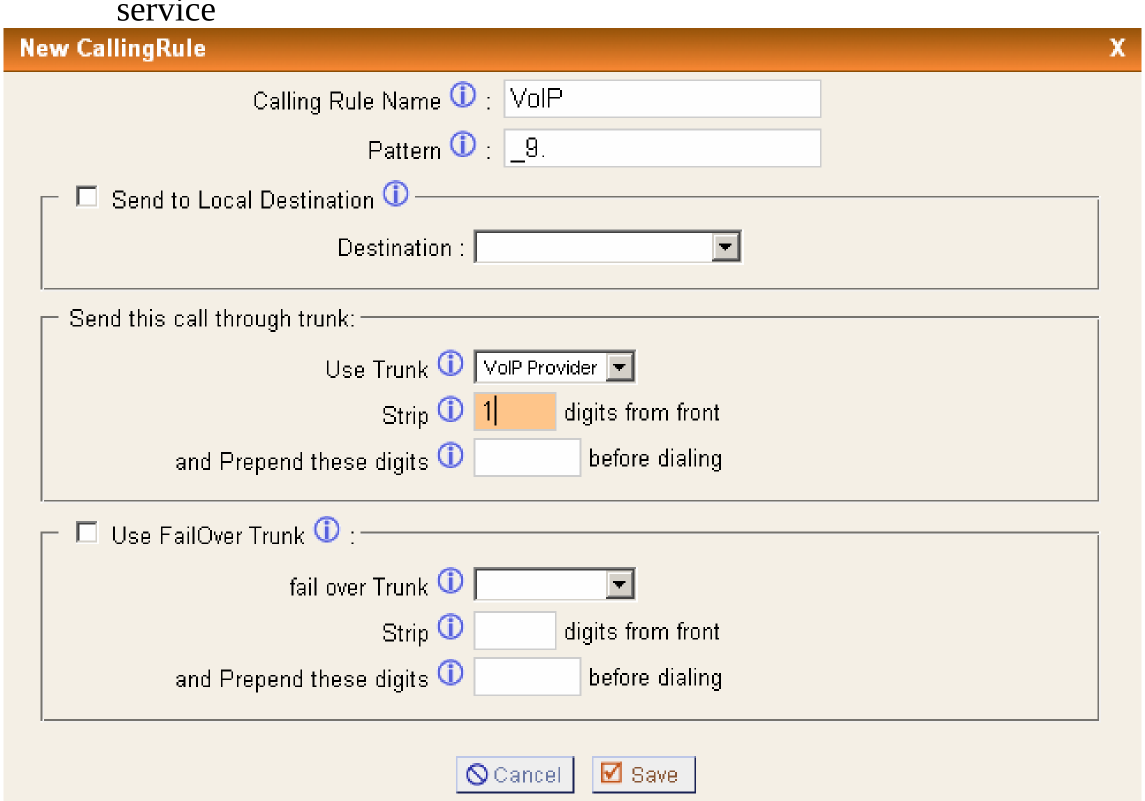


The outgoing rule we set for the PBX is: Any number that you dial starting with 0 will be accepted, its first digit (0) will be cut and the number will be sent to PSTN.

4.4.2 Make outbound calls using a VoIP trunk

Go to “Trunks” VoIP trunk and create new SIP/IAX trunk (IAX in our case)

Enter the information about your VoIP account and add the trunk. Now you have to make another outgoing calling rule for the VoIP service



- The outgoing rule we set for the PBX is: Any number that you dial starting with 9 will be accepted, its first digit (9) will be cut and the number will be sent to the VoIP trunk we just created