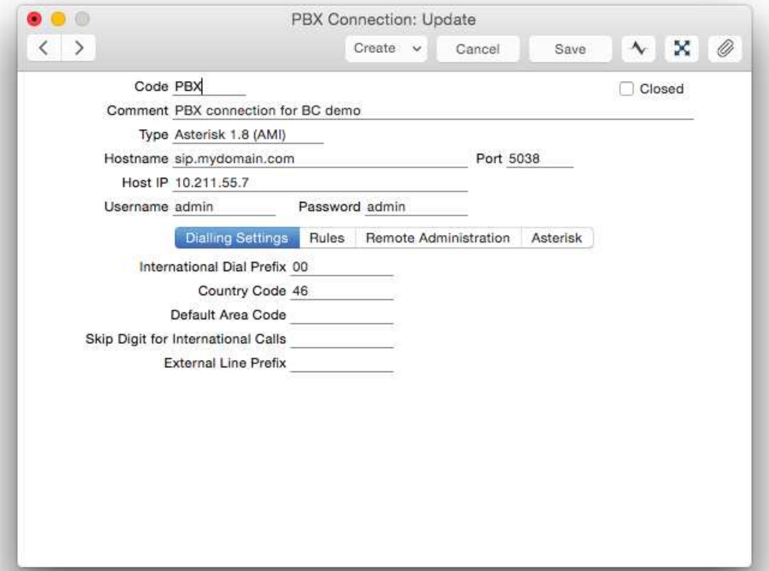
CONFIGURING AN ASTERISK SERVER

From Standard ERP's Telephony module, it is possible to fully configure an Asterisk server, whether it is installed locally on the same server as Standard ERP (only applicable on a Linux-based server) or on a separate remote server. It is also possible to integrate Standard ERP with an existing Asterisk server without managing its configuration directly from Standard ERP. Below are the steps required for a full integration, and will not explain what steps are required for local, remote, and existing servers.

PBX Connection

The basic setting for integrating your Standard ERP system with an Asterisk server is a PBX Connection. A PBX Connection represents a connection to a unique actual VoIP Server.



To start with, you should create a new PBX Connection from the Telephony module in the PBX Connection register.

A PBX Connection is defined primarily by:

Code: The PBX unique identifier.

Comment: A descriptive text.

Type: The type of VoIP server to connect to; this can be chosen from a variety of Asterisk versions, TrixBox, Digium Switchvox, VPBX and 3CX Phone System 12 etc.

Hostname: The hostname for the server being used.

Host IP address: The IP address for the server being used.

Port: The port through which the Asterisk's Management Interface (AMI) can be reached (5038 by default for Asterisk)

Username: Username to connect to the AMI.

Password: Password to connect to the AMI.

Closed: A PBX Connection can be closed when it is no longer in use or valid.

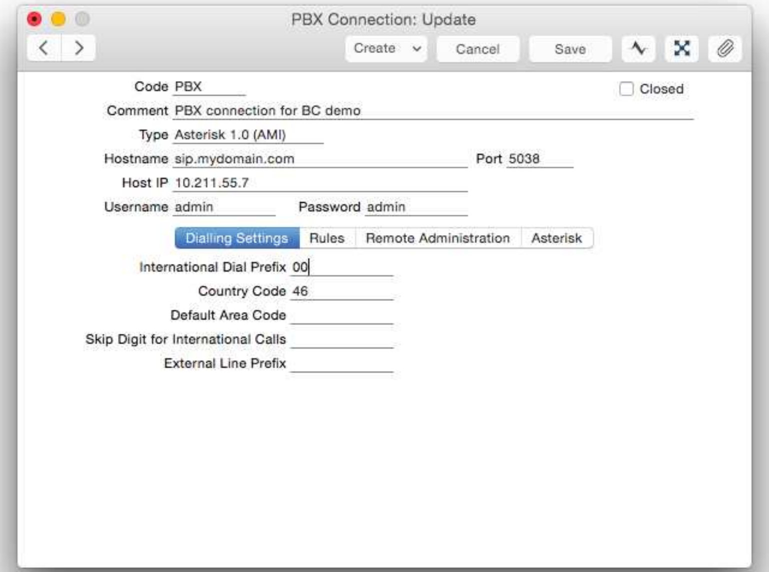
In the case of a locally or remotely installed Asterisk server, the Type should be Asterisk 1.8 (AMI). The username and password can be freely set as they will be configured in the server for you.

In case you are connecting to an existing server, the port, username, and password should match the content of your server's manager.conf file. Here is an example of manager configuration usable with Standard ERP:

```
[general]

enabled=yes

port=5038

bindaddr=0.0.0.0

allowmultiplelogin=yes

displayconnects=yes

timestampevents=yes

[myadmin]

secret=passwordxyz

deny=0.0.0.0/0.0.0.0

permit=1.2.3.4

read=system,call,originate

write=system,call,originate
```

In this case Port would be 5038, username myadmin and password passwordxyz. You should replace 1.2.3.4 by the IP address of your Standard ERP server.

A PBX Connection has other fields organised in four tabs, and which are used when managing a local or remote server entirely from Standard ERP. Administrators using an existing server fully managed by some external means should skip over to the Contact records section.



Dialling Settings

Under this tab, you can configure the various telephony prefixes in use in your system.

International Dial Prefix: To dial out of your country.

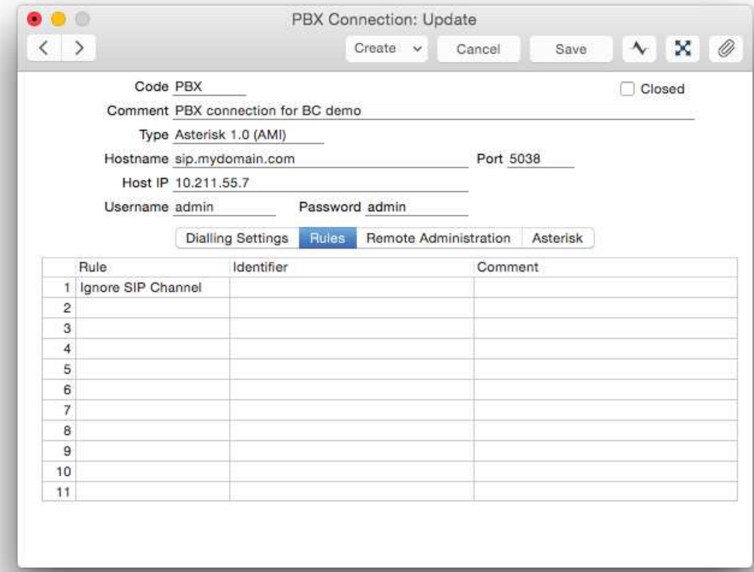
Country Code: To dial in your country.

Default Area Code: To dial in your local area.

External Line Prefix: To dial out of your organisation.

Skip Digit for International Calls: As its name indicates, you may also define digits to skip when dialling internationally.

These parameters are used to place and receive calls, and to identify contacts based on their caller ID, including when using IAX (see below).



Rules

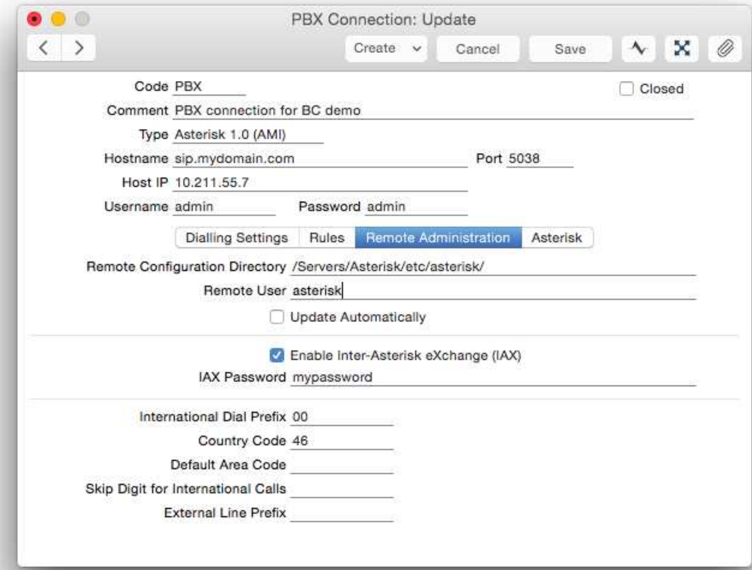
This Tab allows to define a number of rules from various types:

Ignore SIP Channel: In case of an existing server, this will ignore possible intermediary SIP channels to handle calls and instead only care about the end points. The Identifier is the name of the intermediary SIP channel to ignore.

No Act For Calls Between Extensions Shorter Than: Disables the automatic Activity creation for internal calls (detected by the short length of internal extensions). This is only applicable for PBX Connections of the Type Digium SwitchVox. The Identifier is the maximum length.

Track number: Not used.

Unique callers only: With this option, only one call will be displayed in Communicator even if there is more than one call from or to the number configured in Identifier.

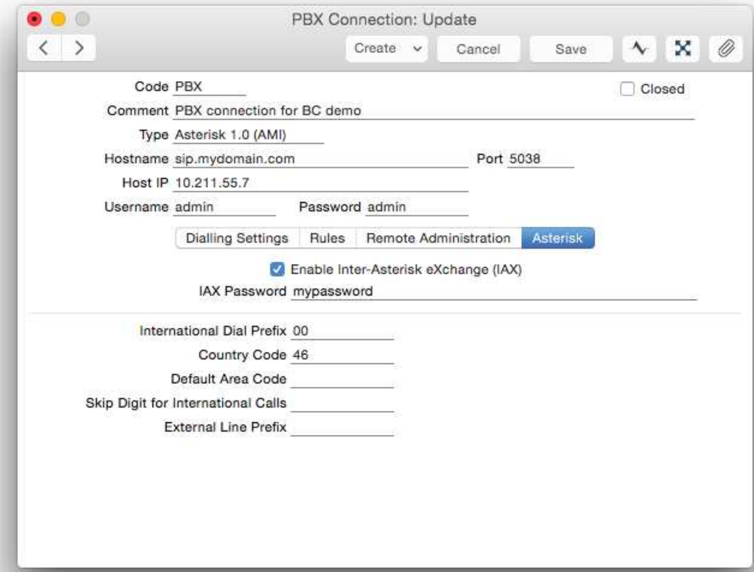


Remote Administration

From this tab, you can define the following:

Remote User: Linux user that will be used to copy the Asterisk configuration files to a remote server

Remote Configuration Directory: The path where to copy said configuration files. As such, it is important that the directory is writable by the Linux user and that your Standard ERP server has been set up to be able to connect directly to the remote Asterisk server without needing to enter a password (namely set up a Public Key Authentication between both servers).



Asterisk

From this final tab, you can enable the connection of your Asterisk server to other Asterisk server using IAX (Inter-Asterisk eXchange).

Enable Inter-Asterisk eXchange (IAX): By ticking this option, you will allow all other PBX Connections configured in Standard ERP and set to use IAX to connect to this particular server as well as allow this server to connect to all other servers enabled for IAX and configured in Standard ERP.

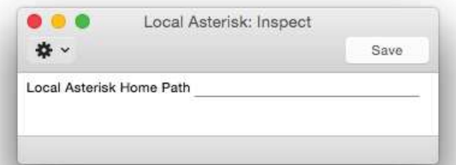
IAX Password: The password used by this server to connect to other IAX servers.

Sending configuration to the server

This section only applies to the local and remote servers situations.

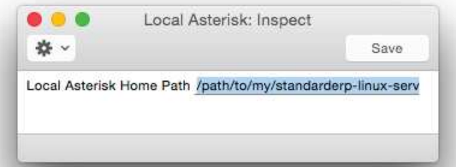
After completing the above configuration of a PBX Connection, you can already send the configuration to an Asterisk server.

If you are running a local server, you can dump the configuration files by using the Local Asterisk Server settings from the Telephony module.



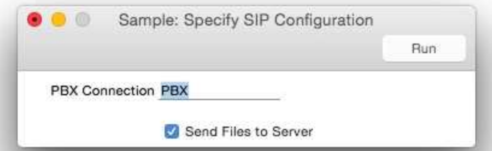
First, you will need to install the Asterisk server by selecting Setup Asterisk Server in the Operation Menu. This will give you a warning pop-up reading “Starting download and installation of Asterisk server”. Click OK. This will download the binaries for the Asterisk Server from the HansaWorld servers and install them on your local server. The server will then be started. You should never have to use Setup Asterisk Server again after this.

If you close the Local Asterisk Inspect window and reopen it, you will see the path where your server is installed.



From the Operation Menu, you can also select “Update Asterisk Server Configuration” (which will dump the current configuration on the Asterisk server configuration directory and restart the Asterisk server so that the configuration is applied), Start Asterisk Server, and Stop Asterisk Server (which should both only rarely be used, for instance for external maintenance purposes).

If you are running a remote server which is fully configured using Standard ERP, you should instead head to the Asterisk SIP Configuration Files maintenance in the Routines of the Telephony module.



Use Paste Special in the PBX Connection to select the server you want to update, and tick Send Files to Server before running. If you do not tick this option, then the files will only be generated locally on your Standard ERP server.

Note that this will only work if you have properly setup your PBX Connection and the Linux environment of your Asterisk server (see above).

You can also select from the following other Maintenance Routines:

- Asterisk SIP trunks.
- Asterisk Users.

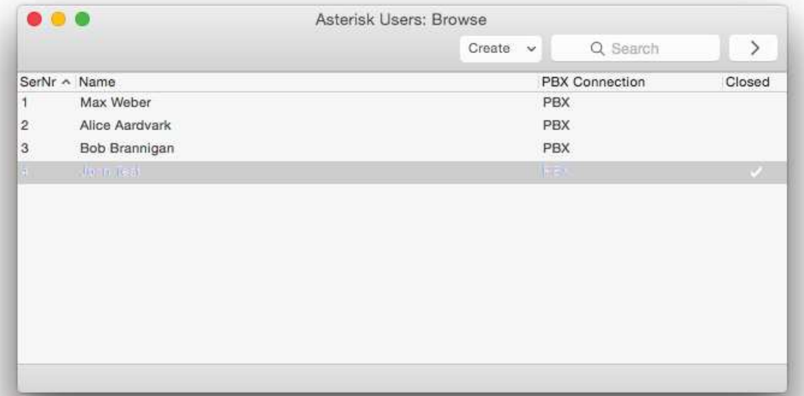
These routines will generate respectively only the configuration files for the SIP trunks of a PBX Connection, or for its users, instead of regenerating all the files.

Note that the files are only sent to the server but not applied. An administrator needs to connect manually to the Asterisk server and reload them. For instance by issuing a 'core reload' command from Asterisk's command line interface.

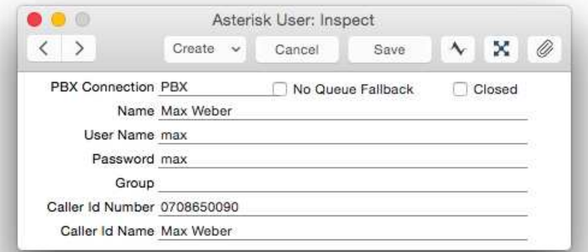
Asterisk users

The next step in setting up your Asterisk server is to create a number of users.

This section is applicable for local and remote servers.



From the Telephony module, you can create new Asterisk Users for your employees or partners.



For each user, you can define:

PBX Connections: One or more servers on which the user will be created and allowed to connect to. Leave blank to create the user on all PBX Connections configured.

Name: A descriptive name.

Username: Will be used to configure their SIP client.

Password: Will be used to configure their SIP client.

Group: No longer used.

Caller ID number: The display number that might be shown to the party this user is calling. Note that this can easily be overridden by the configuration of a SIP client or SIP trunk. Especially when dialling out to international telephone numbers, Caller ID numbers are likely to get lost.

Caller ID name: The display name that might be shown to the party this user is calling. Note that this can easily be overridden by the configuration of a SIP client or SIP trunk. Especially when dialling out to a mobile or landline telephone number, plain text display names will be lost.

Closed: A closed user will simply not be configured on the server and as such, it will not be possible to connect to the server using that user.

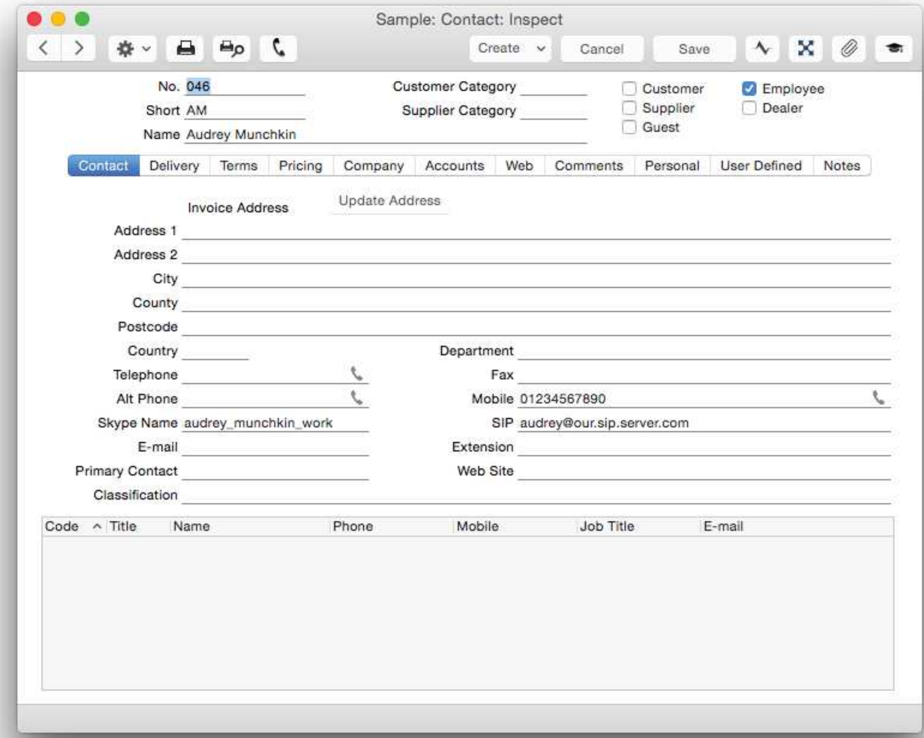
No Queue Fallback: If an Asterisk User is marked as not being part of Queue Fallback, then that user will not be called when a queue is not staffed but is receiving a phone call.

Note that after creating one (or more) Asterisk users, it is necessary to send the configuration to the server, as described above.

Contact records

This section is applicable for all types of servers: local, remote, and existing.

As of now, Asterisk users and Standard ERP users (Persons), and their contact cards are not connected and as such, Contact cards for your users will need to be filled in manually with their SIP contact details.



In the SIP field of the contact record pertaining to your Asterisk user should be filled in as username@hostname. Where username comes from the Asterisk User record, and hostname from the PBX Connection record.

SIP Trunks

This section is applicable for local and remote servers.

At this point of the configuration, you can place calls between users of your Asterisk server. To reach out to the outside world, you will need a SIP trunk or VoIP trunk. Each country usually has several providers that can help you get started. As Asterisk is a commonly used VoIP server platform, it is easy to get help from your provider in general. A simple Internet search should allow you to find a number of SIP providers for your country.

Using the information provided by your subscriber, you will be able to fill in the SIP Trunk record necessary for you to place calls to the rest of the world. A SIP provider will usually be able to sell you the usage of one, or more phone numbers that your contacts will be able to call to reach you. In some cases, your SIP provider might also allow you to place outgoing calls. Make sure to carefully select the SIP provider that is able to provide you with the capabilities you need to run your business smoothly.

Setting up a SIP trunk comes with a wide array of technical possibilities, a number of which are supported in

Standard ERP. We will detail some of those here but it is not possible to list all the possible technical configurations one can encounter.

Code: Select a unique code for your SIP trunk.

PBX Connection: Paste Special the PBX connection on which you want this SIP trunk to be terminated.

Host: Fill in the host name or IP address provided by your SIP provider here. It might be that host and domain have the same value.

Domain: Fill in the domain name provided by your SIP provider here. It might be that host and domain have the same value.

Username: Fill in the username provided by your SIP provider here.

Password: Fill in the information provided by your SIP provider here.

Skip Digit for International Calls: This parameters operates similarly to that set in the PBX Connection but will apply to calls using the SIP trunk.

Country Code: This parameters operates similarly to that set in PBX Connection but will apply to calls using the SIP trunk.

Caller ID: The caller ID of your SIP trunk provider (optional).

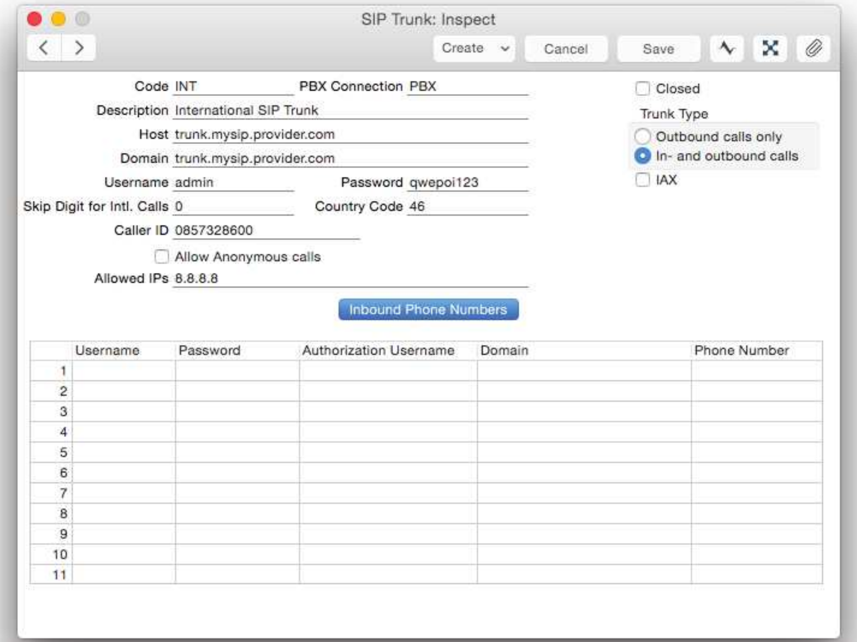
Allow anonymous calls: Lets the system accept anonymous calls coming from your providers.

Allowed IPs: Only incoming calls coming from these IP addresses will be allowed. Please check with your SIP provider to only open the minimum number of addresses. (optional but important security point).

Inbound phone numbers: A SIP provider may very well provide you several telephone numbers using the same SIP trunks. In certain cases, you will be given unique identifiers for each one of them. They should be filled in here. It might be that the usernames and passwords are the same as above.

Trunk type: Set to Outbound calls only if you intend to input a separate configuration for Inbound Phone Numbers in the matrix as described just above. Set to In- and outbound calls if you do not have a separate configuration for Inbound Phone Numbers.

IAX: Select this if your SIP trunk provider is providing you services using an Asterisk IAX trunk.



Queues and Menus

This section is applicable for local and remote servers.

Most of us are familiar with the telephony lines operated by large companies. A welcome Menu plays when you call into the support line of a company, after pushing a few digits on your phone and listening to a few more voice Menu messages, you are placed in a Queue. Thanks to Standard ERP's integration with Asterisk, your company can easily benefit from such technology.

In Standard ERP's terminology, a Menu is used to select between different queues or menus; and a Queue is used to put in relation agents answering calls and external callers. Queues and Menus share a number of settings (Phone Numbers, Opening Times) and capabilities (Playing a sound upon arrival, when closed, etc.).

Instructions for users to use queues can be found earlier in the document.

A Queue contains the following information:

Code: A unique identifier in Standard ERP

Queue ID: A unique identifier in Asterisk which will be used by your employees to connect to the queue and start answering calls.

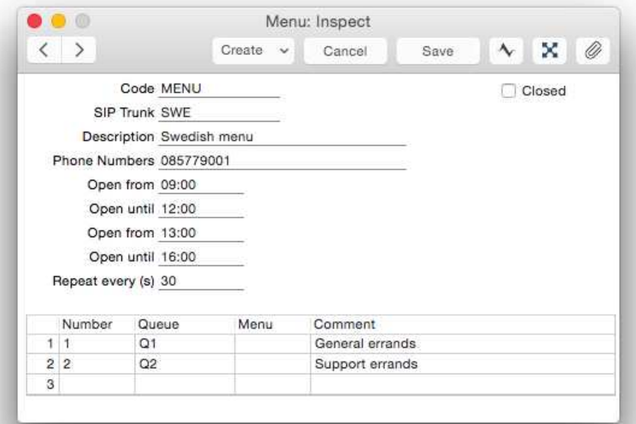
SIP Trunk: The SIP trunk from where the calls will be arriving.

Description: A free-text comment.

Phone Number: (Optional) in the case where you want a direct number for callers to reach the queue without going through a Menu. Note: you can play a greetings message even in the case where callers go straight to a queue. You do not need a Menu to play a welcome message.

Fallback number: An optional number to call in the case i) no agent is available in the queue AND ii) no one is logged in to the Asterisk server or everyone who is logged in is marked with “No Queue Fallback”.

Open from/until: it is possible to define two sets of opening hours (to include the possibility of a lunch break for instance). In case only one set of opening hours is needed, use the first pair of “Open from”/”Open until” fields and leave the second pair blank.



A Menu contains the following information:

Code: A unique identifier.

SIP Trunk: The SIP trunk from where the calls will be arriving.

Description: A free-text comment.

Phone Number: The phone number for your contacts to dial in order to access the Menu. Optional in case the Menu is accessed via another Menu.

Open from/until: It is possible to define two sets of opening hours (to include the possibility of a lunch break for instance). In case only one set of opening hours is needed, use the first pair of “Open from”/”Open until” fields and leave the second pair blank.

Repeat every (s): The number of seconds between repeats of the message explaining to the caller his or her possible choices.

A matrix finally allows you to configure the different Menus and Queues reachable from this Menu:

Number: The digit to press for the user to enter the selected Queue or Menu. Note that in the case pressing the digit leads to entering a Queue, the digit need not be the same as the Queue ID defined in the Queue.

Queue: Paste Special to an existing Queue (note, if you select this, you should not select a Menu as well).

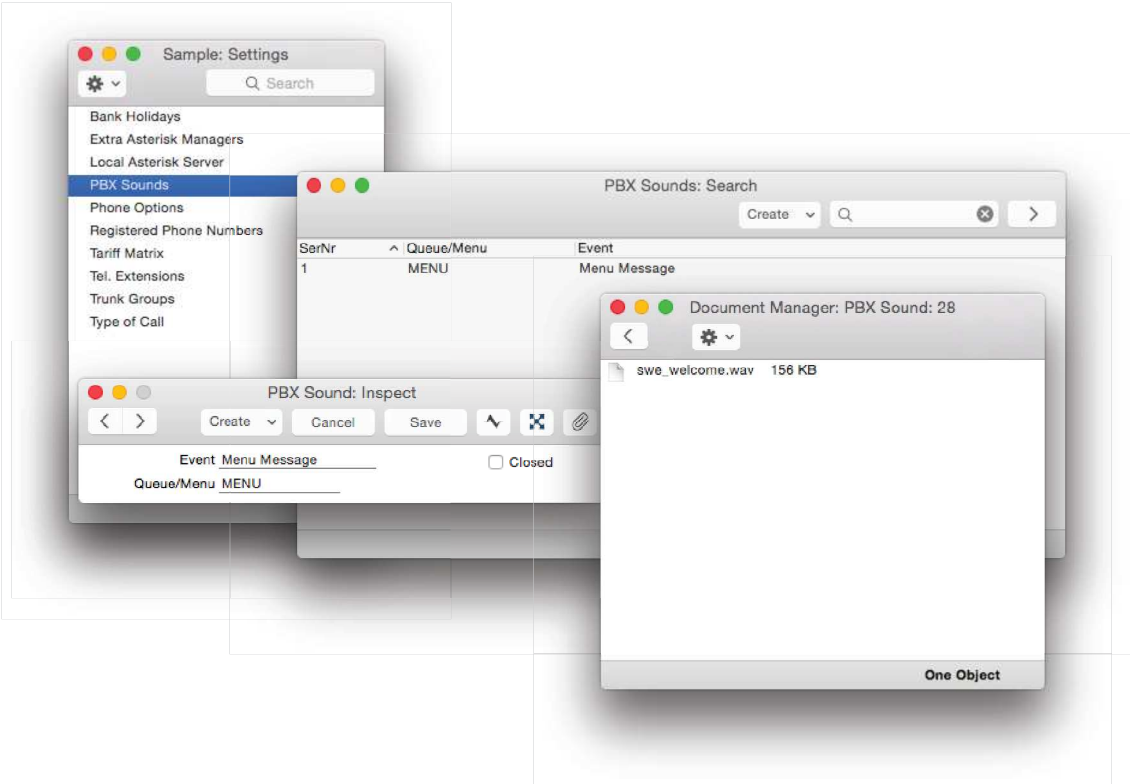
Menu: Paste Special to an existing Menu (note, if you select this, you should not select a Queue as well).

Comment: A free-text comment as a reminder of what the selected Queue or Menu might be.

Using Menus, you can cascade multiple levels of Menus. However, once a caller has joined a Queue, he or she will not be able to go back to another Queue or Menu.

The last remaining part of the configuration is now to assign sound files to be played to guide your callers through your Menus and Queues.

Whereas all the previous configuration was done in Registers of the Telephony module, sounds will be configured from the Settings of the Telephony module. More precisely, from the PBX Sounds setting.



First, create a new PBX Sound. Then in Event, use Paste Special to select the type of Event that will trigger the sound file to play. The Event you select will affect whether you are selecting a Queue or a Menu in the following field. Available Events are:

Initial Queue Message: Played as an initial greeting when a caller reaches a Queue.

Line Busy: Played after 30 seconds of a caller waiting in a Queue.

Menu Closed: Played whenever a caller arrives to a Menu outside of the defined opening hours

Menu Message: Played as an initial greeting when a caller enters a Menu (should also describes the options available from the Menu and the digits associated with each function).

Music on Hold: Music to play while the caller is waiting in a Queue.

Queue Closed: Played whenever a caller arrives to a Queue outside of the defined opening hours.

Once an Event is selected, use Paste Special to select the Queue/Menu where the sound file should be used. Only one Queue or Menu can be selected. After Saving the Record, you can now attach a file to the Record following the usual way of dragging and dropping the file over the paperclip icon or into the Document Manager window which you can open by double-clicking the paperclip icon.

Note: the attached sound file must be a mono.wav file, sampled at 8kHz.

Remember to send the configuration to the server once done. The sound files will be copied during that stage as well.