

Asterisk

In order to use your Asterisk server in conjunction with Standard ERP, your system administrator need to administer the proper setup in the Telephony module. This is covered later in this document.

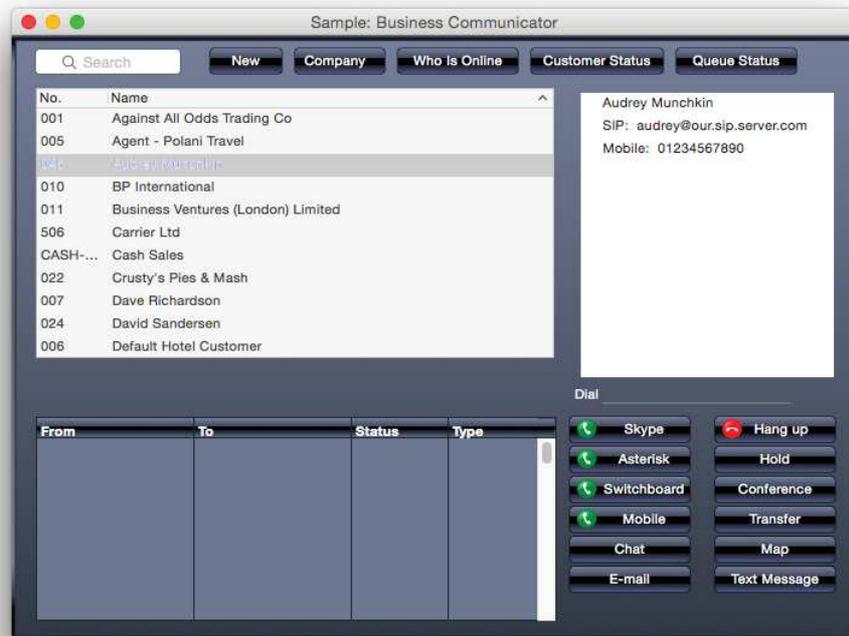
From a user's point of view, the integration with Asterisk is done in the following way:

First, you should ask your IT department for your SIP account details. Normally, a username, password, and server address (or server name) should be enough. There can be small variations depending on the exact setup in your company (you might for instance be given a domain name as well, or a caller ID, for instance).

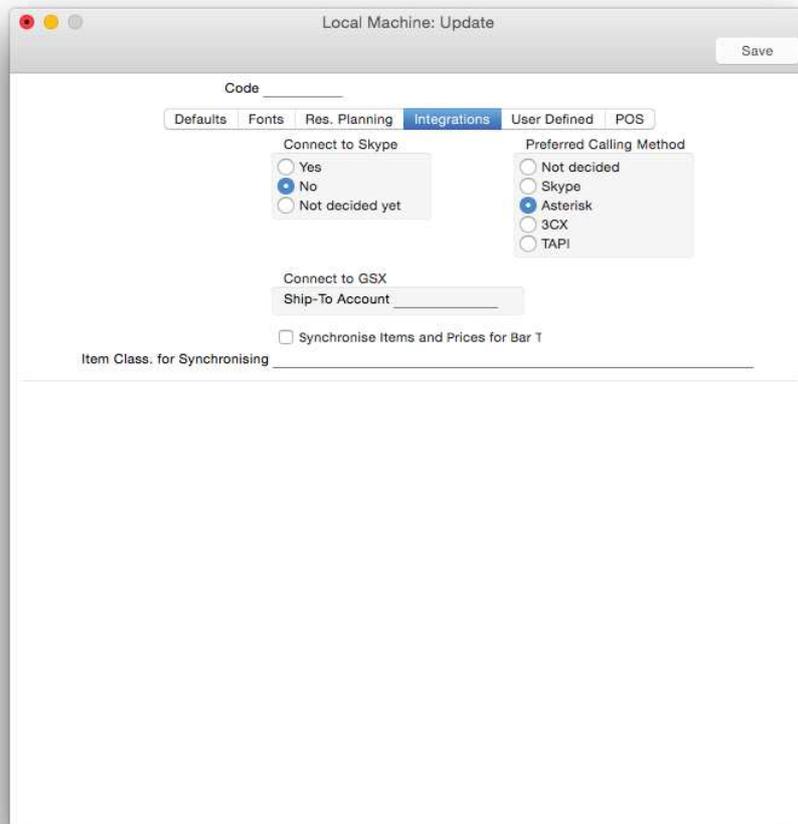
Secondly, you should verify that your user has a contact card and that its SIP extension is filled in correctly.

From the Communicator window, search for your name, click on it, and verify that there is an entry for SIP, like in the below example. If it is not the case, ask your IT department or system administrator for the exact value to enter in the SIP field on your contact card. It should normally be your_username@server_name. You can access your contact card by double-clicking on your name in the Communicator's contact list.

You should then setup your SIP client to connect to your Asterisk (SIP) server. Examples of SIP clients are Zoiper, Eye-Beam, X-Lite, and Telephone. SIP clients can run on physical phones, mobile phones, or computers. Ask your IT department for the recommended SIP client for your company and how to configure it using the information given in the first step.



If Asterisk is your preferred way to get in touch with your Contacts, you should set it up in the User Settings module, under the Integrations tab of the Local Machine setting:



If your company is using a system of Queues and Menus to answer incoming calls from your customers, partners, and suppliers, there are a number of short numbers for you to get acquainted with. These numbers will be used to log in to a queue (as in, become available to answer calls coming into that queue) and log out of a queue.

Whenever you are assigned to a queue by your manager, ask for the Queue ID as defined in the system (more on this later in this document in the section Configuring an Asterisk Server). To log in to that queue at the beginning of your shift, dial *country code*X1* where X is the Queue ID. To log out of that same queue at the end of your shift, dial *country code*X0* on your SIP client. Finally, to log out of all queues, dial *country code*00*.

After dialling these numbers, you should hear a voice message confirming that you are indeed in the queue or have indeed left the queue. You may also verify this using the “Queue Status” button of the Communicator window. Read more on this later in this document in the section named Handling Your Telephony lines: Queue Status.