APPLICATION NOTES

DEVA – SIP CONFIGURATION

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Introduction

Using the VoIP SIP service, it's possible to insert the Deva in a digital telephone network (VoIP) assigning a number to compose on the phone to speak to the people in front of one or more Deva. The following image shows a typical scenario :

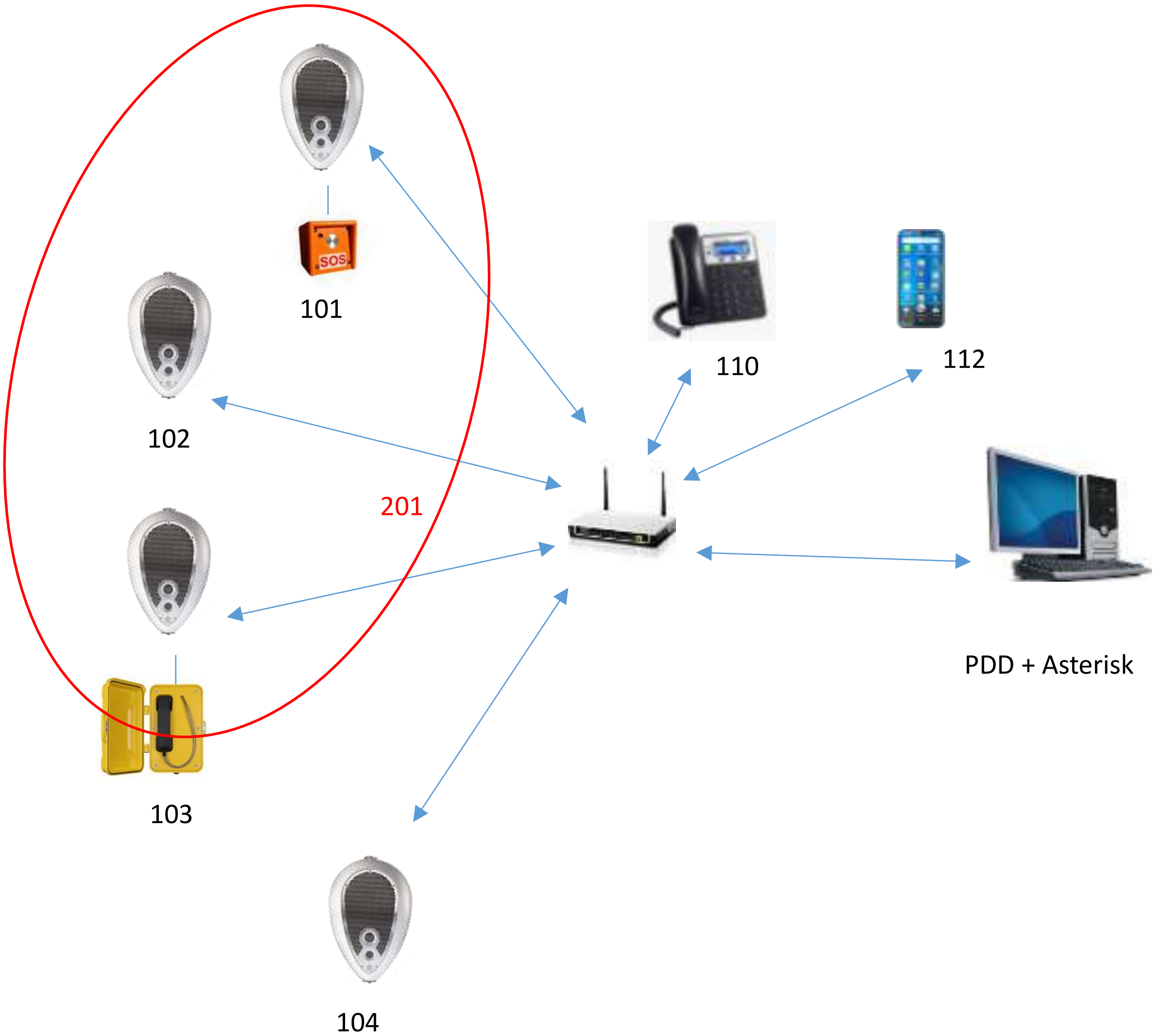

Figure 1 – Example of image display.

On Asterisk (open source SIP server for Linux) the administrator has to set the telephone numbers for the Devas, the telephone and the smartphone APP. He can also set a number for a group of Deva to allow a group call for an area. In the sample above, if the operator types the number 201 on the telephone keyboard, he can speak to the people in the area amplified by the Deva with the number 101, 102 and 103 with a good synchronization. Using the SOS box and the Handset box connected to the Deva, it's possible to have more privacy. When a person push the SOS button or lifts the handset, the Deva will call a number previously set on the Deva System Manager.

Licensing

The SIP function needs a license. To buy the license, contact the Powersoft sales office. After you purchased the license you have to send an email at support.audio@powersoft.com containing the Deva seral number and MAC address. You can find them in the Device Settings page of the DSM (*Deva System Manager*) web interface as shown in the picture below:

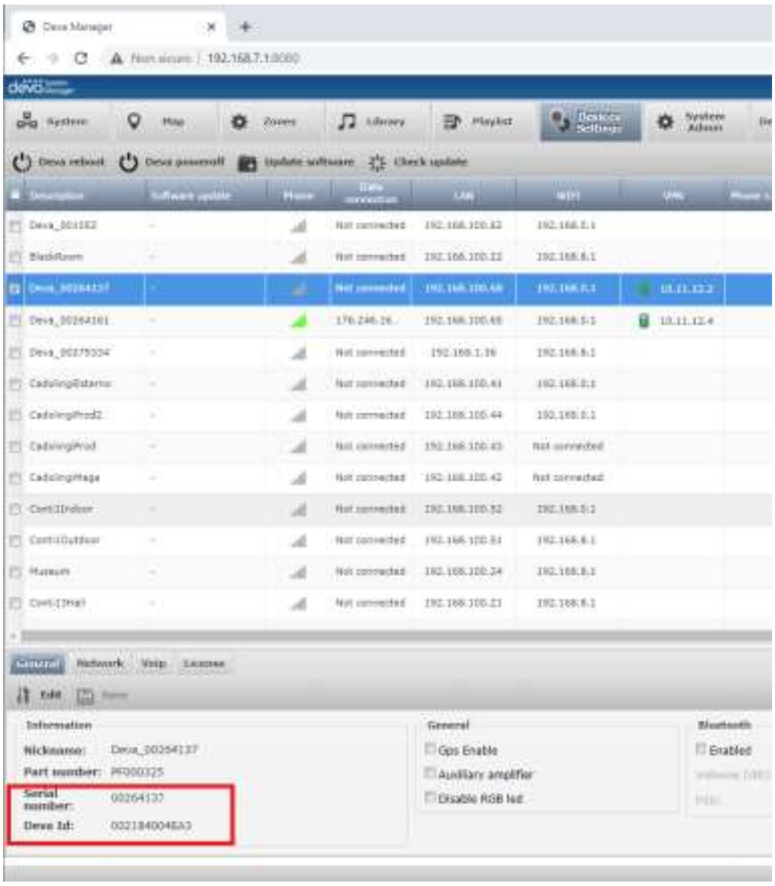

Figure 1 – Device Settings

The Powersoft support team will send you a "*.lic*" file including the validity time. To upload the license file, follow these steps:

- select the Deva in the "*Device Settings*" page
- press the "*License*" button
- press "Upload license file" button
- click on the *"+"* button and browse you disk to select the "*.lic*" file
- Press the "*Open*" button
- press the "*Upload*" button.

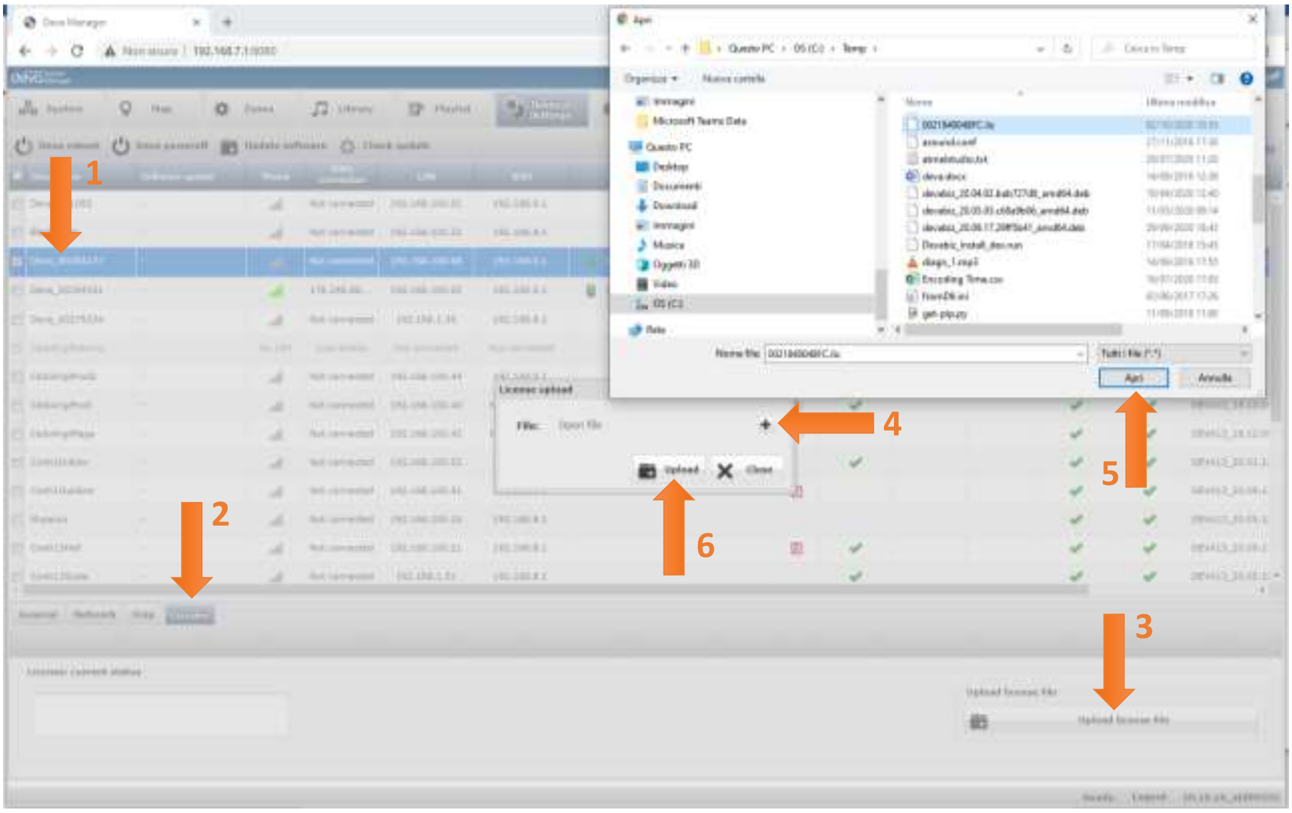

Figure 2 – Importing license

If the license file is valid you will see the licence information in the box on the left button corner:

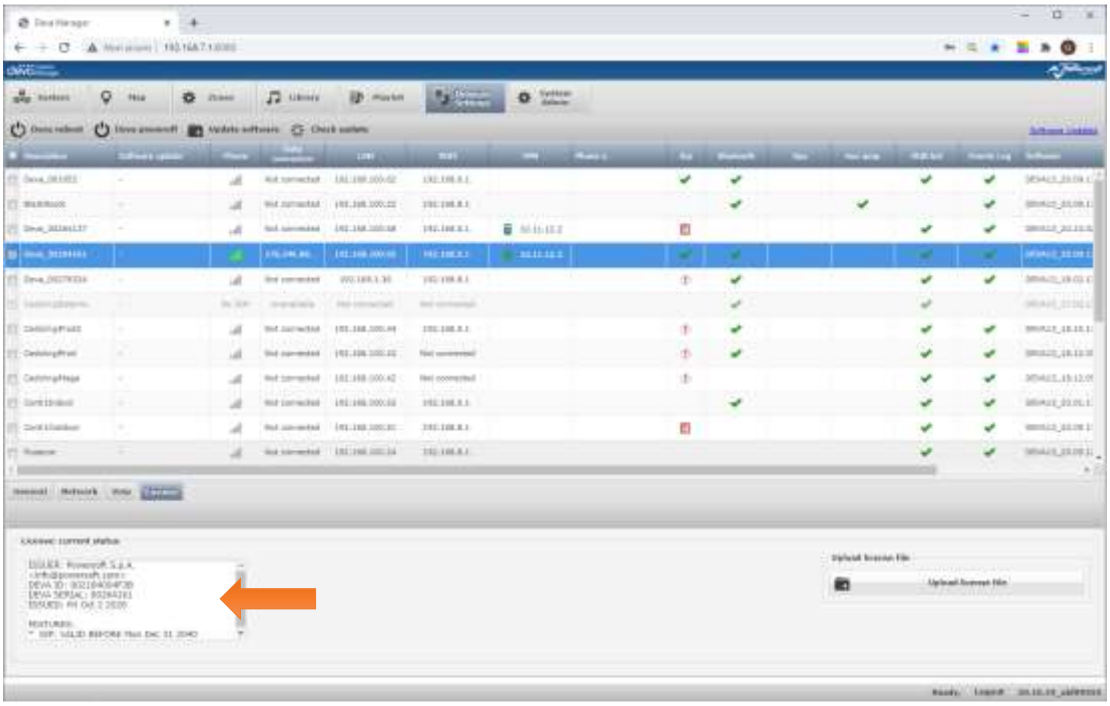

Figure 3 – License details

Configuration

To configure the SIP service for one or more Devas follow these steps:

- In the Device Settings page, select the Deva to configure and the tag “*Voip*”. If you are connected to the Deva internal web interface you see only one row in the “System” and “*Device Settings*” page Deva list.
- Using the button “*Edit*” insert the Registar (the SIP server like *Linux Asterisk* or *Genetec Sipelia*) IP address and the User and the Password assigned by the administrator of the SIP server.
- Select the “*Bind Interface*”, lan or wlan depending on which network is used by the Deva. If the Deva uses both networks you can type lan:wlan or wlan:lan depending on which priority you want to assign.
- Select the “*Audio*” volume level (attenuation in dB) and the audio output: internal speaker or AUX output.
- Press the “*Save*” button. After some seconds the green check icon will be displayed in the “*Sip*” column, if the settings are correct.

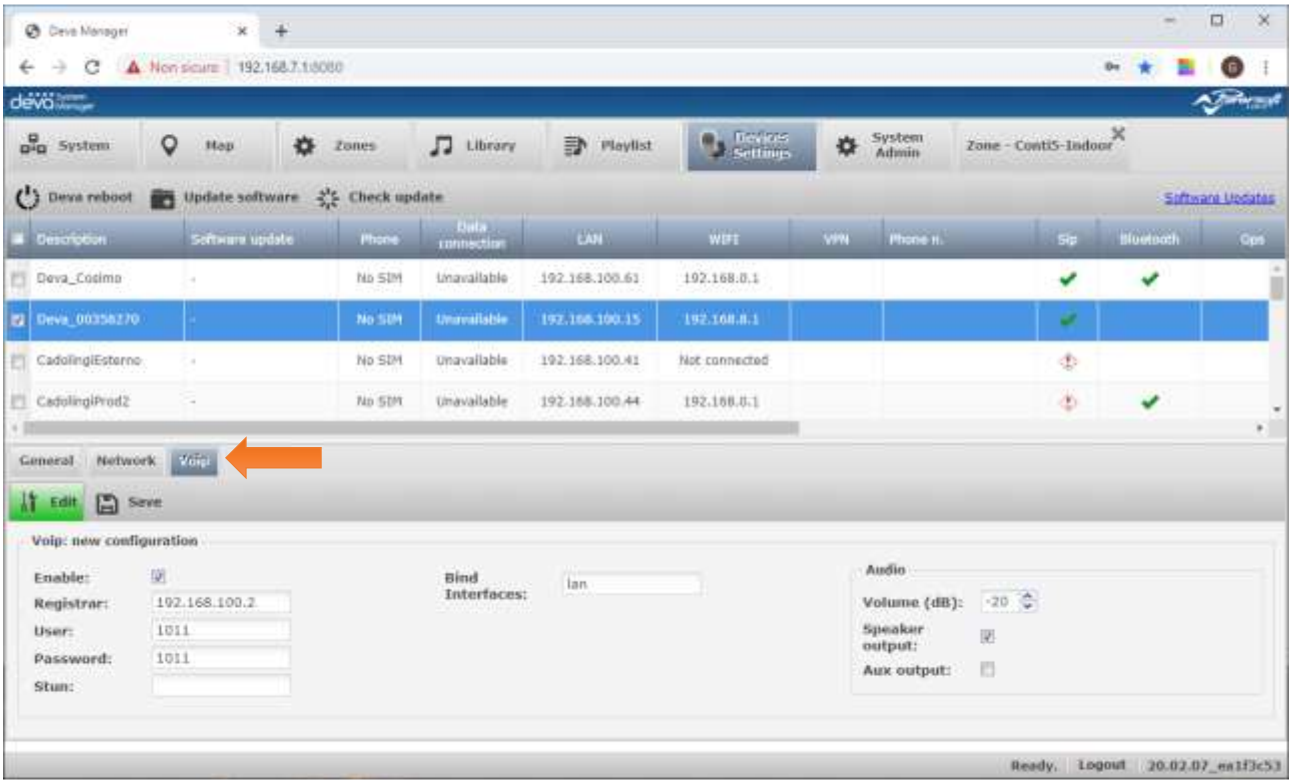

Figure 4 – Voip configuration tab

These are the icons meaning:

Missing license: please upload a valid license.

Communication problem: please verify the connection parameters: Registar IP address and login data.

Communication OK: the SIP function is working.

The Deva answers only to the calls from authorized users so you must insert them in the Contacts list. To do it, follow these steps:

- In the "*System Admin*" page select the "*Contacts*" left menu item.
- Add a new user or modify an old one and check the "*Voip call*" option.
- Click on the "*Save*" button.

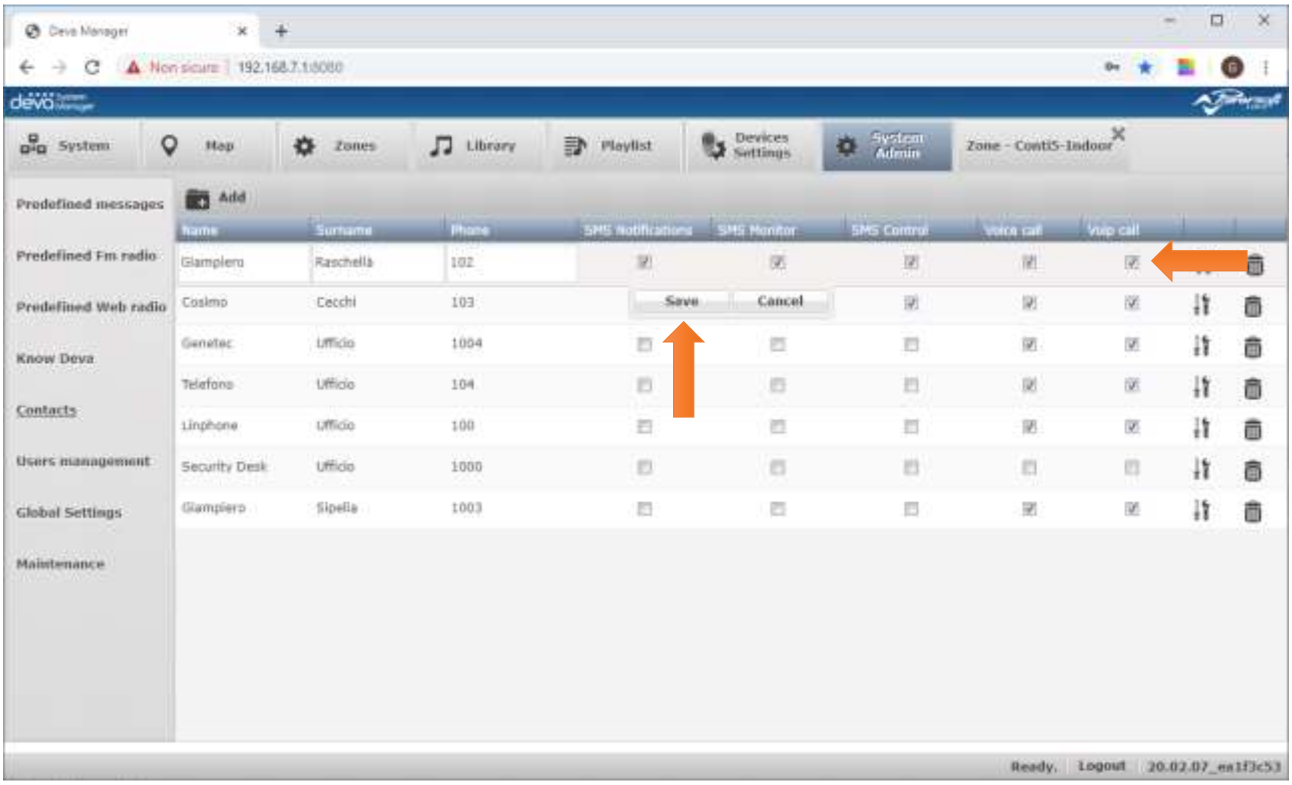

Figure 5 – Contacts window

The Deva now is like a SIP telephone with its number. You can call it and what you say will be played by the Deva like a hand free system. The Deva internal microphone is currently disabled so you cannot hear a person in front of the Deva. To do this you need at optional Box installed at man height with a speaker and microphone and a call button (useful like SOS button), to ensure privacy. Below a couple of examples.

Figure 6 – Example of SOS station

Thank to this new communication channel there is a new action in the “Events” page settings, so when an event is triggered the Deva can call a SIP number. It will rotate all the contacts as far as one answers. If none answers, there is a timeout.

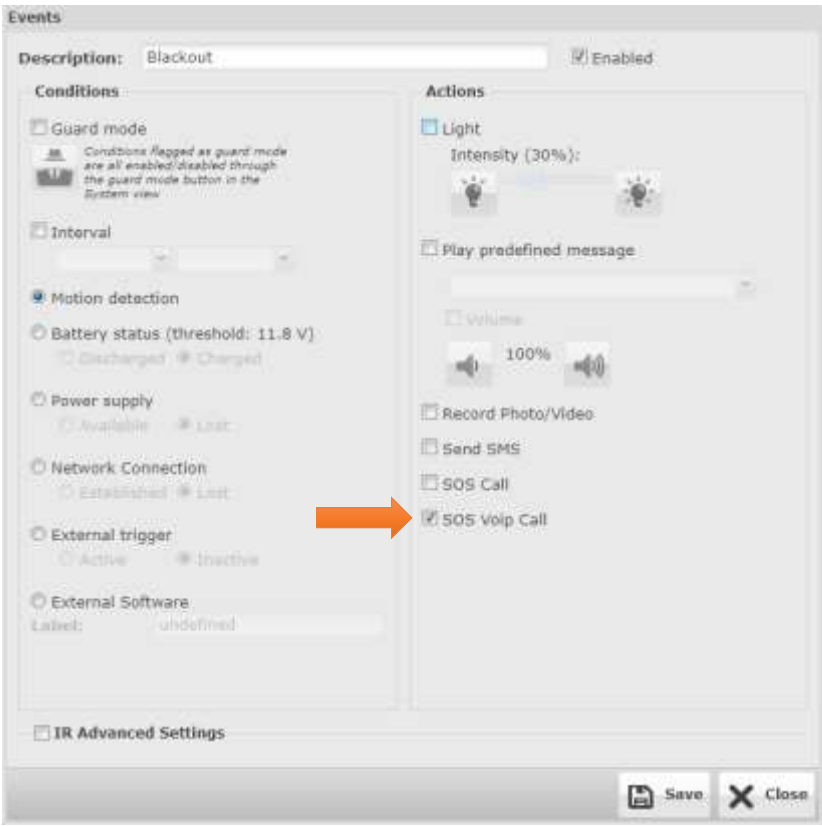

Figure 7 – Event window

APPENDIX

How to install Asterisk SIP server on PDD

- On the PDD Linux desktop, open a terminal. Be sure there is an Internet connection.
- Install the **Asterisk** packet using the following command (you will need the root password):

```
sudo apt-get install asterisk
```

- Lauch the gedit text editor using the following command:

```
sudo gedit
```

- Open the file.

```
/etc/asterisk/users.conf
```

- Add at the end of the text the following rows for 6 users:

```
[100]
fullname = 100
secret = 100
hassip = yes
context = users
host = dynamic

[101]
fullname = 101
secret = 101
hassip = yes
context = users
host = dynamic

[102]
fullname = 102
secret = 102
hassip = yes
context = users
host = dynamic

[103]
fullname = 103
secret = 103
hassip = yes
```

```
context = users
host = dynamic

[104]
fullname = 104
secret = 104
hassip = yes
context = users
host = dynamic

[105]
fullname = 105
secret = 105
hassip = yes
context = users
host = dynamic

[106]
fullname = 106
secret = 106
hassip = yes
context = users
host = dynamic
```

- Open the file

  `/etc/asterisk/extensions.conf`:

- Add a the end of the text the following rows, always for 6 users

```
[users]
include => default
exten => 100,1,Dial(SIP/100)
exten => 101,1,Dial(SIP/101)
exten => 102,1,Dial(SIP/102)
exten => 103,1,Dial(SIP/103)
exten => 104,1,Dial(SIP/104)
exten => 105,1,Dial(SIP/105)
exten => 106,1,Dial(SIP/106)
exten => 200,1,Page(SIP/101&SIP/102)
```

Please note that the last row is bold to highlight that it defines a group of Deva: if you call the 200 the Deva with the number 101 and the Deva with the number 102 will both answer to the call and you will speak on both.

- On the termina restart the Asterisk using the following command:

```
sudo service asterisk restart
```

With this procedure you will have 6 SIP users with the user number from 100 to 106 and the password (field “secret”) like the user number. You can modify the two files to add new users or to change the passwords.

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