VOICIS Core

by itcenter

User guide

ENG -1 : Accounting and CDRs

The call information about the calls is kept into 3 different tables:

acc: This table is used by the ACC module to report on transactions - accounted calls.

missed_calls table: is used by the ACC module for keeping track of missed calls. This table is similar to the 'acc' table.

crd :table is built crossing the information of **acc** and **missed_calls** to create a **cdr** table with duration and disconnect reason code.

Table 1.1. Table "acc"

name	type	size	default	null	key	extra attributes	description
`id`	`unsigned int`	`10`		no	primary	autoincrement	unique ID
`method`	`string`	`16`	`''`	no			A method is the primary function that a request is meant to invoke on a server.
`from_tag`	`string`	`64`	`''`	no			The tag parameter serves as a general mechanism to identify a dialog, which is the combination of the Call-ID along with two tags, one from participant in the dialog.
`to_tag`	`string`	`64`	`''`	no			The tag parameter serves as a general mechanism to identify a dialog, which is the combination of the Call-ID along with two tags, one from participant in the dialog.
`callid`	`string`	`255`	`''`	no			Call-ID header field uniquely identifies a particular invitation or all registrations of a particular client.
`sip_code`	`string`	`3`	`''`	no			SIP reply code
`sip_reason`	`string`	`128`	`''`	no			SIP reply reason
`time`	`datetime`	`not specified`		no			Date and time when this record was written.

Table 1.2. Table "missed_calls"

name	type	size	default	null	key	extra attributes	description
`id`	`unsigned int`	`10`		no	primary	autoincrement	unique ID
`method`	`string`	`16`	`''`	no			A method is the primary function that a request is meant to invoke on a server.

from_tag	string	64	''	no			The tag parameter serves as a general mechanism to identify a dialog, which is the combination of the Call-ID along with two tags, one from participant in the dialog.
to_tag	string	64	''	no			The tag parameter serves as a general mechanism to identify a dialog, which is the combination of the Call-ID along with two tags, one from participant in the dialog.
callid	string	255	''	no			Call-ID header field uniquely identifies a particular invitation or all registrations of a particular client.
sip_code	string	3	''	no			SIP reply code
sip_reason	string	128	''	no			SIP reply reason
time	datetime	not specified		no			Date and time when this record was written.

Table 1.3. Table "cdrs"

Field	Type	Null	Key	Default	Extra	Description
id	int(11)	NO	PRI	NULL	auto_increment	Record ID
src_ip	varchar(100)	YES		NULL		Source IP Address
src_origin	varchar(64)	NO				Original Calling Number
scr_destination	varchar(64)	NO				Original Called Number
src_proxy_ip	varchar(100)	NO				Proxy IP Address (Address seen by the Carrier)
dst_ip	varchar(100)	YES		NULL		Destination IP Address (Carrier/SIP Trunk Address)
dst_origin	varchar(64)	NO				Final Calling Number (After Translation)
dst_destination	varchar(64)	NO				Final Called Number (After Tran slation)
call_start_time	datetime	NO		0000-00-00 00:00:00		Call Start Time
duration	bigint(20)	YES		NULL		Duration of the call
sip_call_id	varchar(128)	NO				SIP Call ID
sip_reason	varchar(255)	NO		NULL		SIP Reason Disconnect code (487 Request Terminated)

ENG - Manual

ENG - 1 - Account

Account criation:

Click on **Accounts**

To create a new account click on **Add Accounts**

Fill the data (name, realm, Availiable Apps):

The realm should make sense.

Available Apps

Choose the apps for each account. Finish click on **Save**.

To use the account.

To select the account click on the desired account.

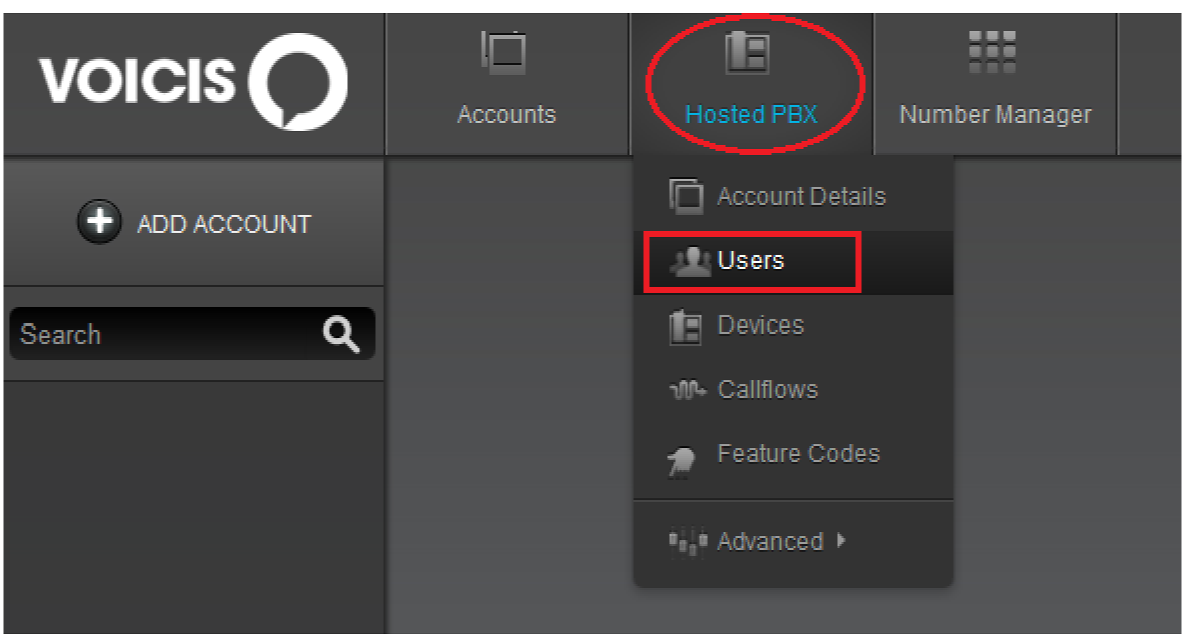
Click on **Use Account**, to configure the client Feature.

ENG - 2 - User Management

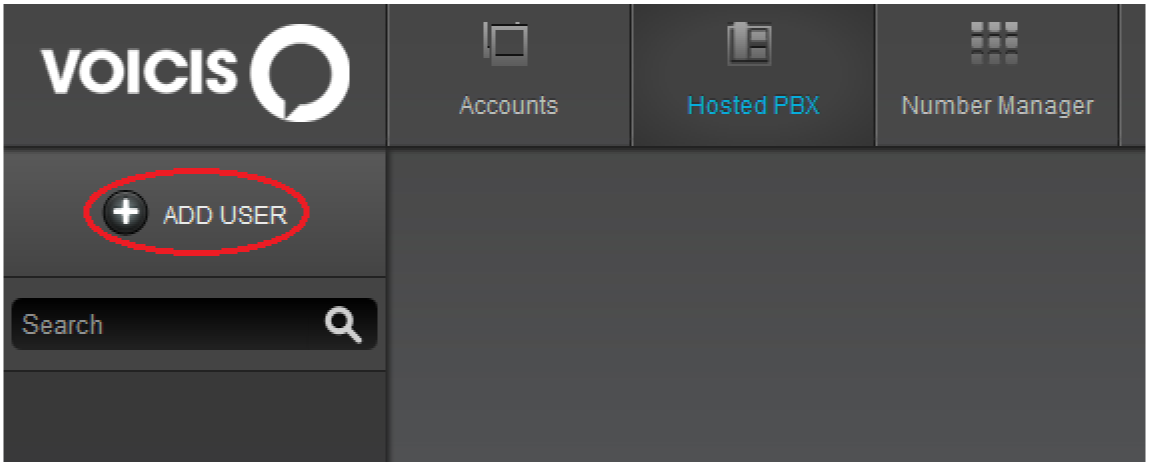
ENG - 2.1 - Administration

User add

Choose **Users** options:



> Choose **Add User**

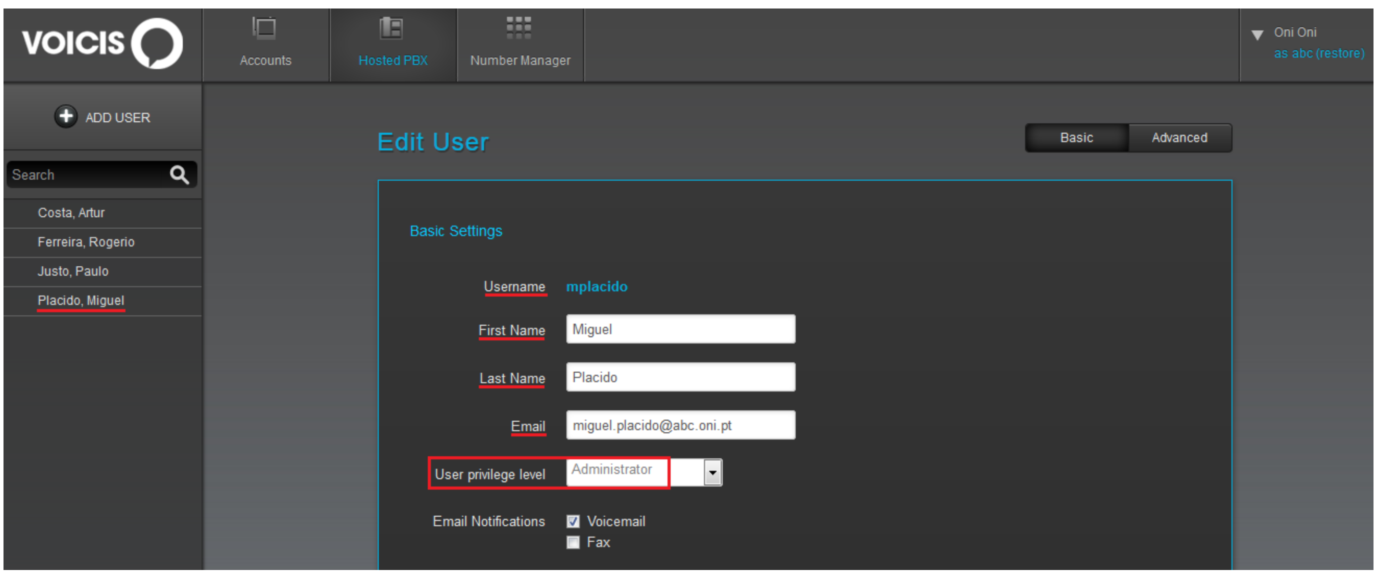


> Privilege Level: **Administrator**

Users configured with this privilege level are capable of total administration of this account and sub-accounts

Fill in the required fields: Username (key database), First Name, Last Name, Email (receive voice and fax notifications by attach, when applicable)

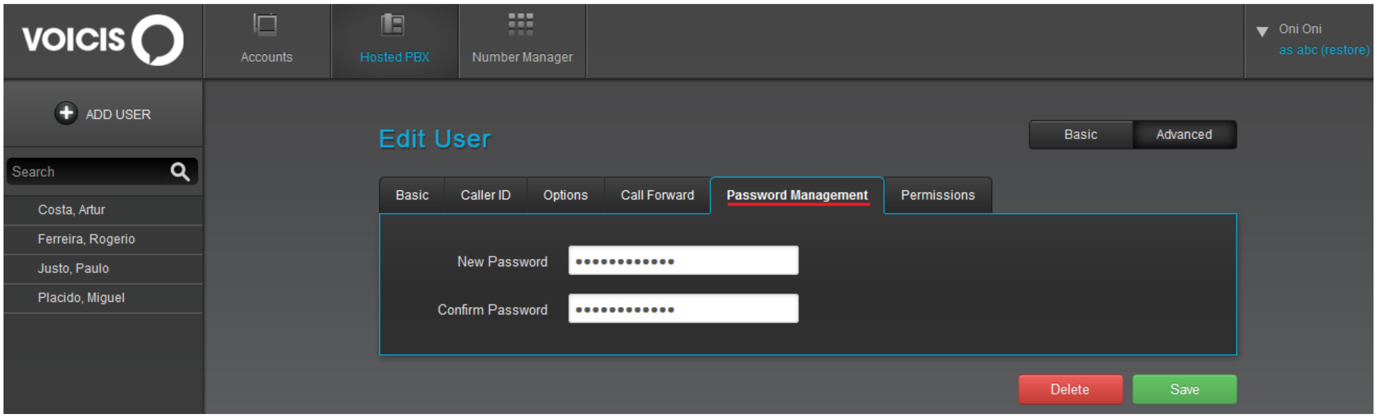
This information will be available in the corporate directory.



The configured information in Devices, Call Flow and User (this time) will be displayed in the corporate directory.

> Password Setup

The user will receive, via email, a password to access, automatically generated by the system.



App Store

Allocation of privilege after the user logs

The admin user of each account should endorse the applications available by clicking the right corner of the screen, in the **App Store**

Enable the following required options:

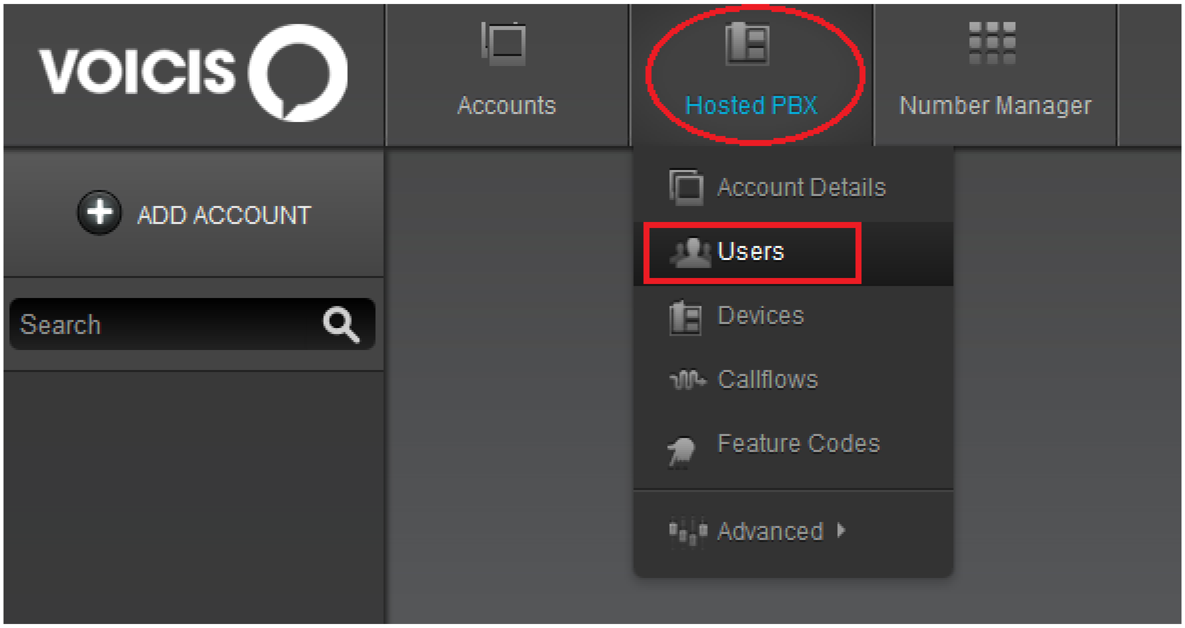
The different applications are:

APP	Description
Hosted PBX	Allows to configure users and telephony, devices, callflow, etc..
Accounts	Allows the creation of SubAccounts, only makes sense in the case of a service provider.
Number Manager	Allows the service provider to add a DDI to each Account or that the own account manager could acquire a DDI in the future

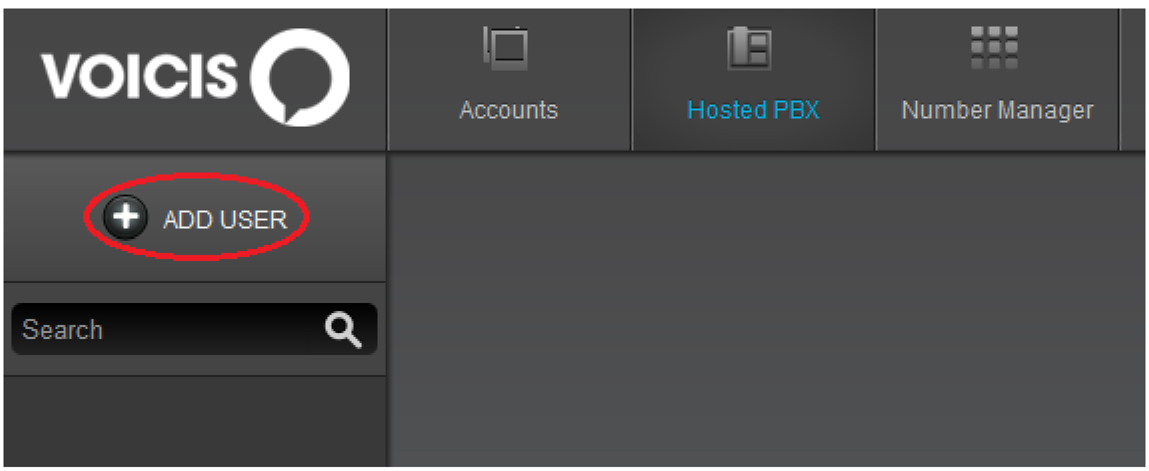
ENG - 2.2 - Devices

To add a user

Choose option (**users**):



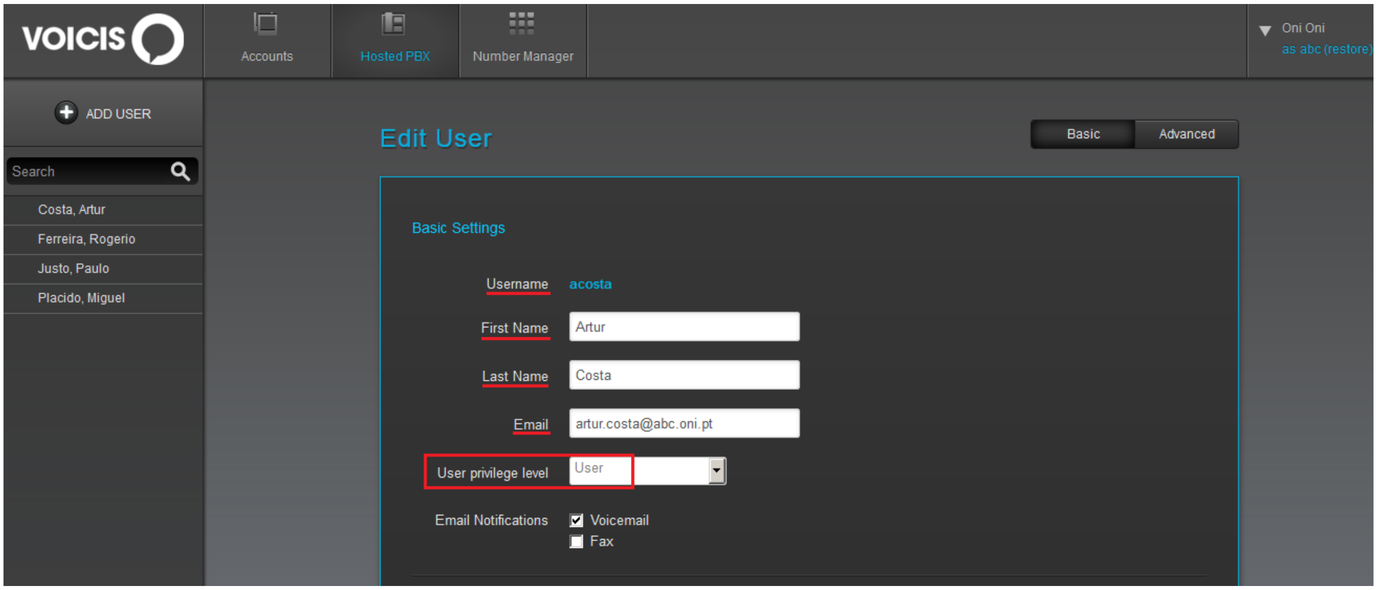
> Choose **Add USer**



> Privilege Level: **User**

Fill in the required fields: Username (key database), First Name, Last Name, Email (receive voice and fax notifications by attach, when applicable).

This information will be available in the corporate directory.

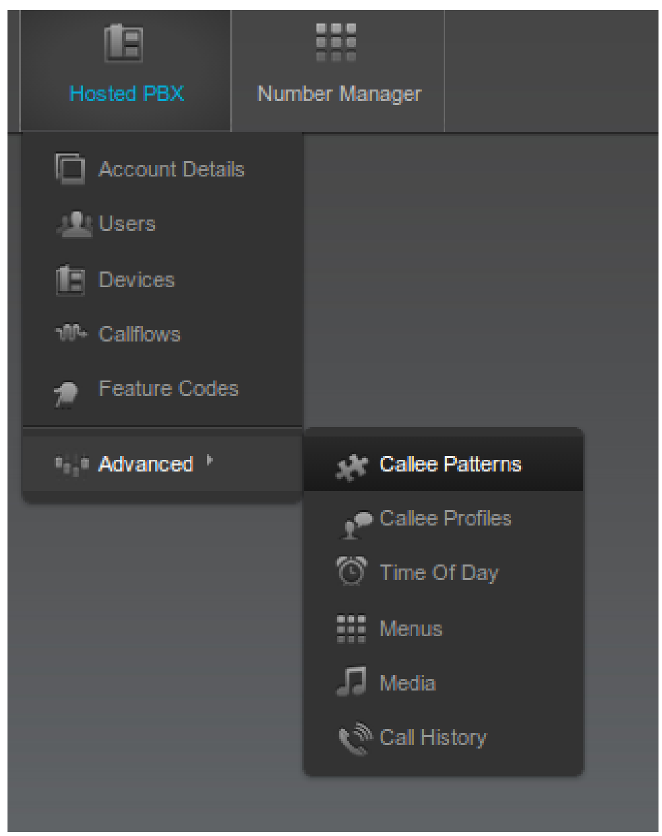


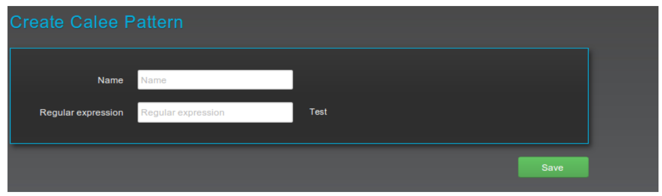
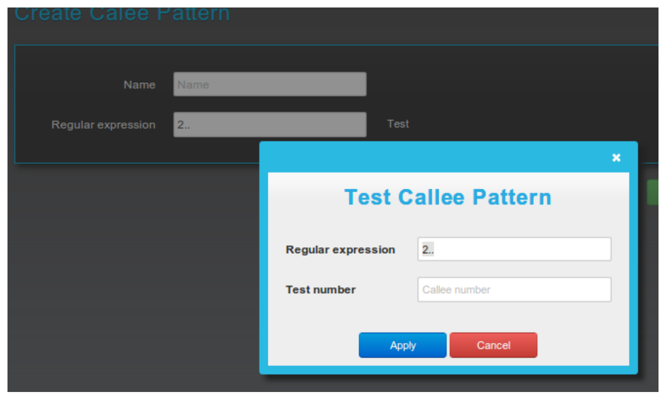
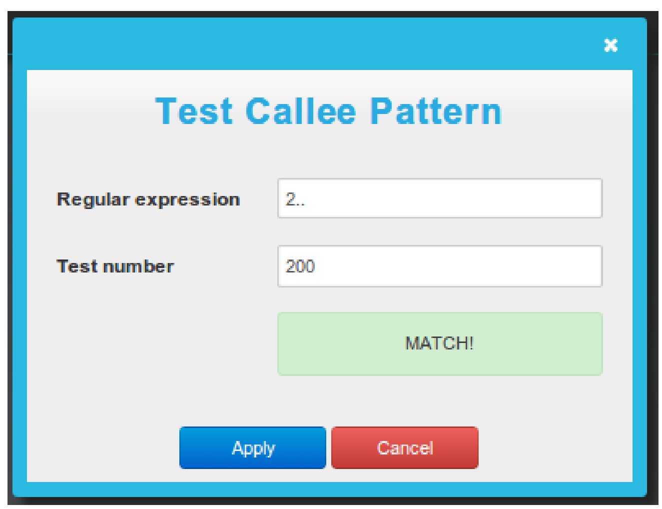
The configured information in Devices, Call Flow and User (this time) will be displayed in the corporate directory.

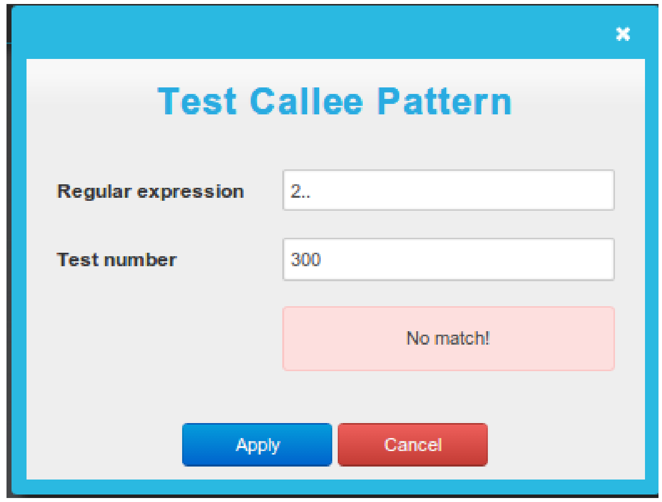
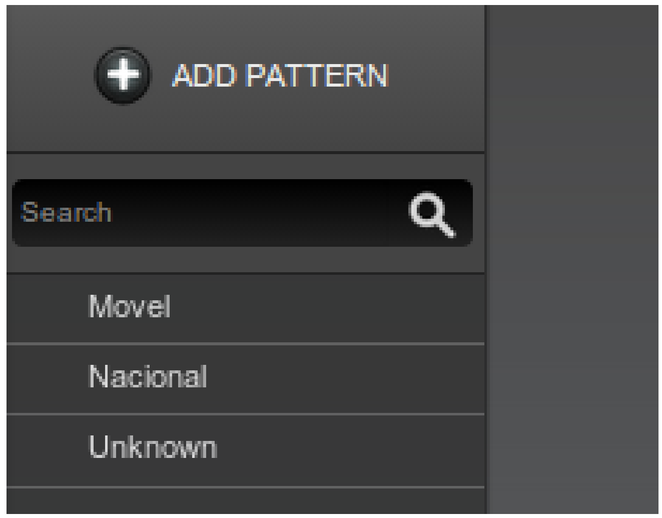
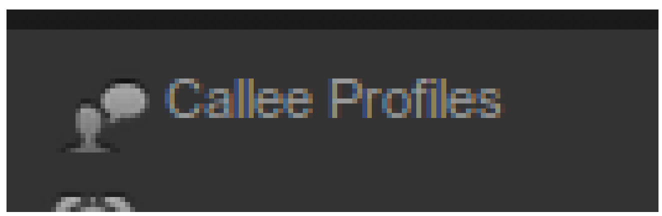
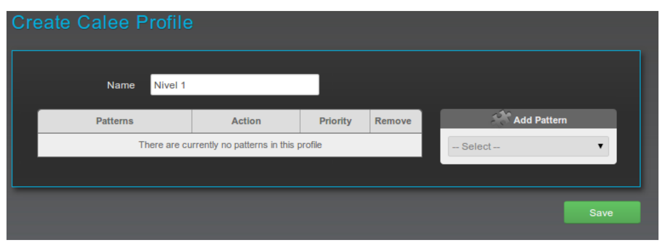
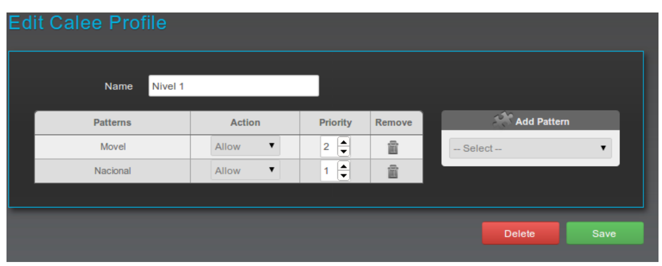
ENG - 3 - Phones Access levels

ENG - 3.1 - Phone Access Levels

To limit access to the outside we have to create Caller/Callee Pattens and Caller/Callee Profiles.

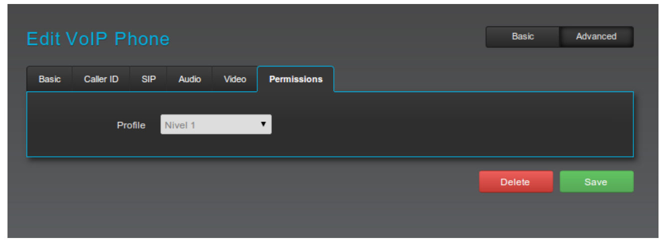
Action	Image
On the menu **Hosted PBX** choose the option **Advanced**	

Choose **Callee Patterns**	
Create a new **Callee Pattern** in **ADD PATTERN**	
Associate a name Associate a pattern using regular expression see http://pt.wikipedia.org/wiki/Express%C3%A3o_regular	
Choose **Test** and you can test it	
Test against Regular Expression 2.. with number 200 with positive results	

Test against Regular Expression 2.. with number 300 with negative results	
Once created various **Callee Patterns** In this example two: • National : **2........** • Mobile: **9........**	
Create a Callee Profile Callee where we associate. Choose **Calle Profiles.**	
Associate the **Callee Patterns** profile Choose a profile name	
In **Add Pattern** add the **Callee Patterns.** Indicate whether this is allowed or not (Allow, Deny) Caution: The priority is important.	

<table>
<tr>
<td>Associate the profile to the Device or to the User.

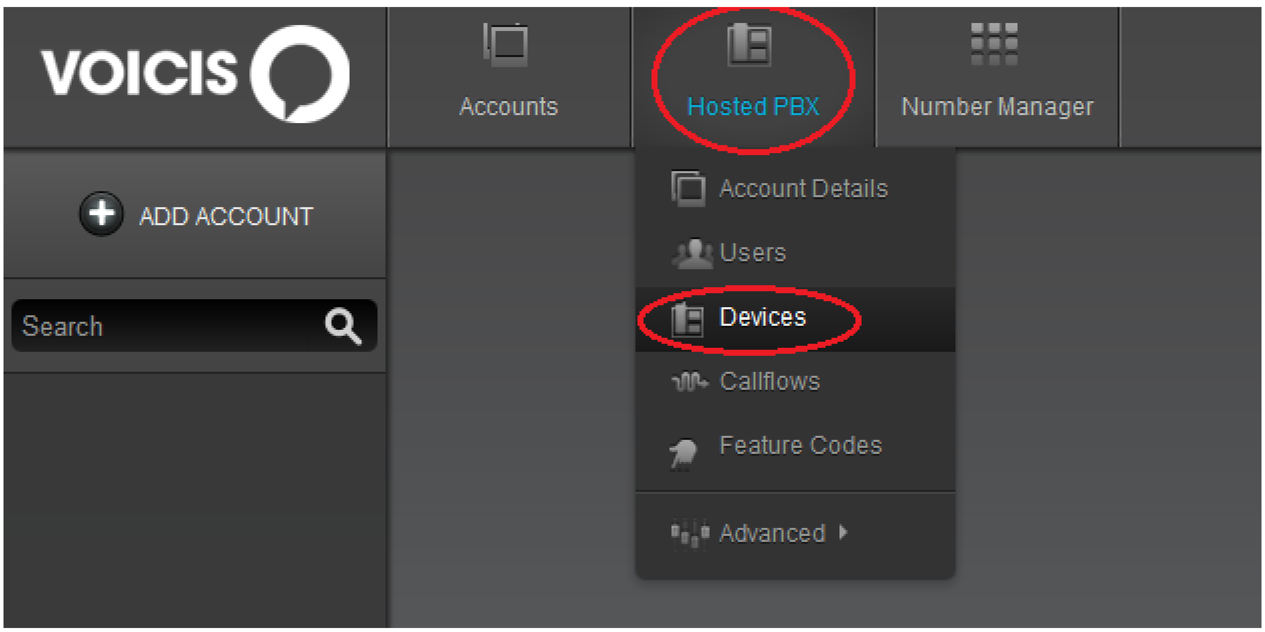
Choose Advanced (

)</td>
<td>
</td>
</tr>
</table>

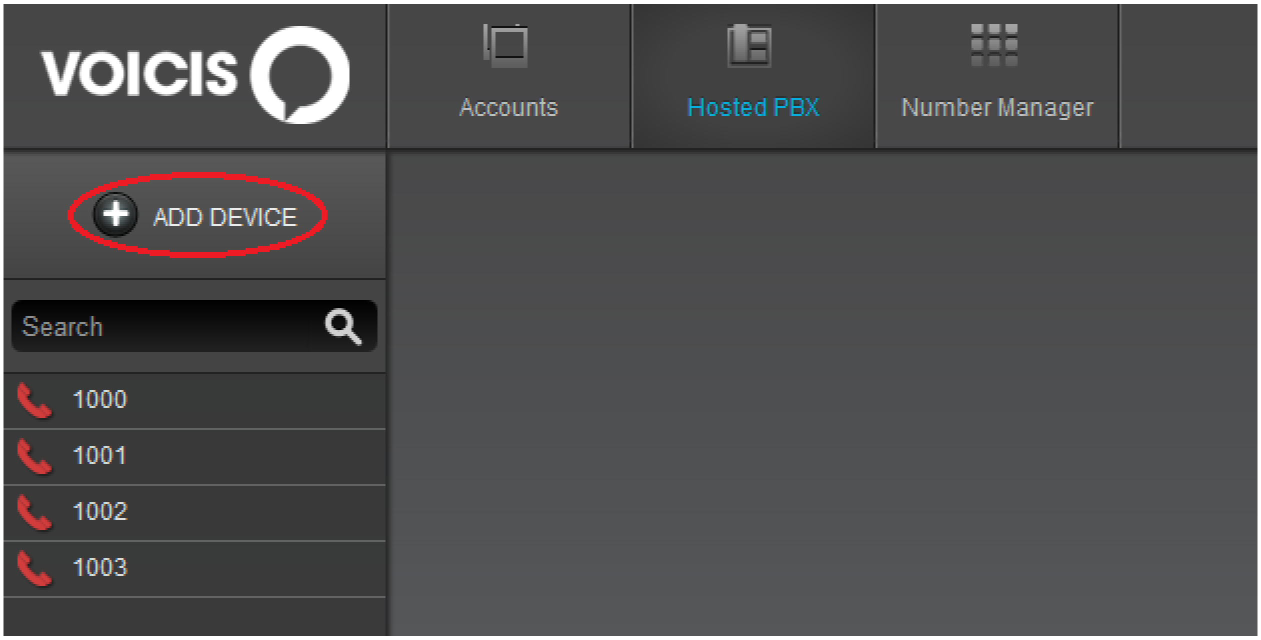
ENG - 4 - Phone configuration

Adding a phone

Choose the option **Devices**

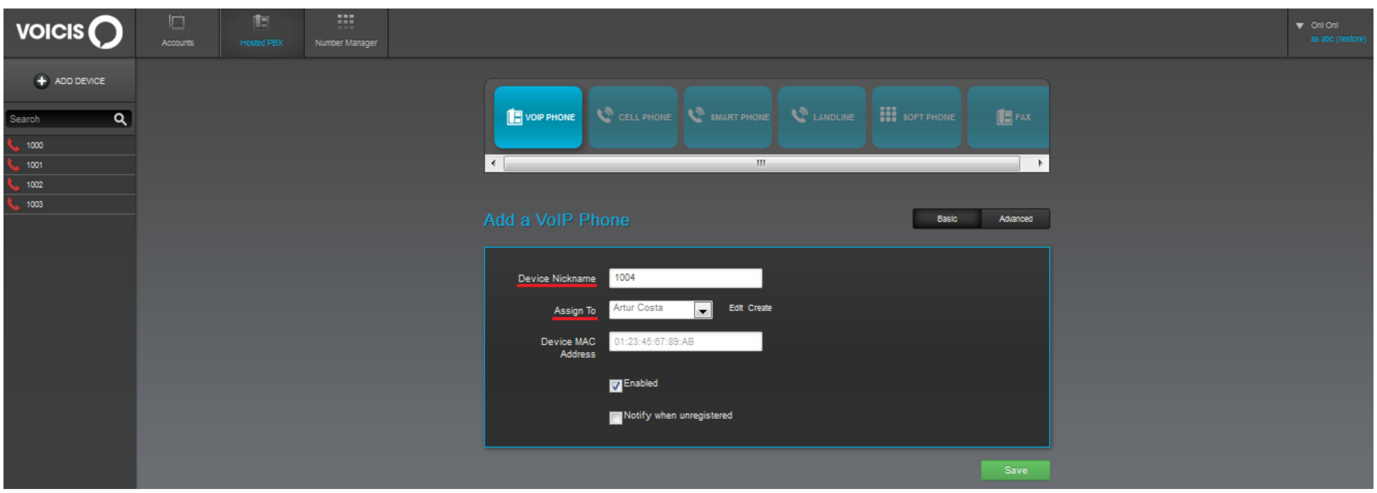


Choose **ADD DEVICE**



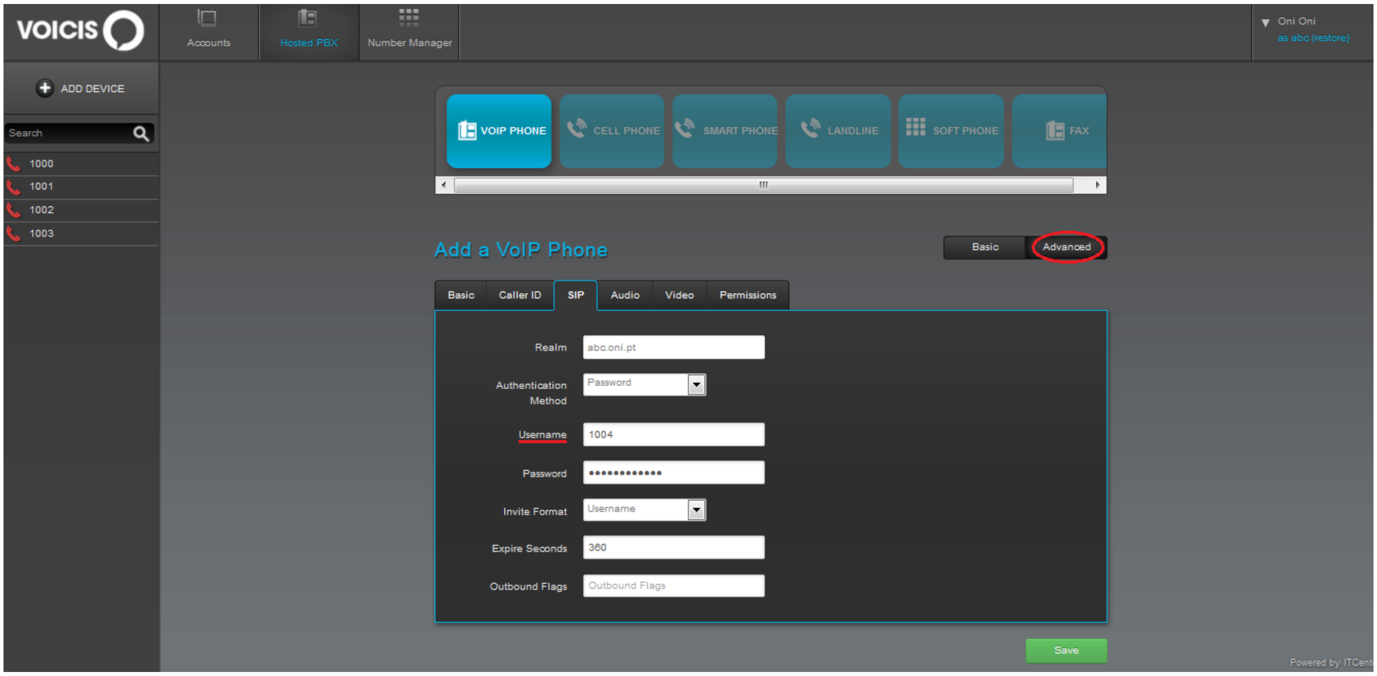
Choose the option **VOIP PHONE**

Fill in the required fields (Device nickname and a user in Assign To)



In the **Caller ID** tab, you don't need to fill in the respective fields, whether they are fulfilled in defining users (corporate directory).

Choose Advanced option to access information from SIP



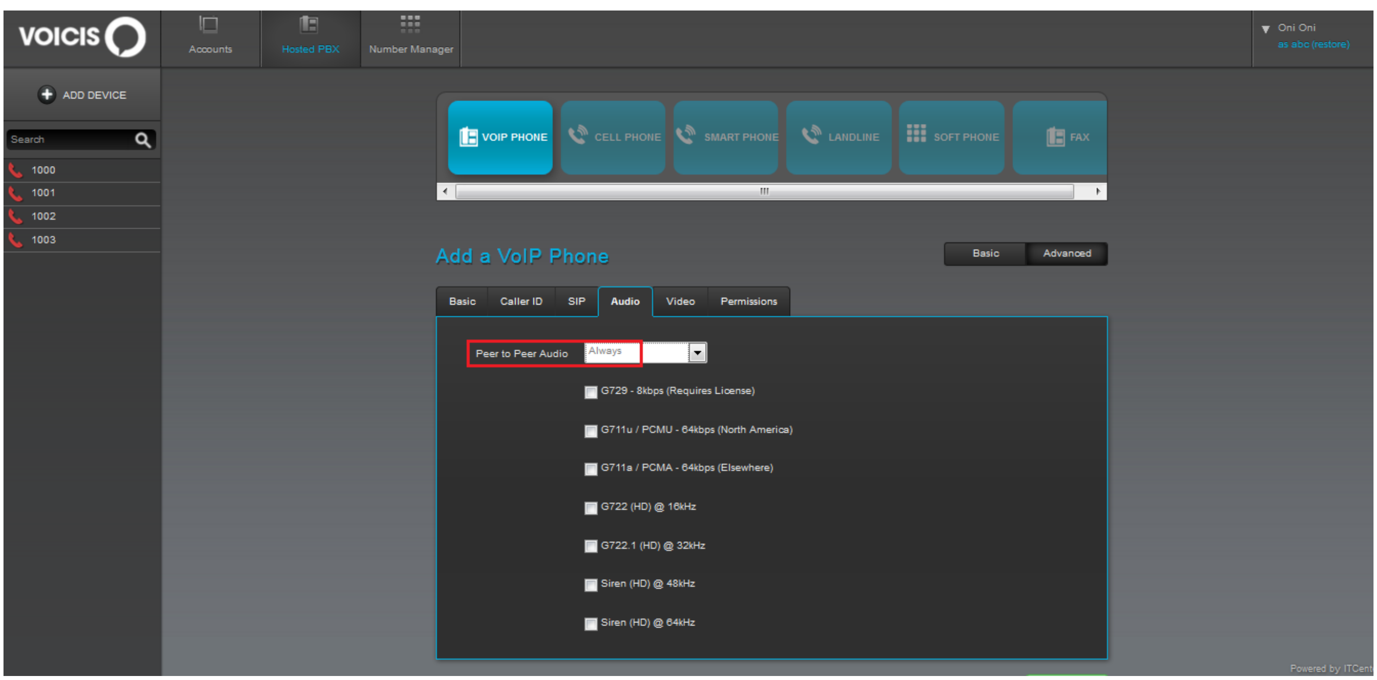
In the present, you must configure the password. In the future will be automatic

NOTE: The SIP Username must be equal to the Device Nickname.

Direct RTP

For the RTP to go directly from one phone to another we should choose the option of peer to peer Audio to **Always**.

Any of the other options RTP always passes via VoicisCore.



It is not necessary to define the codecs list.

The permissions do not need to be set because it has already been done in the configuration of users.

Phone Configuration

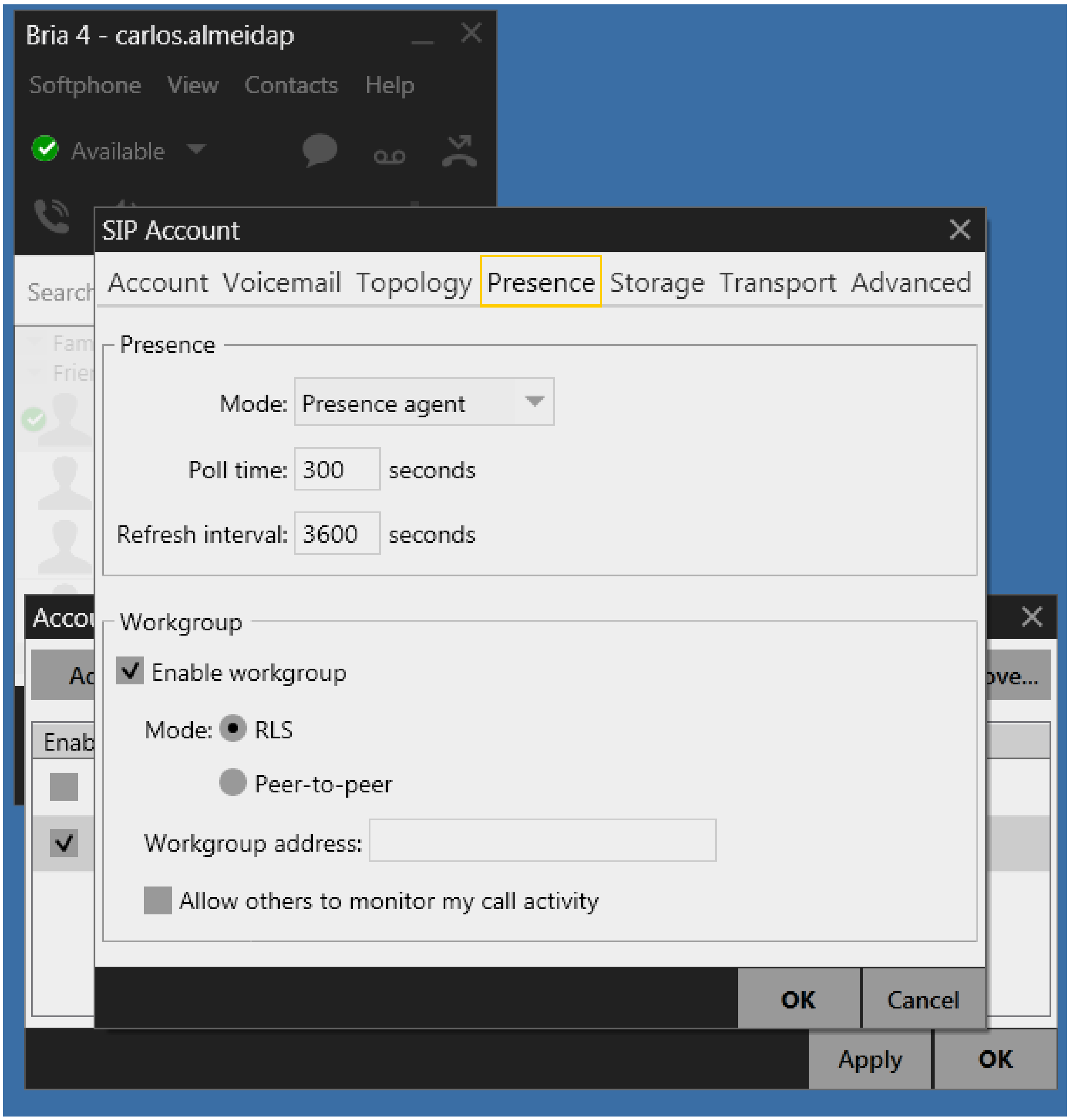
For the phone to be able to register in voiciscore up correctly, you should use the following fields on the client or SIP phone. The use of our DNS server is **REQUIRED** for me the phone to operate in our solution. The mechanism is based on DNS redundancy.

11.4.1 - Chat and Messaging

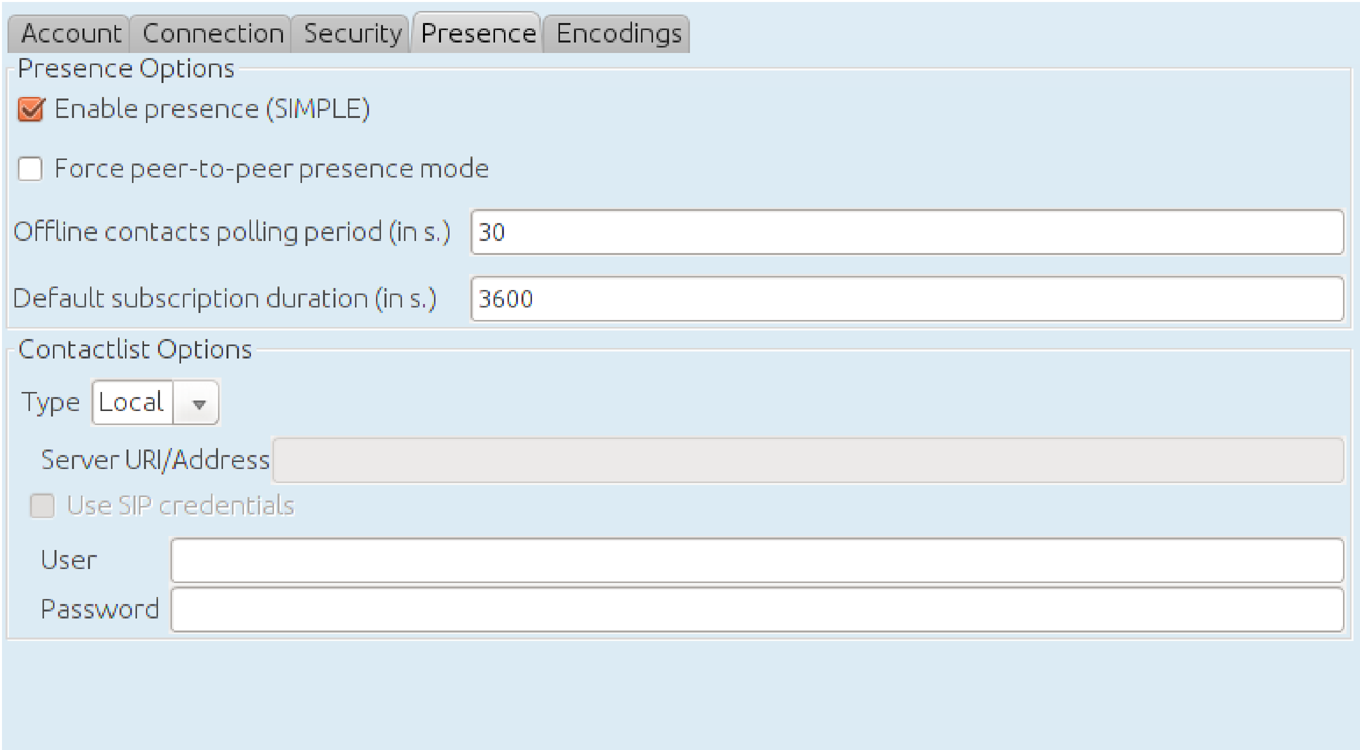
To use Chat and messaging you need need to enable PRESENCE SIP SIMPLE into the client configuration

Example:

For BRIA



For JITSI



ENG - M9 Handset Configuration

1 Access the Web interface of the M9 base

2 Choose one of the identities to setup the extension (Identity X)

3 Account Tab

- Identity active: ON
- Server type: No specific server type
- Display name: <Nome a apresentar no display do telefone>
- Account: Numero da extensão <sip username>
- Registrar: Dominio
- Outbound Proxy: Dominio
- Authentication Name: Numero da extensão <sip username>
- Password: password
- Password (repeat): password

  Choose Save

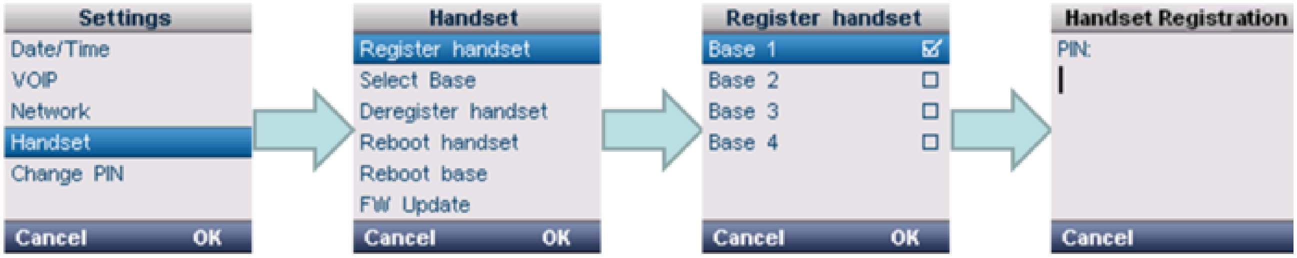
4 Sip Tab

- RTP Encryption: OFF
- Offer ICE: OFF

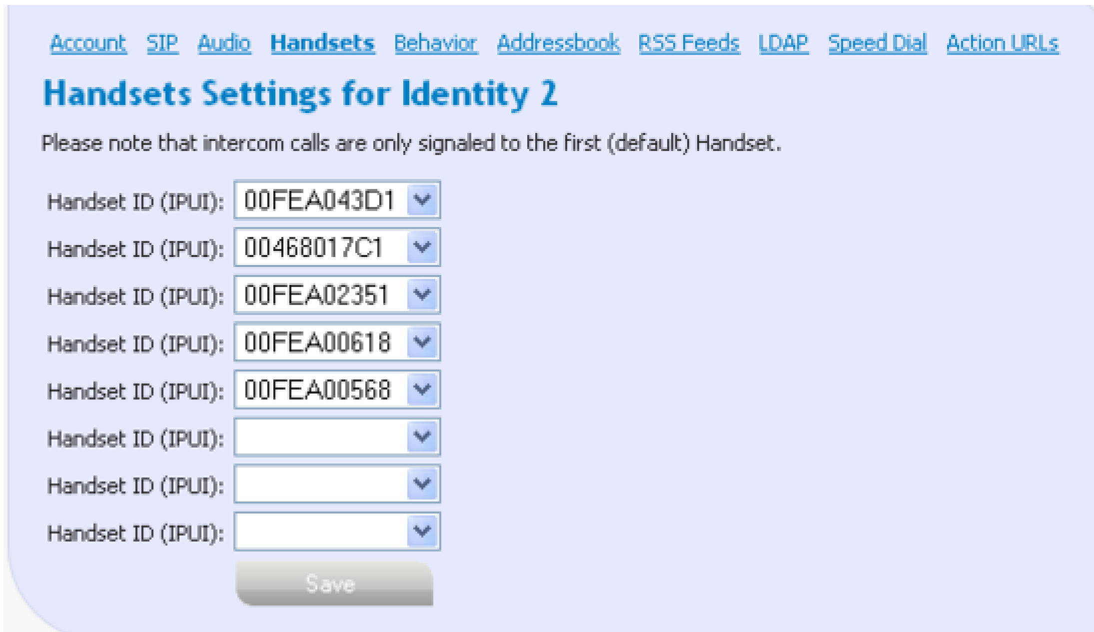
  Choose Save

5 Handsets

- You must start the permitting process for the registration of the DECT base station so that the base is visible to the equipment, it is necessary to go to the option:
  - DECT Tab
  - Choose **START** near the option where you can read "In order to allow handsets to register to the DECT base station, push the start button. After this, you will see the call LED light flash up. The registration will be open for approximately 10 minutes."
- After attempting to register with the m9 handset with the base it will be available in the IPUI list



- Note: Default PIN is 0000
- Choose Handset ID

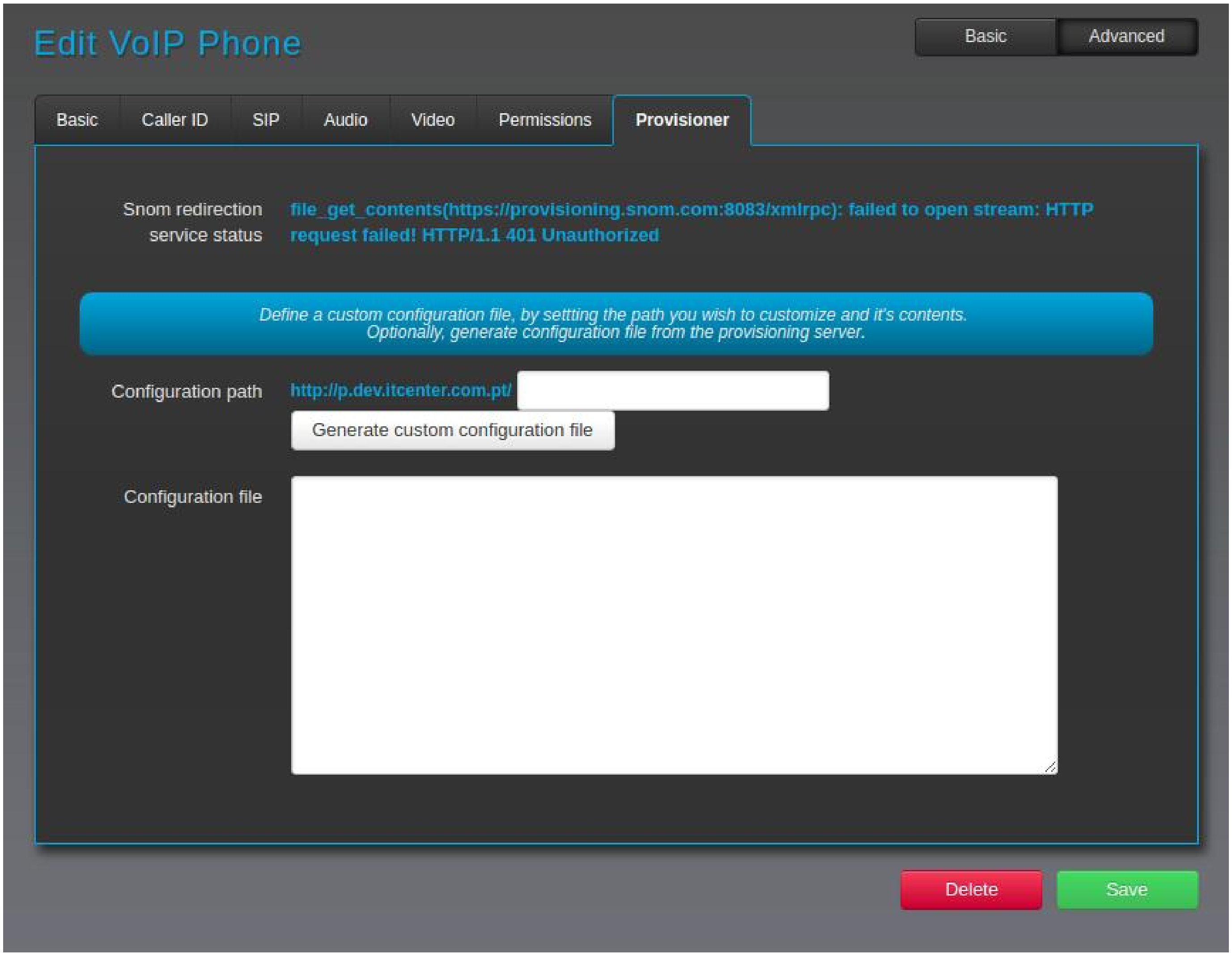


Choose Save

Registering Handset for Snom M300

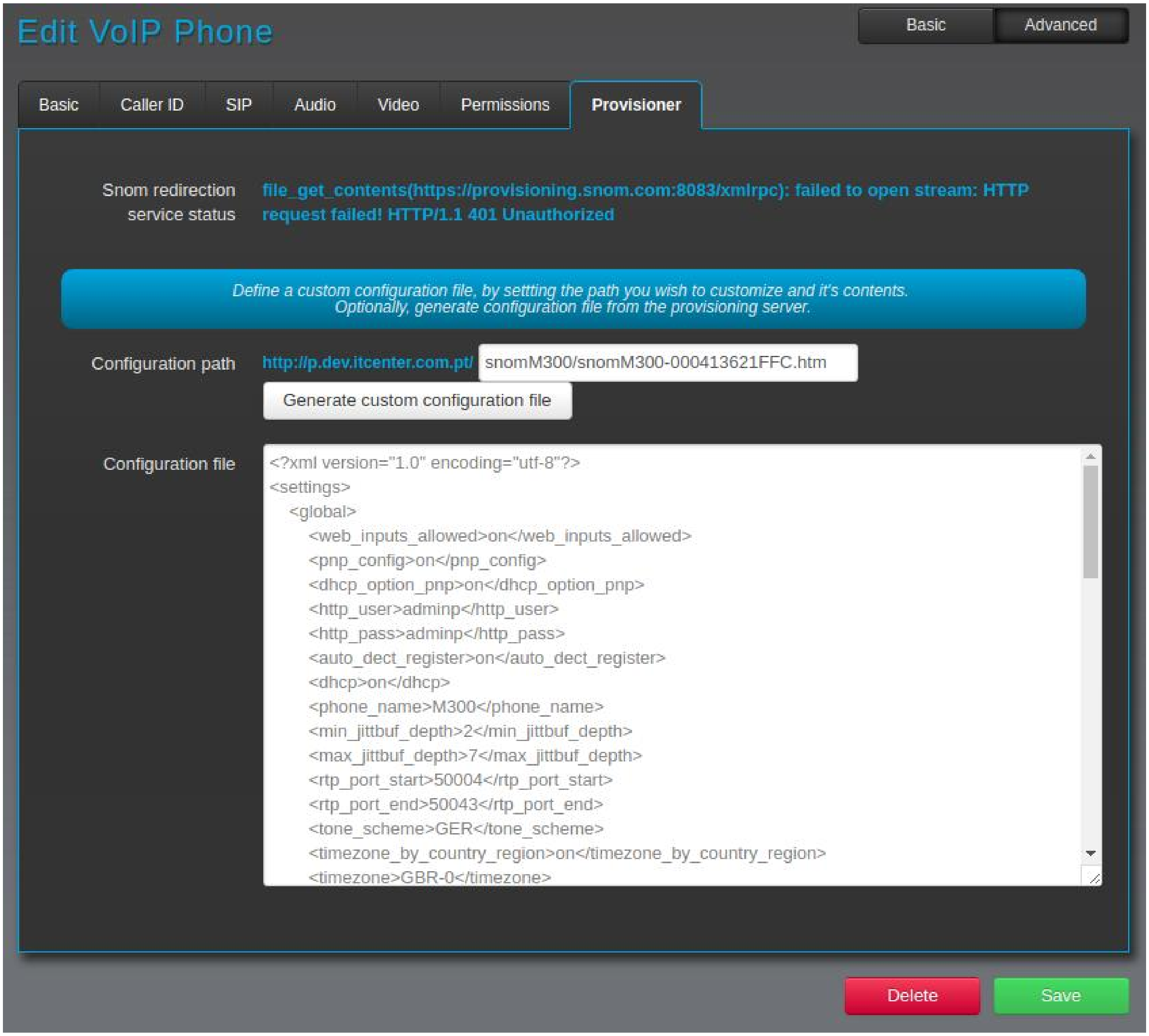
Snom M300 can plug several DECT handsets (M25, or either M85 and M65), and this procedure can be done by web interface or by provisioning xml, generated in Voicis Core platform, that can be edited before importing into Snom M300 base station.

Login into Voicis Core, select Account, and under Hosted PBX > Devices, select Device. Open Advanced tab and Provisioner sub-tab, as the following image shows:



Enter snomM300/snomM300-{mac}.htm in the "Configuration path" text box. Then click "Generate custom configuration file".

URL
Replace {mac} with the MAC address of the Snom M300 base station that will be provisioned. All caps and without colon (:)



The textarea "Configuration file" will be fulfilled with the xml code for provisioning Snom M300. To add Handsets to the xml code, the IPEI from the handset(s) must be provided.

Find IPEI in Handset
The IPEI can be found under Menu > Settings > Status (scroll down until IPEI is visible)

The following example xml is only for tutorial purposes. It must be always automatically generated.

For handset configuration part refer to "<!-- Handset Configuration -->"

Add Handsets to Provisioning XML

```
<?xml version="1.0" encoding="utf-8"?>
<settings>
    <global>
        <web_inputs_allowed>on</web_inputs_allowed>
        <pnp_config>on</pnp_config>
        <dhcp_option_pnp>on</dhcp_option_pnp>
```

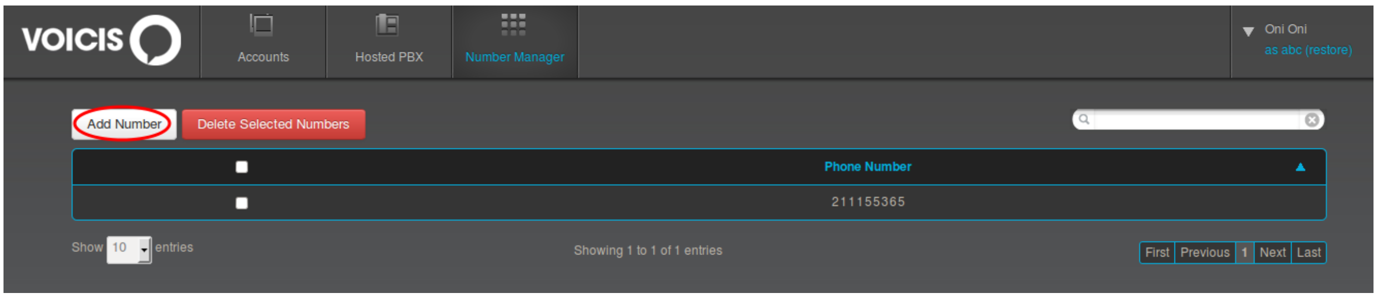
```
        <http_user>adminp</http_user>
        <http_pass>adminp</http_pass>
        <auto_dect_register>on</auto_dect_register>
        <dhcp>on</dhcp>
        <phone_name>M300</phone_name>
        <min_jittbuf_depth>2</min_jittbuf_depth>
        <max_jittbuf_depth>7</max_jittbuf_depth>
        <rtp_port_start>50004</rtp_port_start>
        <rtp_port_end>50043</rtp_port_end>
        <tone_scheme>GER</tone_scheme>
        <timezone_by_country_region>on</timezone_by_country_region>
        <timezone>GBR-0</timezone>
        <dns_server1>192.168.203.45</dns_server1>
        <dns_server2>192.168.203.46</dns_server2>
        <dst_by_country_region>on</dst_by_country_region>
        <dst_enable>auto</dst_enable>
        <dst_fixed_day_enable>on</dst_fixed_day_enable>
        <dst_start_month>3</dst_start_month>
        <dst_start_date>0</dst_start_date>
        <dst_start_time>2</dst_start_time>
        <dst_start_day_of_week>1</dst_start_day_of_week>
        <dst_start_wday_last_in_month>2</dst_start_wday_last_in_month>
        <dst_stop_month>11</dst_stop_month>
        <dst_stop_date>0</dst_stop_date>
        <dst_stop_time>2</dst_stop_time>
        <dst_stop_day_of_week>1</dst_stop_day_of_week>
        <dst_stop_wday_last_in_month>1</dst_stop_wday_last_in_month>
        <language>Português</language>
    </global>
    <phone-settings e="2">
        <ntp_server>ntp.dev.itcenter.com.pt</ntp_server>

 <!-- Account/User Account 1 -->
  <!-- Handset Configuration -->
  <!-- idx for handset must match idx of the user idx -->


  <subscr_dect_ipui idx="1">FFFFFFFFFF</subscr_dect_ipui> <!-- IPEI -->
  <subscr_sip_hs_idx idx="1">1</subscr_sip_hs_idx> <!-- Account/User
Account to be associated with -->
  <subscr_sip_ua_data_server_id idx="1">1</subscr_sip_ua_data_server_id>
<!-- Account Server to be associated with -->
  <subscr_sip_ua_pref_outg_sip_id
idx="1">1</subscr_sip_ua_pref_outg_sip_id> <!-- Account Outbound Server to
be associated with -->
  <subscr_sip_line_name idx="1">Line1</subscr_sip_line_name> <!-- Handset
Line Name -->

        <srv_srtp_auth idx="1">off</srv_srtp_auth>
        <subscr_sip_hs_idx idx="1">1</subscr_sip_hs_idx>
        <subscr_sip_ua_data_server_id
idx="1">1</subscr_sip_ua_data_server_id>
        <subscr_sip_ua_pref_outg_sip_id
```

```
idx="1">1</subscr_sip_ua_pref_outg_sip_id>
        <subscr_sip_line_name idx="1"></subscr_sip_line_name>
        <user_host idx="1">dev.itcenter.com.pt</user_host>
        <user_outbound idx="1">dev.itcenter.com.pt</user_outbound>
        <user_dtmf_info idx="1">off</user_dtmf_info>
        <srv_sip_server_alias idx="1">Voicis</srv_sip_server_alias>
        <user_srtp idx="1">off</user_srtp>
        <user_active idx="1">on</user_active>
        <user_pname idx="1">1086</user_pname>
        <user_name idx="1">1086</user_name>
        <user_pass idx="1" perm="">1234567</user_pass>
        <user_realname idx="1">david goncalves</user_realname>
        <user_expiry idx="1">3600</user_expiry>
        <user_dtmf_info idx="1" perm="">off</user_dtmf_info>
        <codec_priority_list
idx="1">pcmu,pcma,g729,telephone-event</codec_priority_list>
```

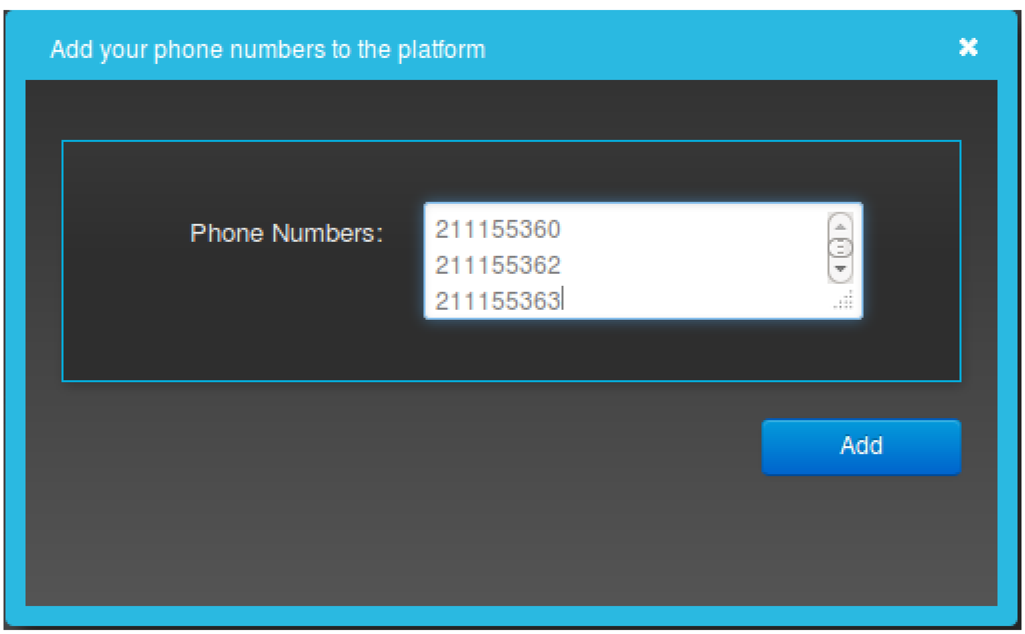
```
    </phone-settings>
</settings>
```

ENG - 6 - DDIs Provisioning

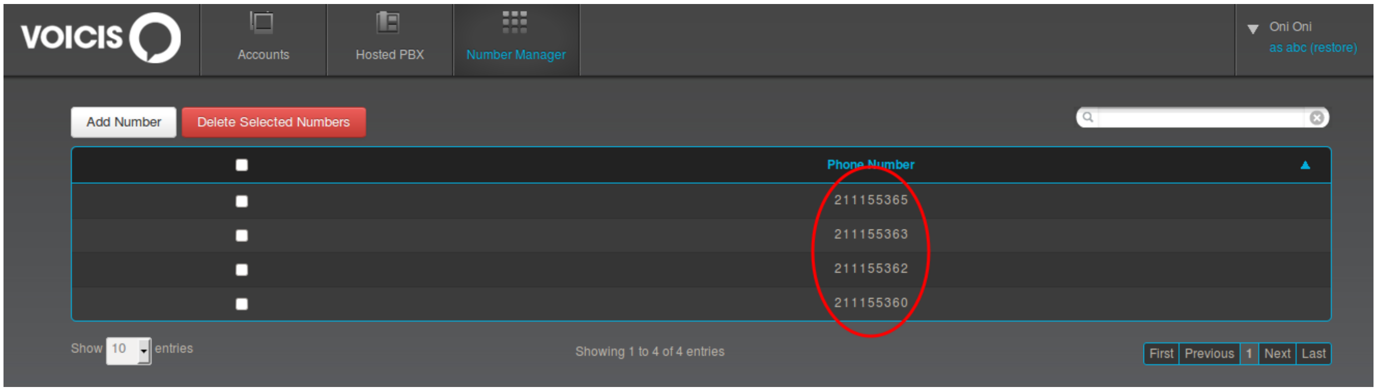
1. Add DDIs



2. Set all DDIs assigned to the client (account)



The DDIs are available to be assigned to employees of the company.



ENG - 7 - System Features

ENG - 7.1 - Call Transfer

1. Direct

- Extension **"A"** calls to extension **"B"**
- **"B"** picks up call
- **"B"** presses ***Transfer***
- **"B"** calls to extension **"C"** and presses check (?)
- **"A"** the call is automatically and immediately transferred

2. With consultation

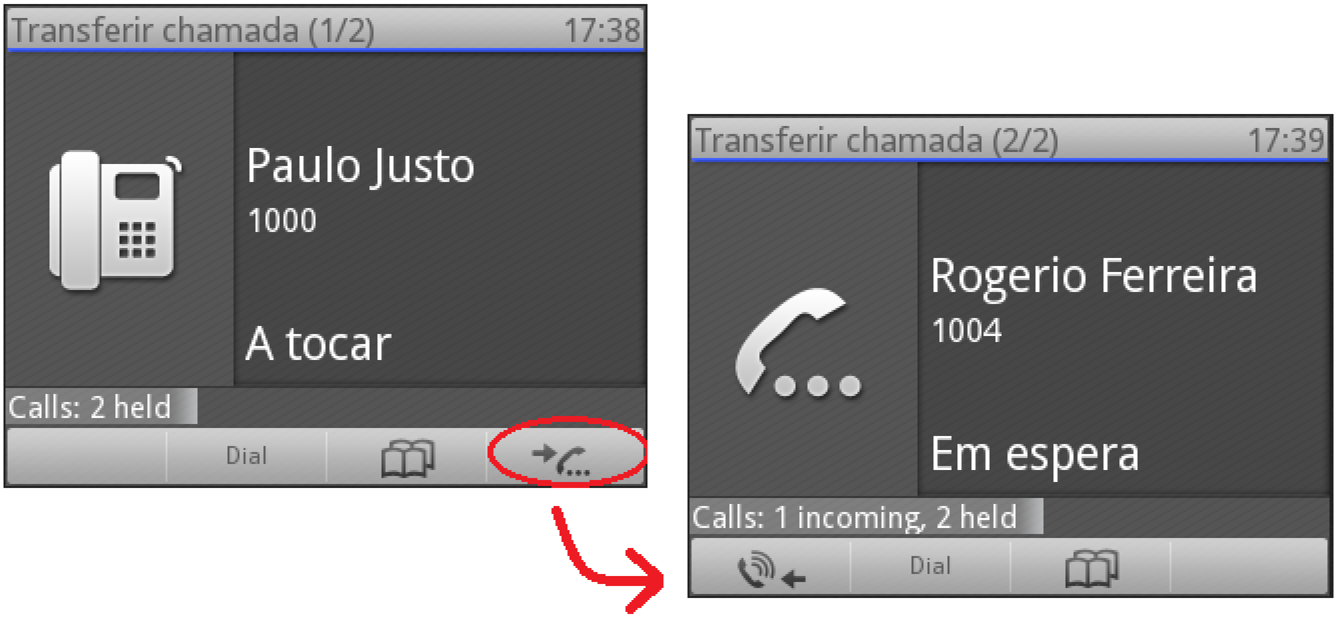
1 - <u>With one single call</u>

- **"A"** calls **"B"**
- **"B"** picks up call
- **"B"** puts the call on Hold
- **"B"** calls **"C"**
- **"C"** picks up and speaks with **"B"**
- **"B"** presses the key with the call on hold where **"A"** is, and reports that is going to transfer the call to **"C"**
- **"B"** presses the key ***Transfer,*** and selects the call to transfer on the screen
- **"A"** and **"C"** can now speak

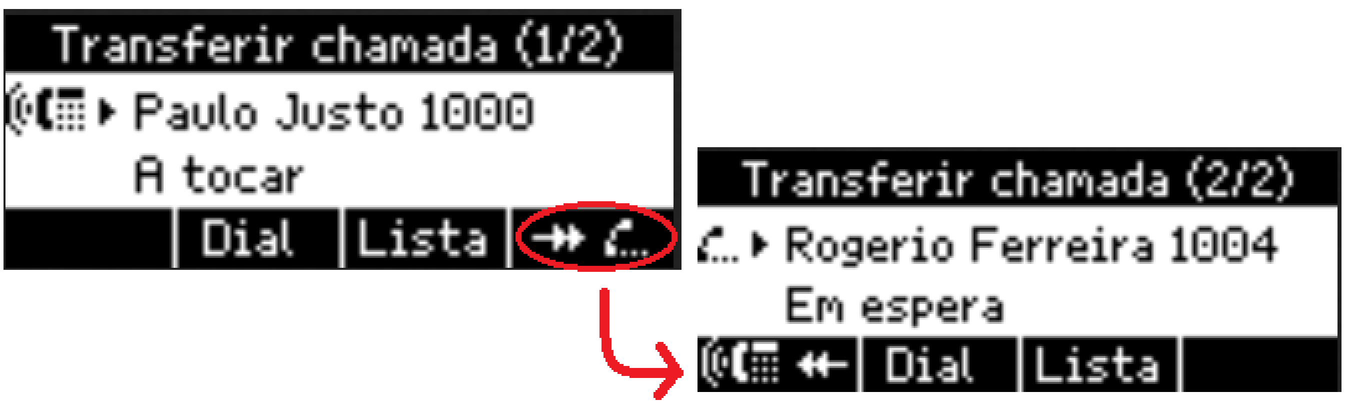
2 - <u>With calls on hold</u>

- **"A**" calls **"B"**
  - (meanwhile **"B"** receives a second call)
- **"B"** picks up **"A"**
- **"B"** puts **"A"** call on Hold
- **"B**" calls **"C"**
- **"C**" picks up and speaks with **"B"**
- **"B**" presses the key where "**A**" is, reports that is going to transfer the call to **"C"**
- **"B**" presses ***Transfer,*** selects the call to transfer on the screen and presses check (?)

(Model 760)



(Model 720)



- **"B"** is available to answer the second call

ENG - 7.2 - Conference

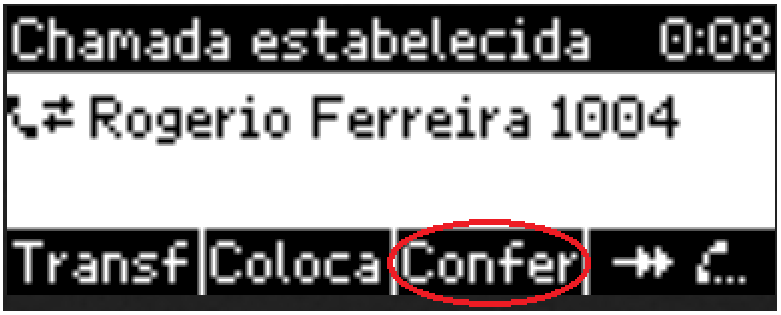
1. Conference

- Extension **"A"** calls to **"B"**
- Extension **"A"** puts **"B"** on HOLD

- Extension **"A"** calls to **"C"**
- Extension **"A"** calls in "CONFER" or

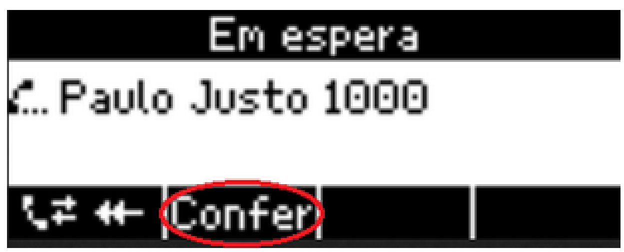
- **"A"**, **"B"** and **"C"** are now in Conference

1.1 Images Snom 710

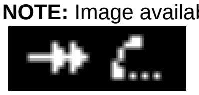


NOTE: Image available on main screen

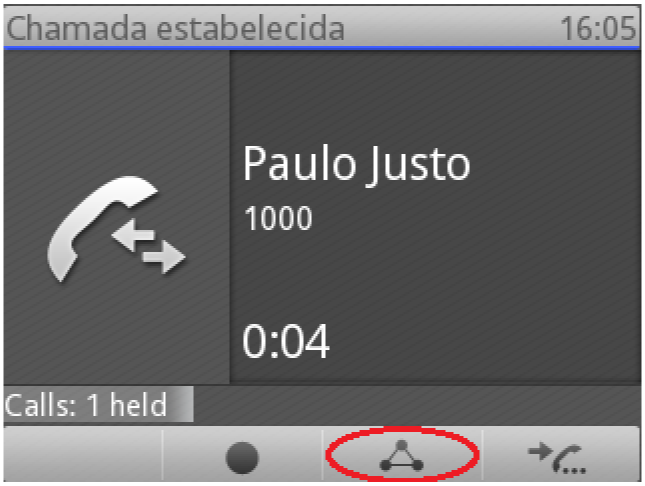
1.2 Images Snom 720



NOTE: Image available on second screen after pressing



1.3 Images Snom 760



NOTE: Image available on main screen

ENG - 7.3 - Call Pickup

1. Call Pickup with BLF

IMPORTANT: Only operates with BLF

- Extension **"A"** calls to **"B"**
- During, **"C"** (who has BLF of **"B"**) presses the BLF and captures the call from **"A"** to **"B"**.

2. Creating Devices Groups

VoIP Services > Advanced > Groups

1. Add Group
2. Name grupo
3. Add Devices
4. Save

3. Creating Callflow "Group Pickup"

Allows to capture calls that are ringing in one of the endpoints of the pickup group with a determined number of Pickup Call.

Note: You can only pickup calls in a group where the device you want to capture belongs to the same group.

VoIP Services > Callflows

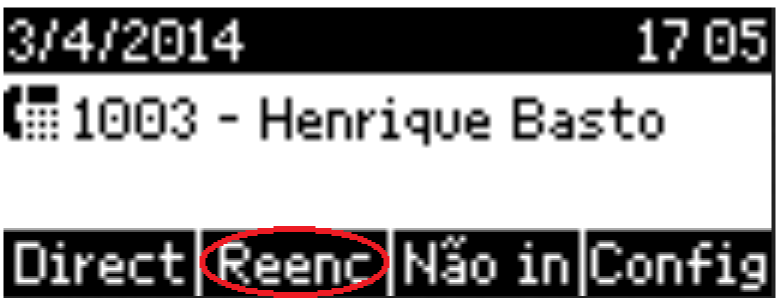
1. Add Callflow
2. Define name of the Callflow
3. "Click to add number" to associate to pickup call number
4. Actions > Advanced > Drag "Group Pickup" to Callflow
5. Select Group of Endpoints or Endpoint to associate to the Pickup Group

6. Save Changes

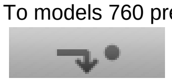
ENG - 7.4 - Call Forwarding

1. Activate Unconditional Forwarding

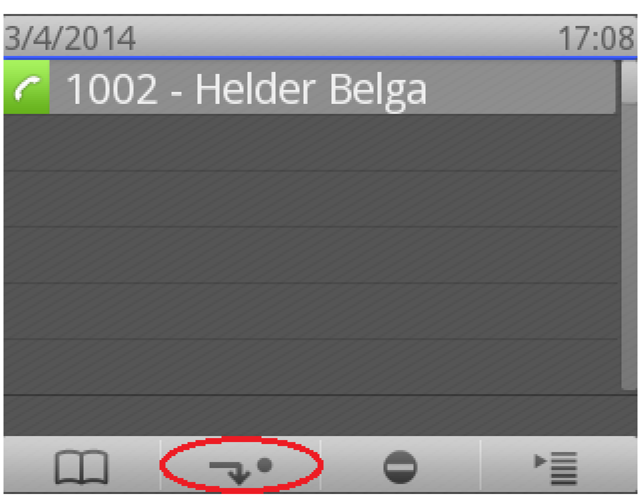
- To models 710/720 press **"Reenc"** and dial destination number followed by check (?)



- To models 760 press

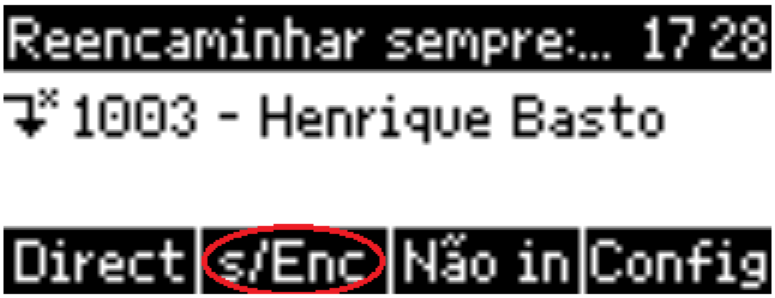


, dial destination number followed by check (?)

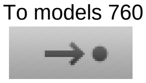


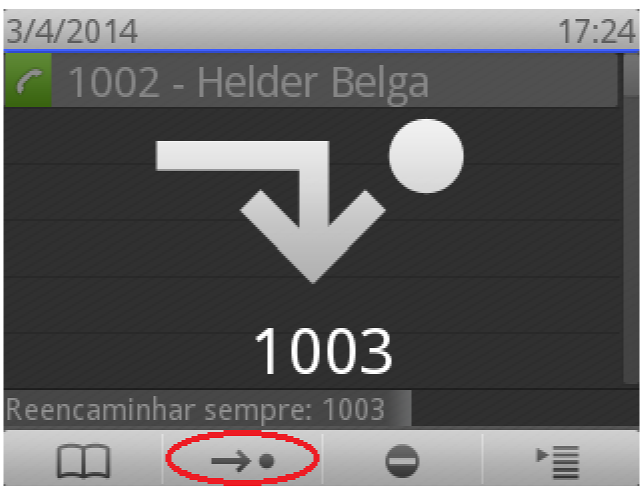
2. Deactivate Unconditional Forwarding

- To models 710/720 press **"s/Enc"**



- To models 760 press





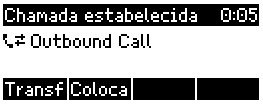
ENG - 7.4 - Callfoward on Busy

This feature allows to forward a call without answering the same, using BLF keys or pressing the number of the extension manually.

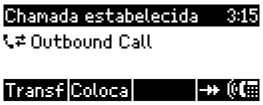
When we're having a phone call or when we receive a new call appears at the phone screen the option "**Transf**".

Inbound Call

Outbound Call



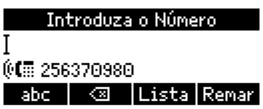
When we are with a call at some moment, we know we have a new incoming call because of the sign on the earphone. Also because of the icon that appears on the right of the phone screen.



By selecting the phone icon we're selection the new Incoming call.

By selecting "**Transf**" we have the possibility to transfer the call.

With navigation keys of the phone we choose the call we want to forward.



Finally, choose the destination, or manually placing the number or pressing a BLF key.

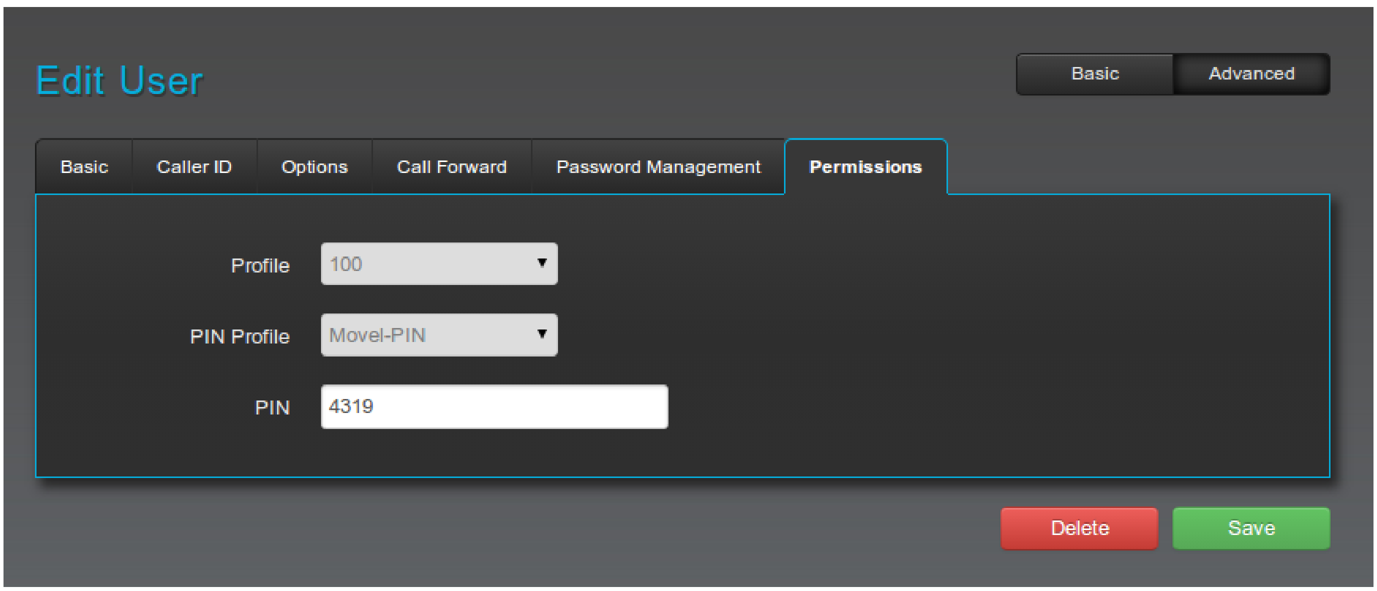
ENG - 7.5 - Authorization codes

This function allows a user to place a call to a destination that will be usually blocked,

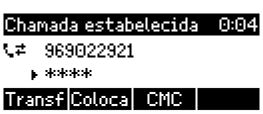
To this we associate a pattern of numbers to a user. Using the PIN of the user the call can be made.

First we have to create a **Callee Profile (3 - Phones Access levels)**.

After creating a **Callee Profile** is necessary to associate the same to the user.



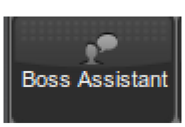
When we enter a number associated with the Pin Profile route, we receive a message that we have to enter the PIN to be able to continue the call.



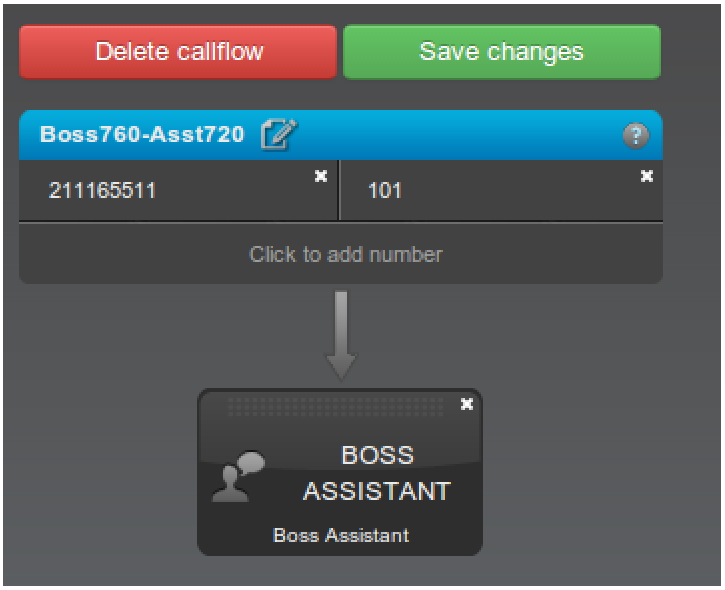
ENG - 7.6 - Boss Assistant

The Boss Assistant is a feature that allows you to forward calls from a particular source to a wizard.

To configure a Boss Assistant you have to create a Callflow and add the icon **Boss Assistant**.



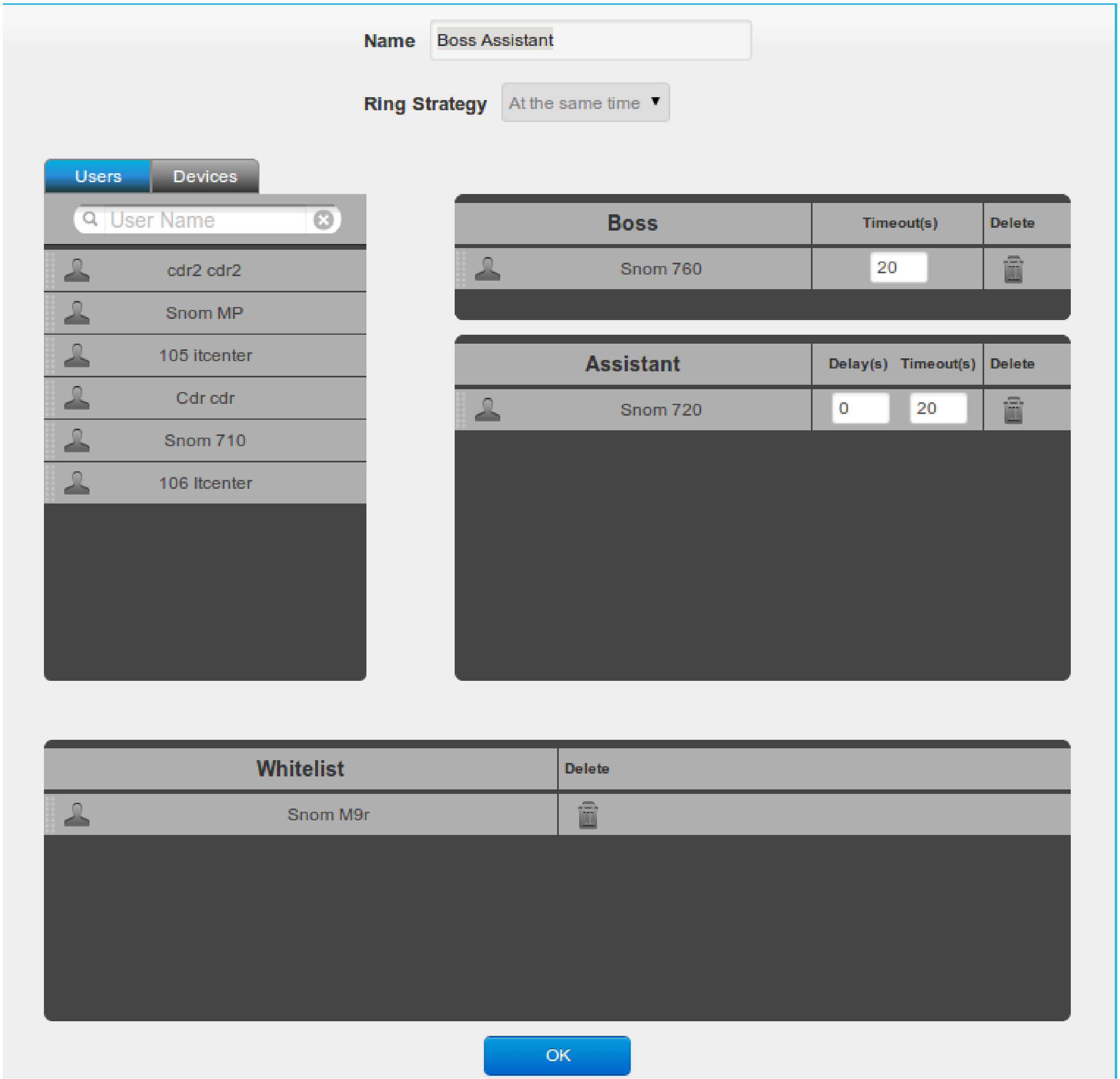
Example of Callflow with **Boss Assistant** used.



Then we have to select the **Boss Assistant** to configure filters and associated destinations.

Name	Description
BOSS	Call Destination whose origin is in the whitelist table
Assistant	Call Destination whose origin is not in the whitelist table
Whitelist	Source list

Configuration screen of the **Boss Assistant**.

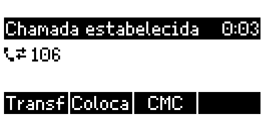


ENG - 7.7 - Call Matter Code

This function allows you to associate a code with a call for future treatment.

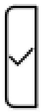
This code is included in the CDRs of the call. For this we have the established call.

Call in progress.

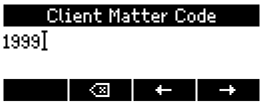


Select the CMC option.

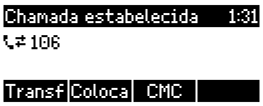
Introduce the value you want and press check key (



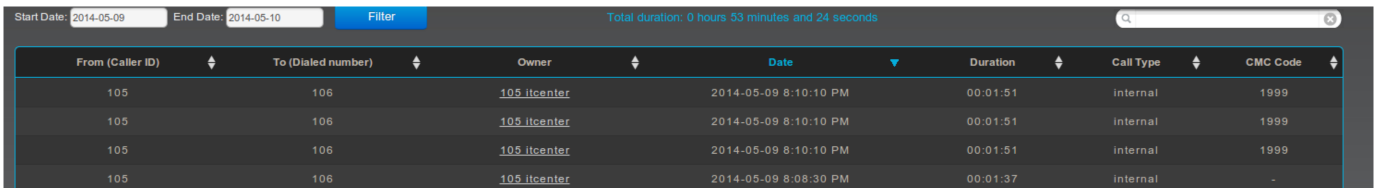
)



The CDR for this call will be marked with the value of the CMC.



View from the CDRs with the value of the CMC.

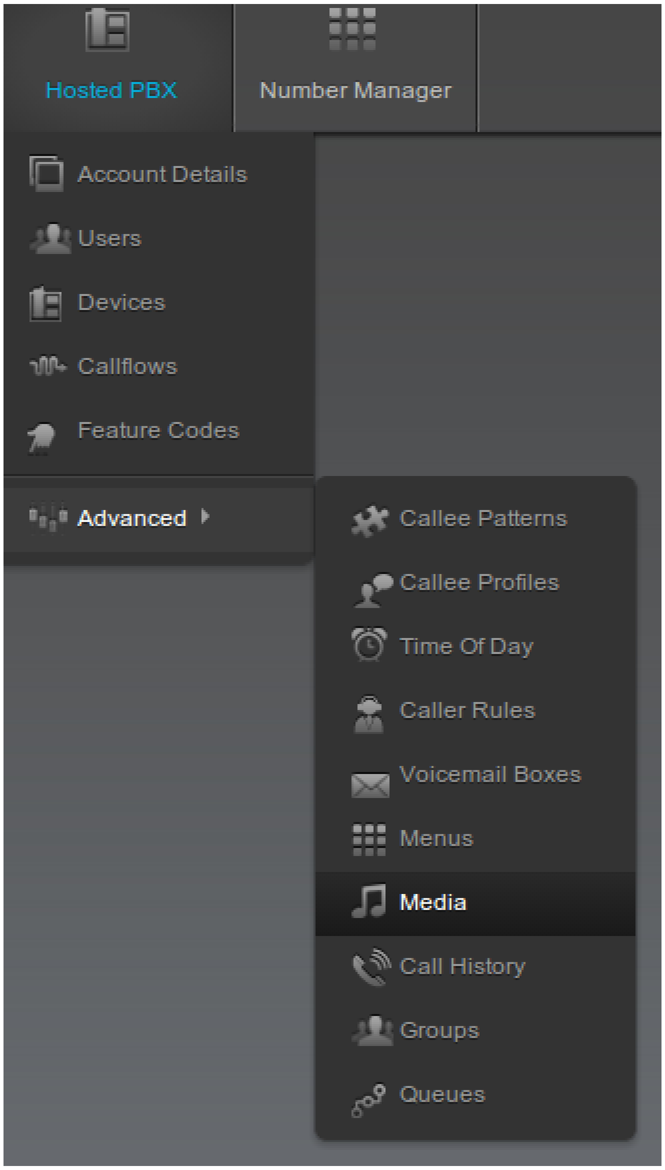


Start Date: 2014-05-09 End Date: 2014-05-10 Filter Total duration: 0 hours 53 minutes and 24 seconds

From (Caller ID)	To (Dialed number)	Owner	Date	Duration	Call Type	CMC Code
105	106	105 itcenter	2014-05-09 8:10:10 PM	00:01:51	internal	1999
105	106	105 itcenter	2014-05-09 8:10:10 PM	00:01:51	internal	1999
105	106	105 itcenter	2014-05-09 8:10:10 PM	00:01:51	internal	1999
105	106	105 itcenter	2014-05-09 8:08:30 PM	00:01:37	internal	-

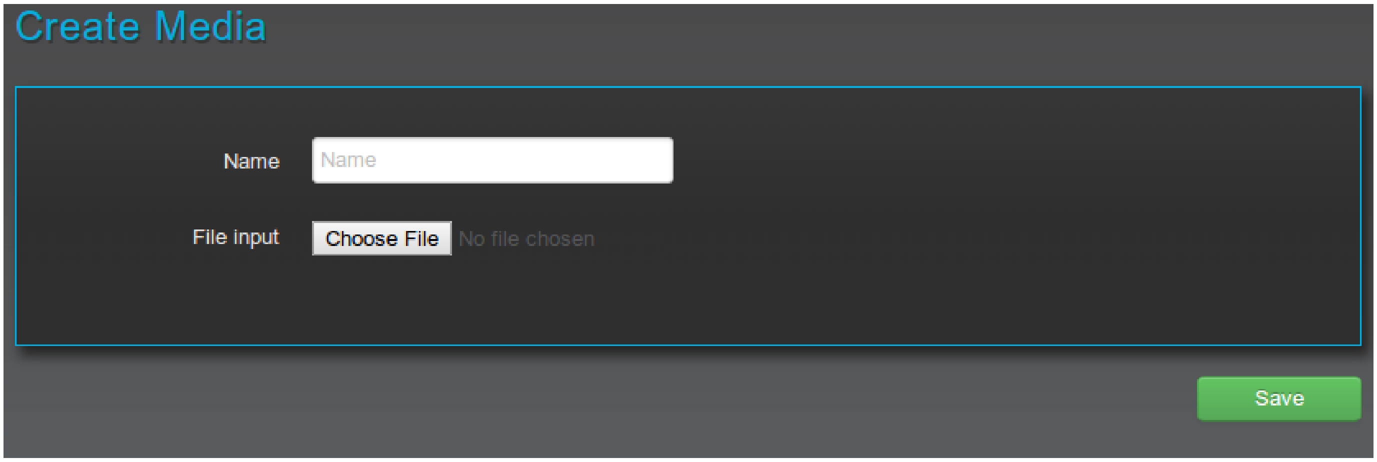
ENG - 7.8 - Music or Prompt Upload

We have to choose the option **Media** in **Advanced**, **Hosted PBX**:



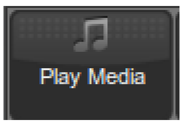
To add a music we choose "**ADD MEDIA**"

Then we choose a name for the song or prompt and select a media that is in our PC. The maximum file size is 1MB.

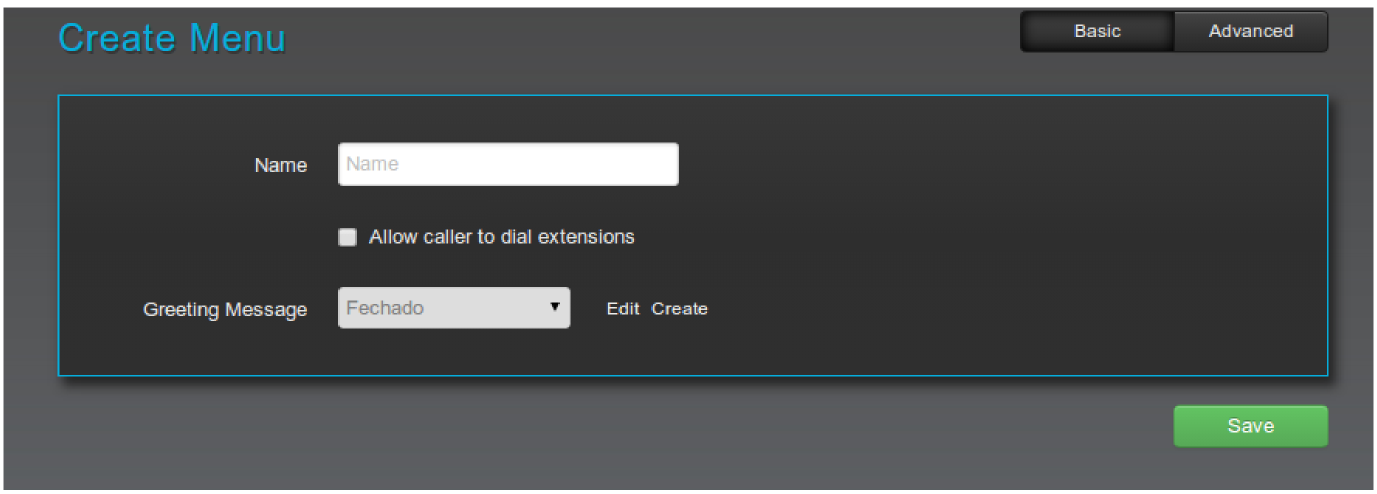


After creating a new media that can be used in a Callflow from a menu or as music on hold (MOH).

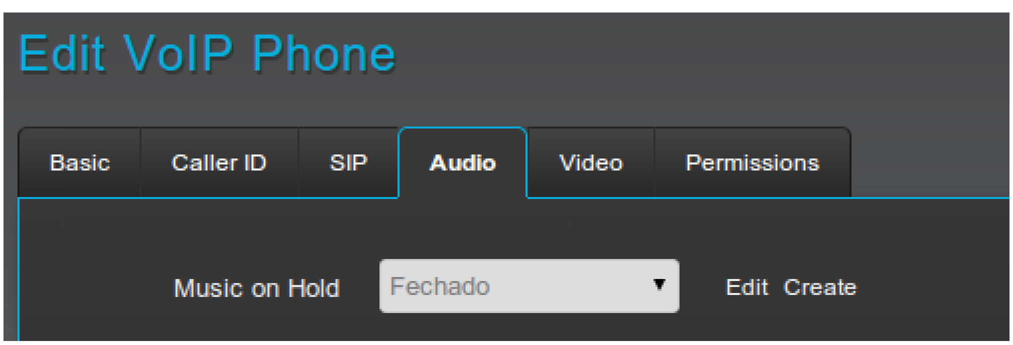
Icon in one Callflow



Menu option



MOH option in phone setup.

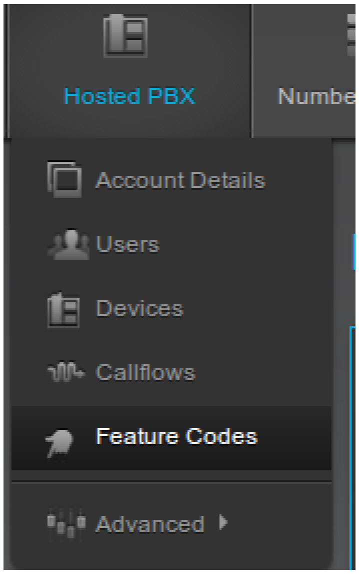


ENG - 7.9 - Night and holiday service

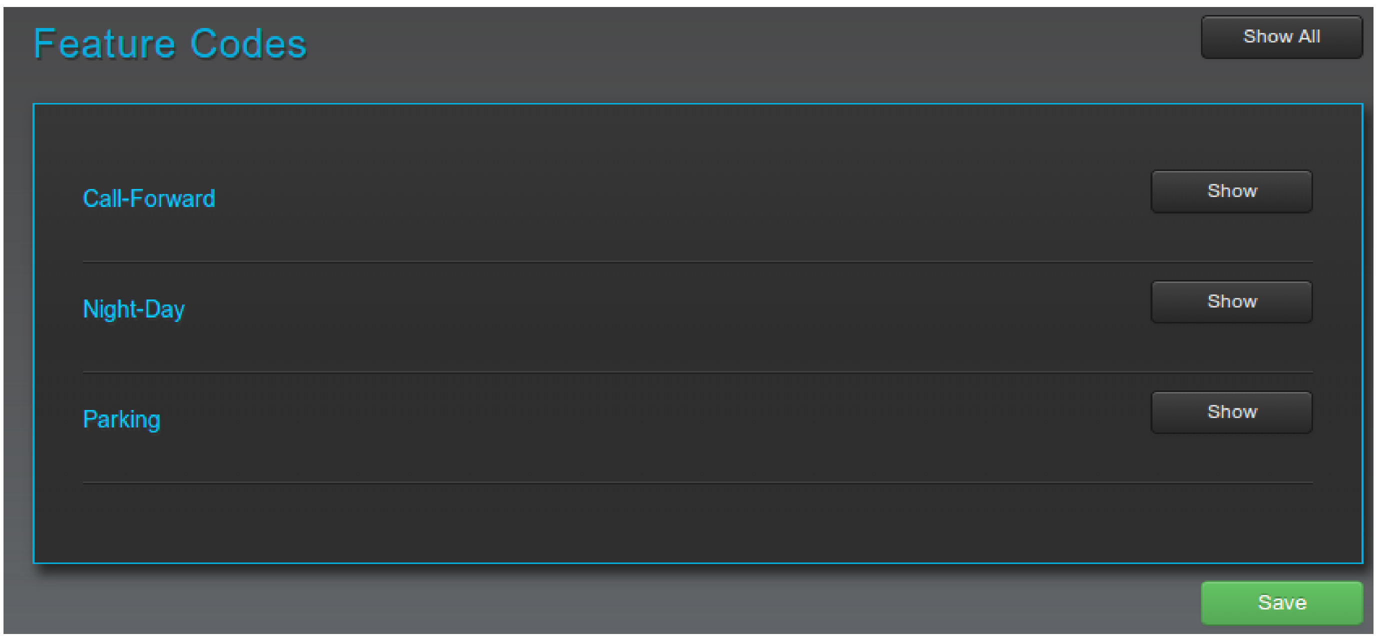
This feature allows you to define a state of present or Away and associate a different action for each state.

This feature is associated to **Callflow**.

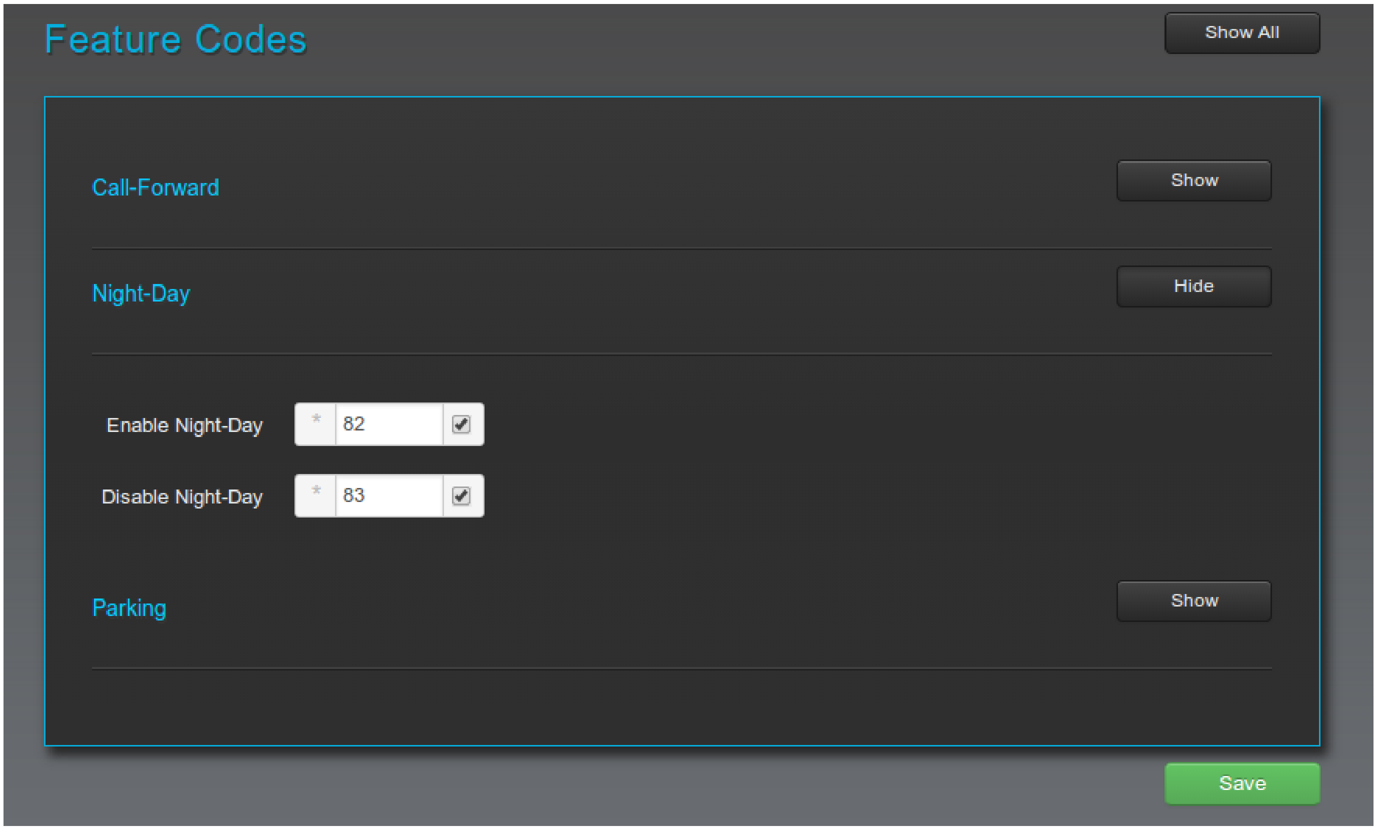
The **Night and Day** function has to be activated in **Hosted PBX** > **Feature Codes** :



Choose **Show** to verify the state of the feature:



Activate the **Night-Day** on the combo box.

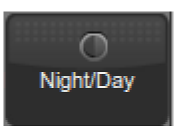


After activating the Night and Day we can use the feature on the Call flow.

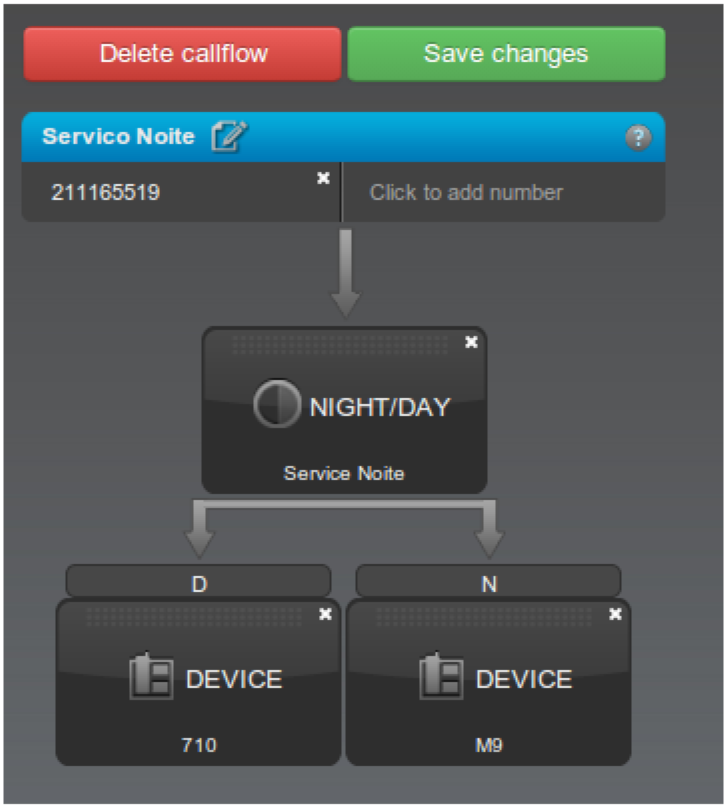
When using this function the user can be in night mode or day mode. To change the state we have to use the features codes.

Feature Code	Action
*82	Day mode
*83	Night mode

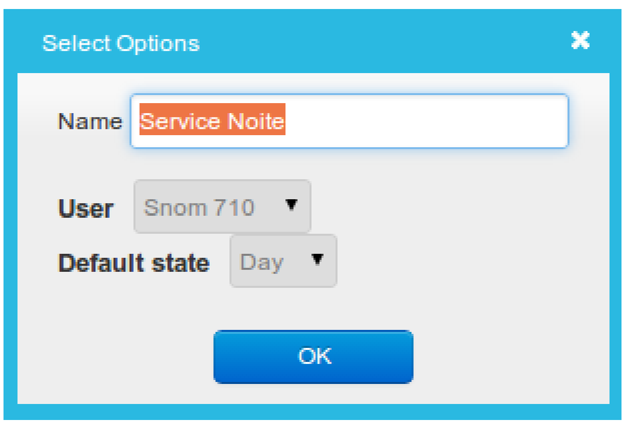
Night and Day Icon



Callflow with **Night and Day**.



Night and Day setup, where we associate the **Night and Day** to the **User**



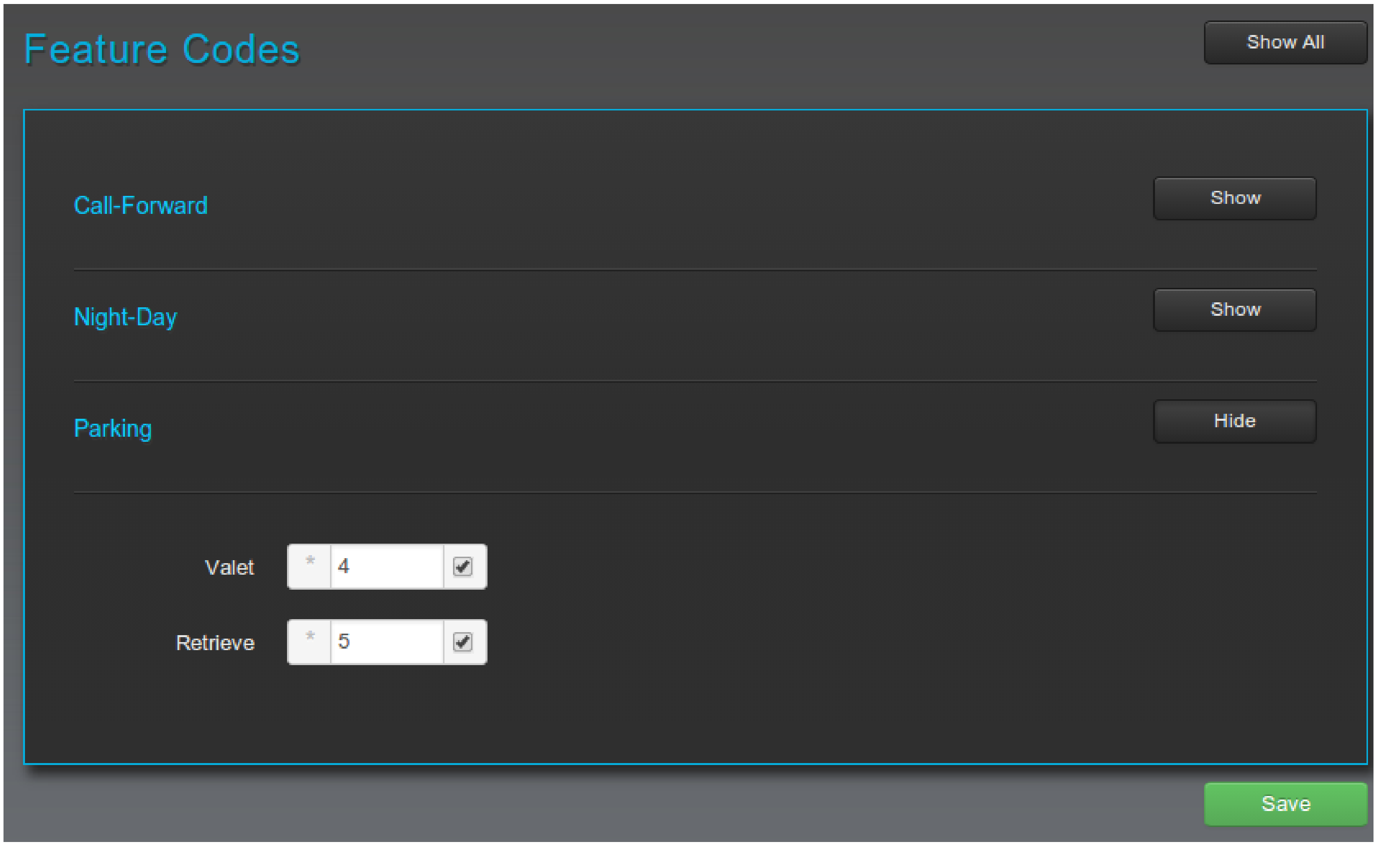
ENG - 7.10 - Call Park

This functionality allows to Park a Call, that means to put it on hold on voiciscore.

We can go back to retrieve the call by using a code associated with the position of the call.

The **Park** function has to be activated in **Hosted PBX** > **Feature Codes** :

Choose **Show** to verify the state of the feature:



Parking values

Name	Value
Valet	*3
Retrieve	*4

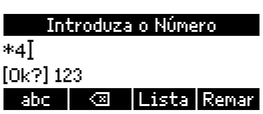
This are the codes associated with the **Call Park** function.

Put the Call in Park

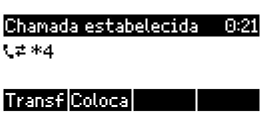
To Park a call we must already be in the conversation.

Choose **Transf** key to transfer the call to the **Park** system.

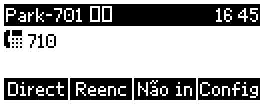
We introduce the number associated to the **Call Park.**

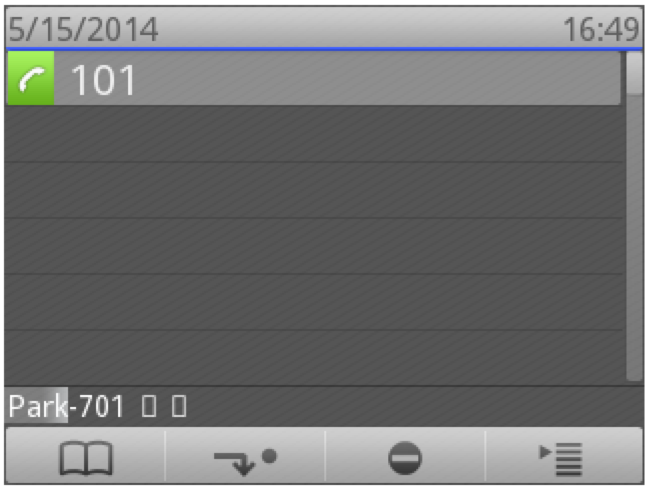


Parked Calls, on the destination if it has an IP phone with display.



Visual information of the **Park** number **(Park-701)**

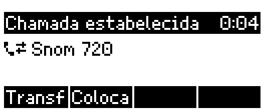




Retrieve Park

Call the number associated with the Park then the number of the **Park** position.

Retrieved Call

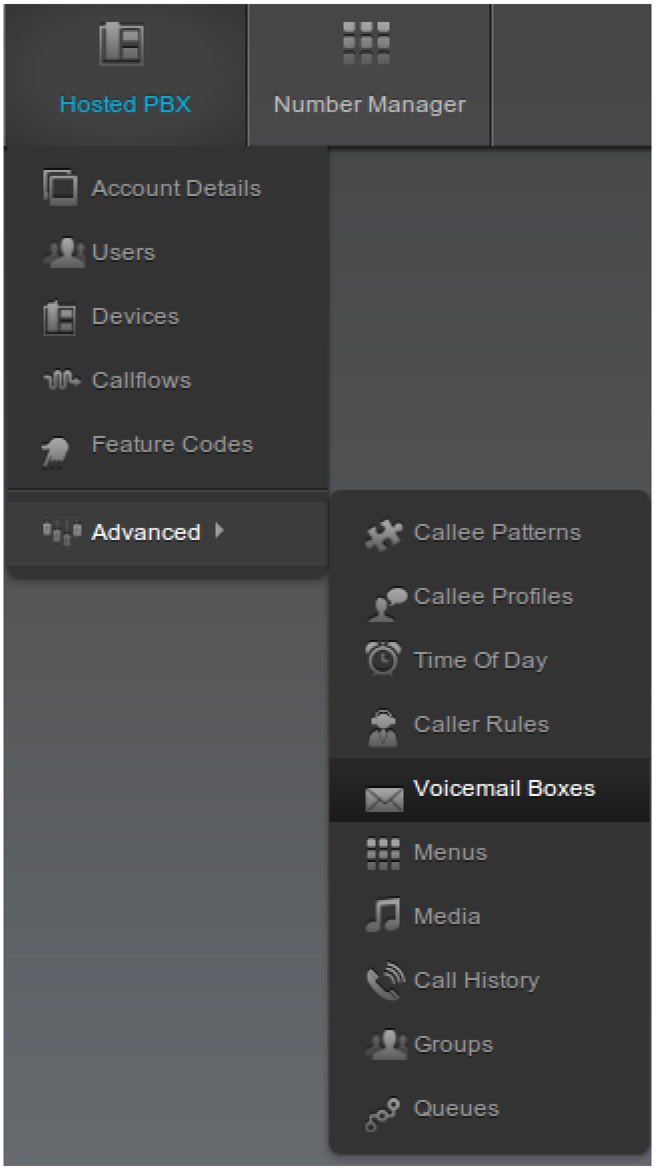


ENG - 7.11 - Voicemail

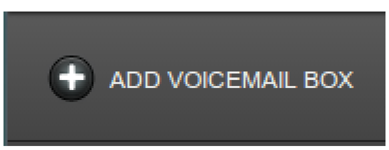
This feature allows the Caller to leave a voice message if the call is not being answered.

To use this feature, we have to create the **Voicemail**. One user can have several Voicemail Boxes.

Hosted PBX > **Advanced** > **VoiceMail Boxes**



To add a new Voicemail Box we have to choose **ADD VOICEMAIL BOX**



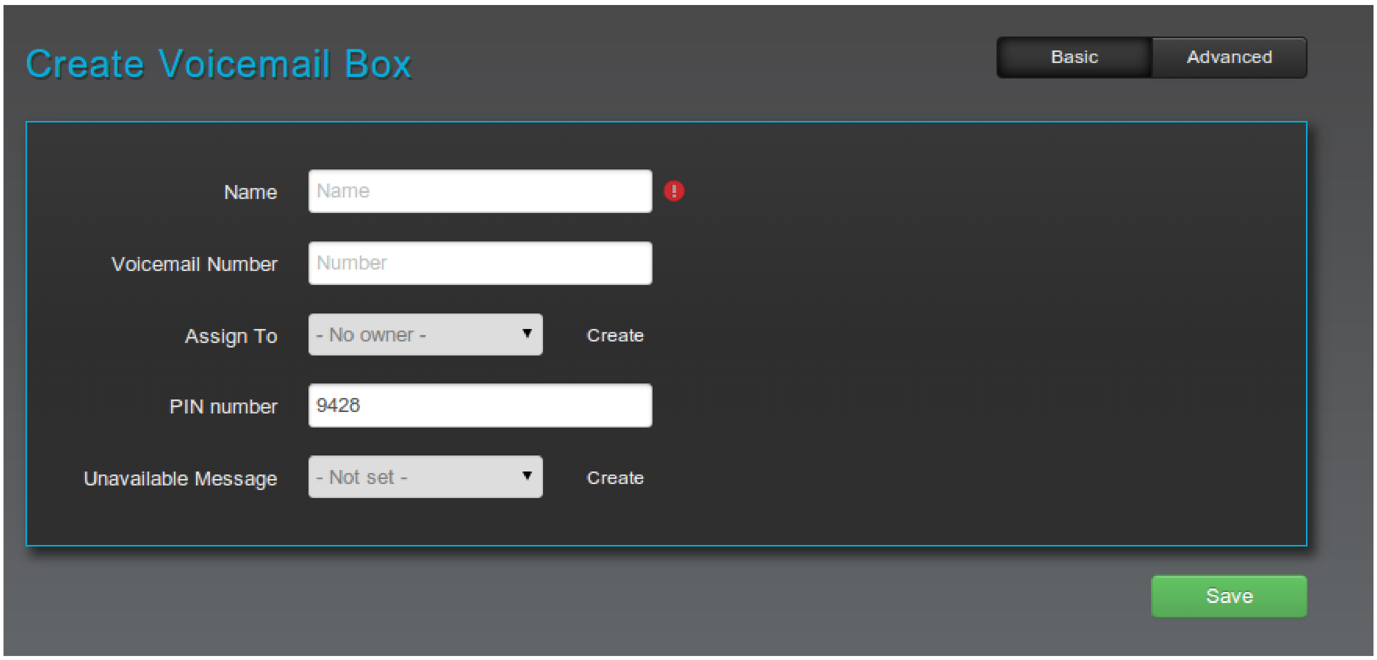
The **Voicemail Boxes** settings are:

Name	Value	Description
Name		Name assigned to the mailbox
Voice Mail Number	Number	Number to be assigned to the mailbox, number used for authentication
Assign to	User	User associated to this mailbox
PIN Number	Number	PIN to access the mailbox
Unavailable Message	Media	Voice message that is played when we are asked to leave a message

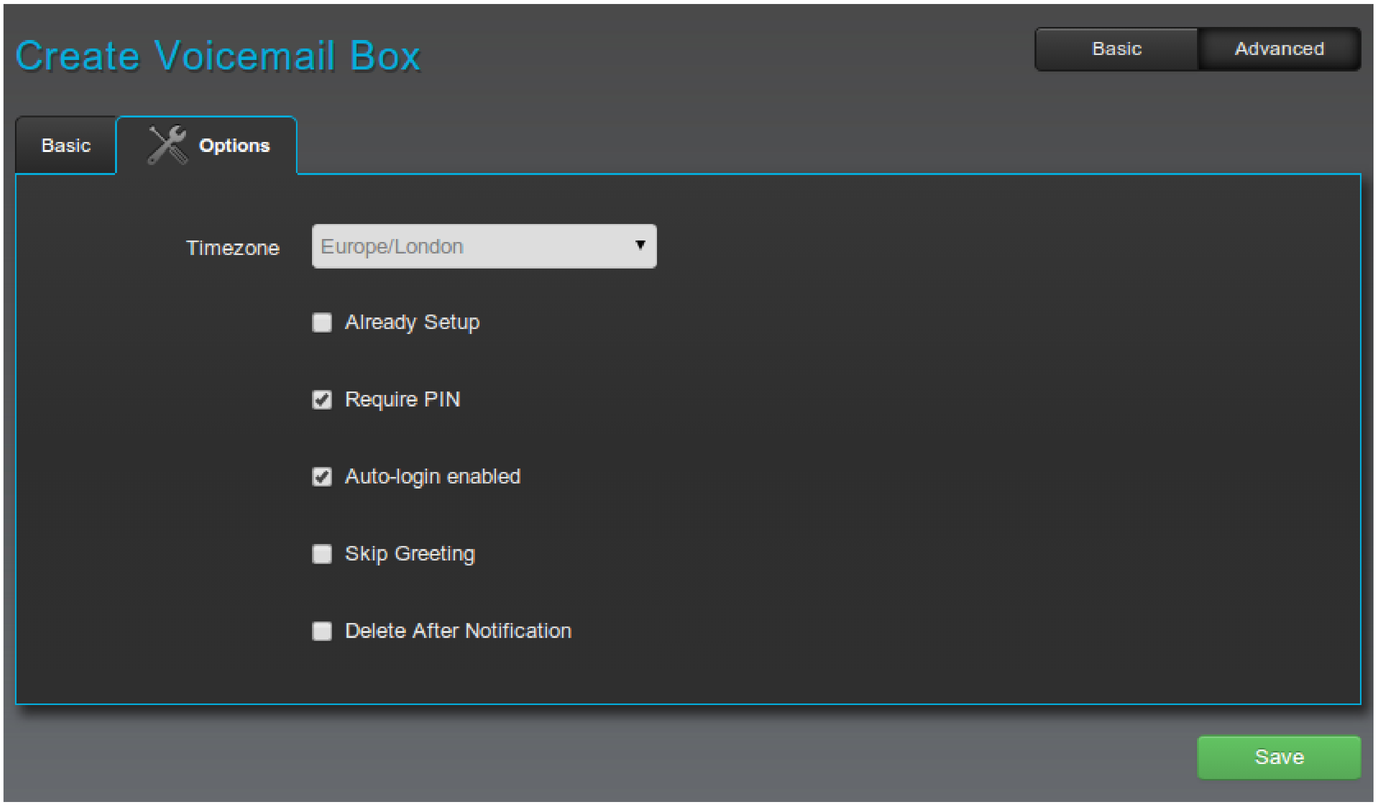
Advanced Settings:

Name	Value	Description
Timezone	Zona	Time the message is displayed
Already Setup	True/False	If the box has been set or not
Require Pin	True/False	PIN use to access the mailbox
Auto-Login Enable	True/False	Authentication use to access the mailbox
Skip Greeting	True/False	If a welcome message is played
Delete After notification	True/False	If you delete MWI after listening messages

Voicemail basic



Advance Options

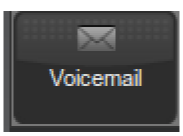


After creating a **Voicemail Boxe** we can use the voicemail function in a **Callflow**.

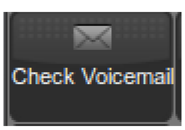
There are two icons for Voicemail, one to send a call to the mailbox, another for the user to retrieve the **Voicemail**.

Icons

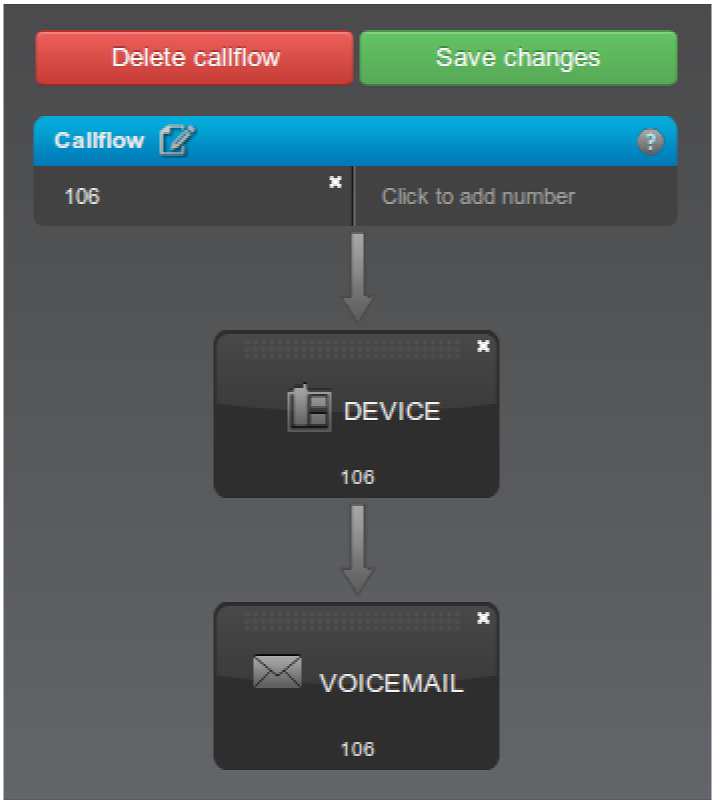
Leave message



Listen Mensagem

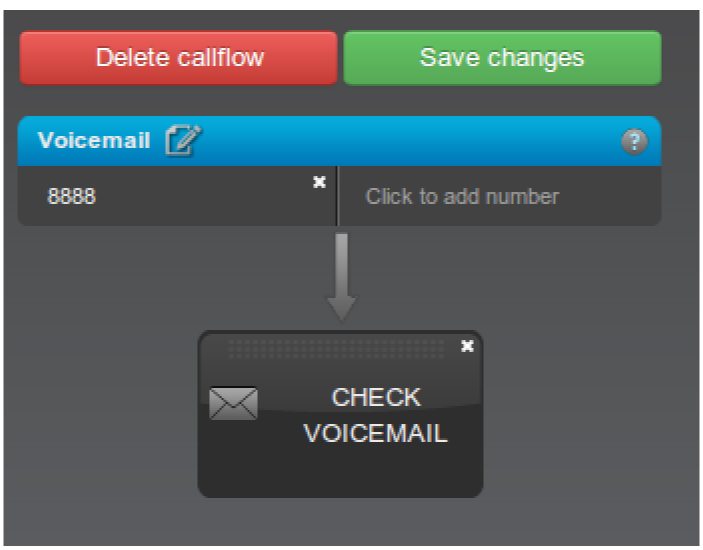


CallFlow to leave message



Callflow to listen message

Usually, we choose the number 8888 to voicemail. It is the number associated with the provisioning.

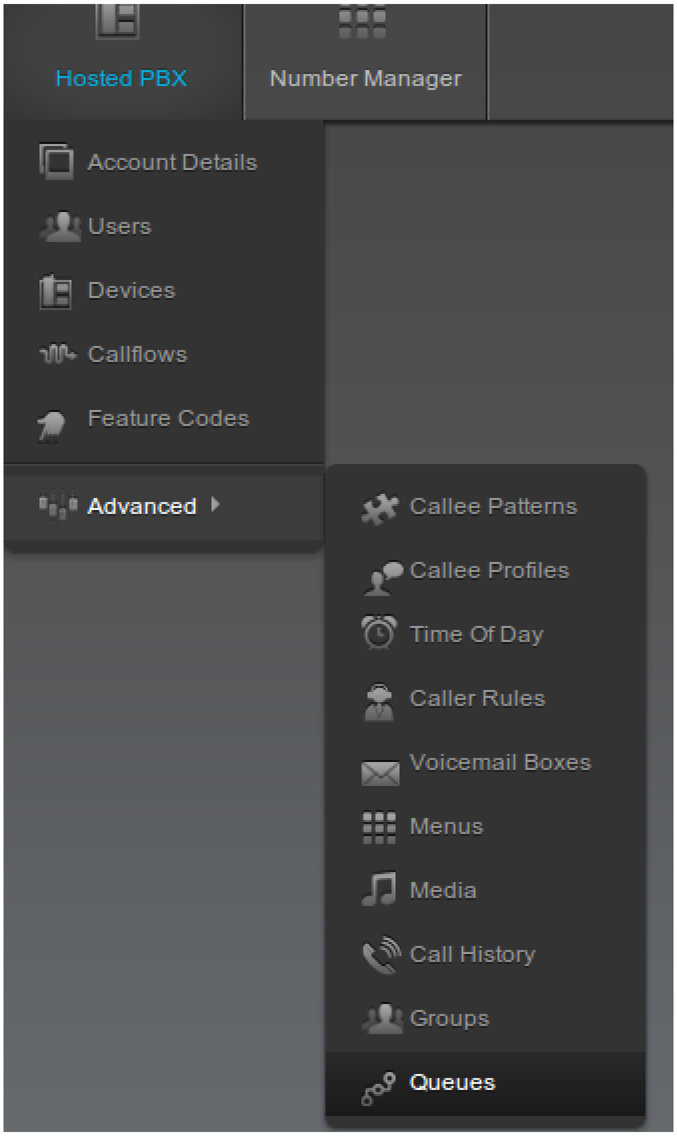


ENG - 7.12 - CallQueues

The **CallQueues** feature allows the Caller to hear music until the Call is answered.

To enable we have set up a **Callqueue** group.

Menu **Hosted PBX** > **Advanced** > **Queues**



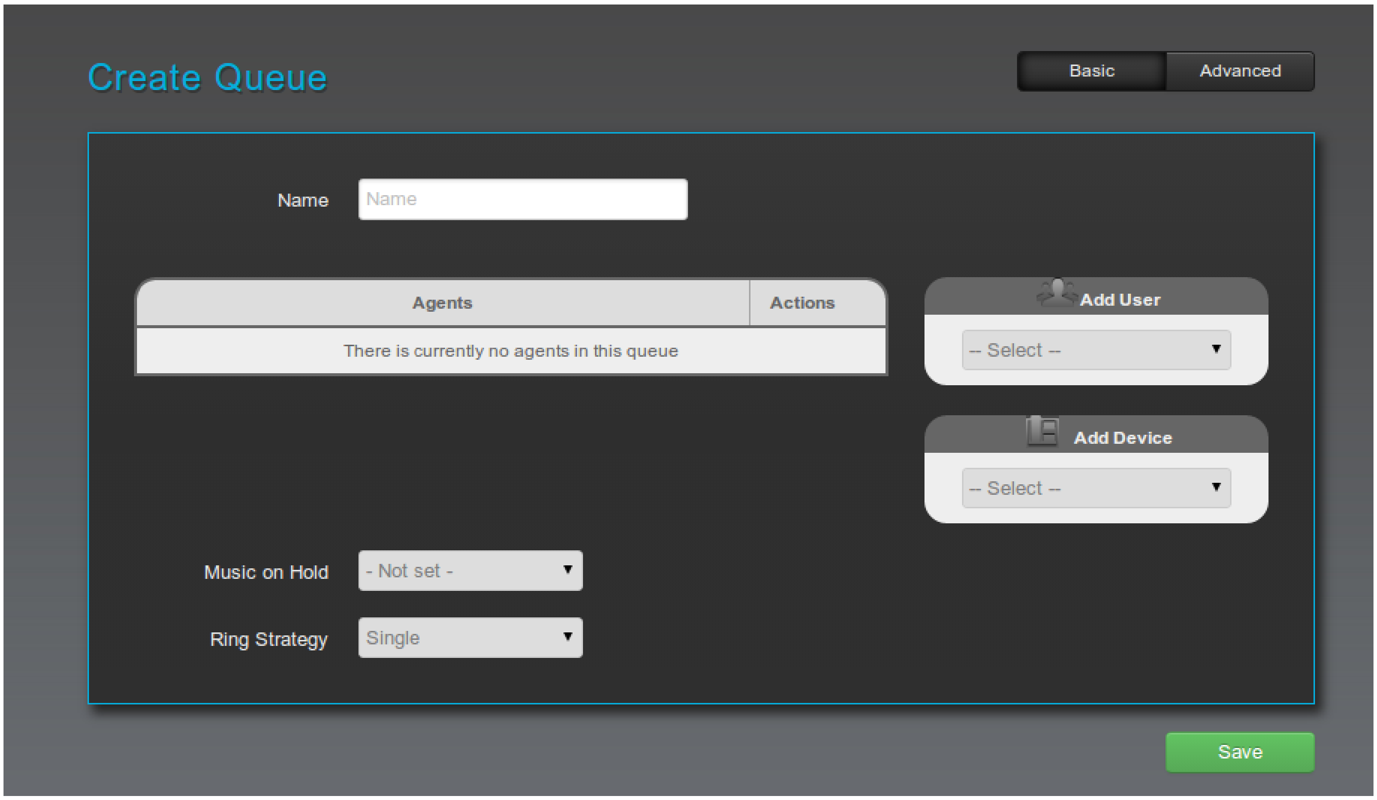
Create the **Queue** choosing

Then we have to configure the following elements in the **Queue.**

Name	Value
Name	Name of the Queue
Agent	Phone or user that is going to take part in the Queue
Music on hold	Media source, for when the caller is on hold
Ring Strategy	Single (Play only on one phone) Ring All (Play in all phones)
Connection Timeout	Time (in seconds) that a caller is in the queue before exiting the same and then the next Callflow
Agent ring timeout	How long (in seconds) does the phone of the agent rings

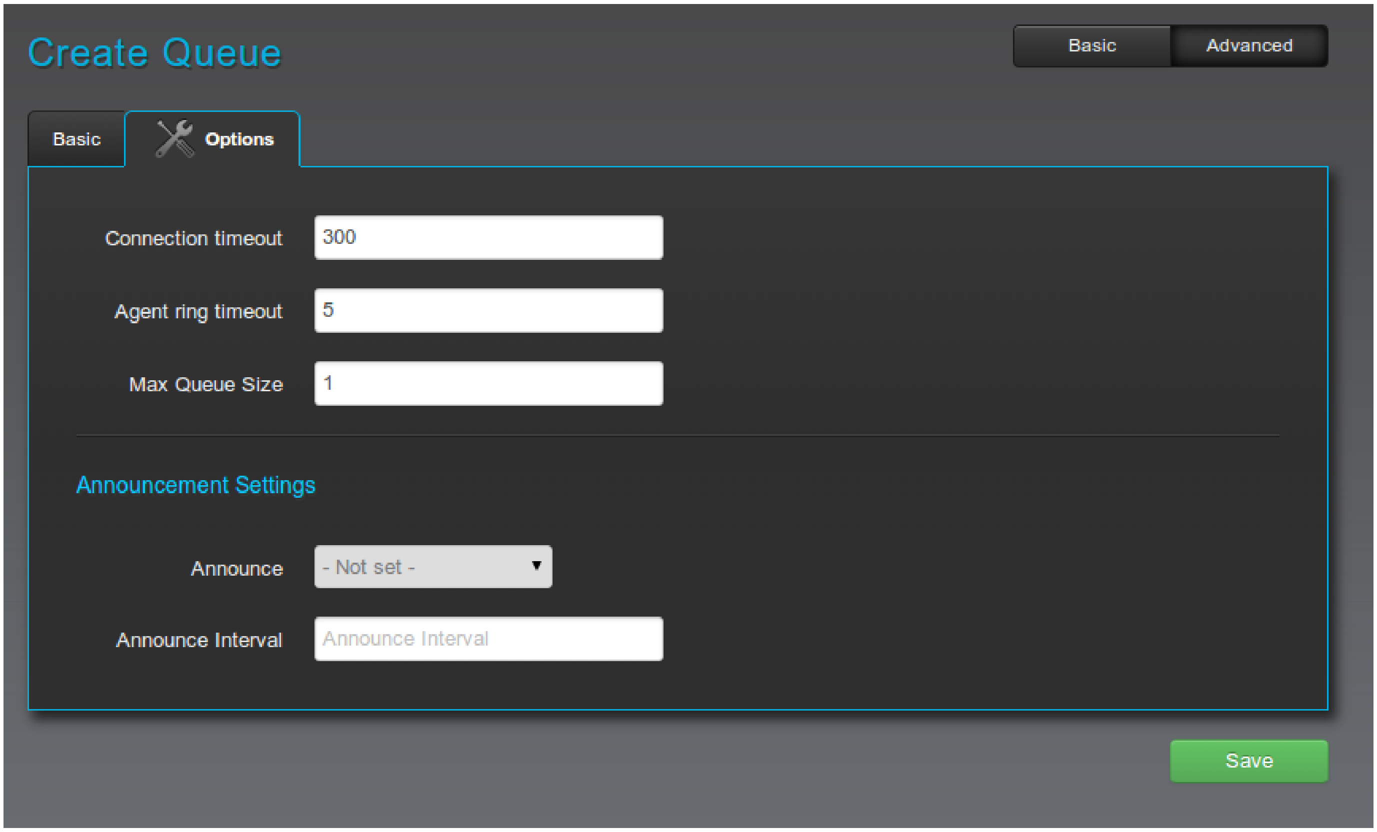
Max Queue Size	Maximum of people in the queue, above this value the call goes to the next action in the queue
Announce	Announce while we are in the queue (may be periodic see below)
Announce Interval	Interval in seconds that the announcement is played

Menu Create Queue



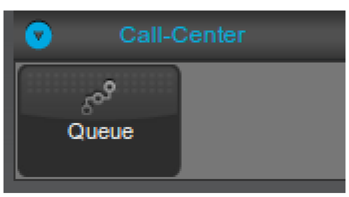
Tab Queue Advanced

Menu Queue Advanced Options.



After creating the **Queue** we can associate it to the **callflow**.

Icon Queue



CallFlow with associated Queue, when the call leaves the **Queue** or because it was not answered by the agent continues on the callflow.

In the example below we put an announcement indicating a message.

ENG - 7.13 - eFax integration

ENG - 7.14 - Call Forwarding from selected callers

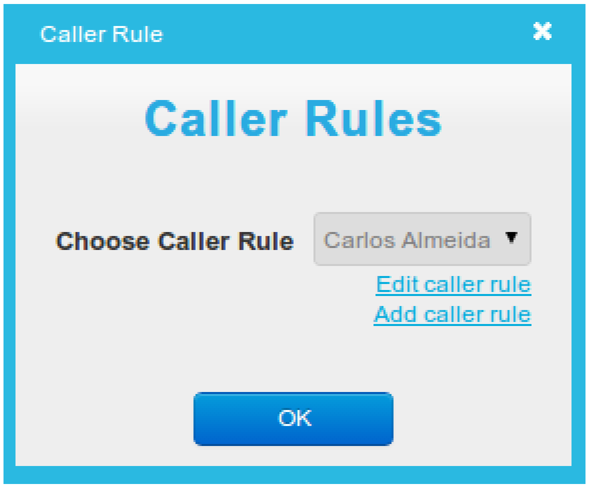
ENG - 7.15 - Caller Rules

This feature allows different response depending on the caller.

To make one router based on the number of the caller we have to use the **Caller Ruler** on the Callflow.

Icon **Caller Rules**

When added a destination we can choose a rule of origin.

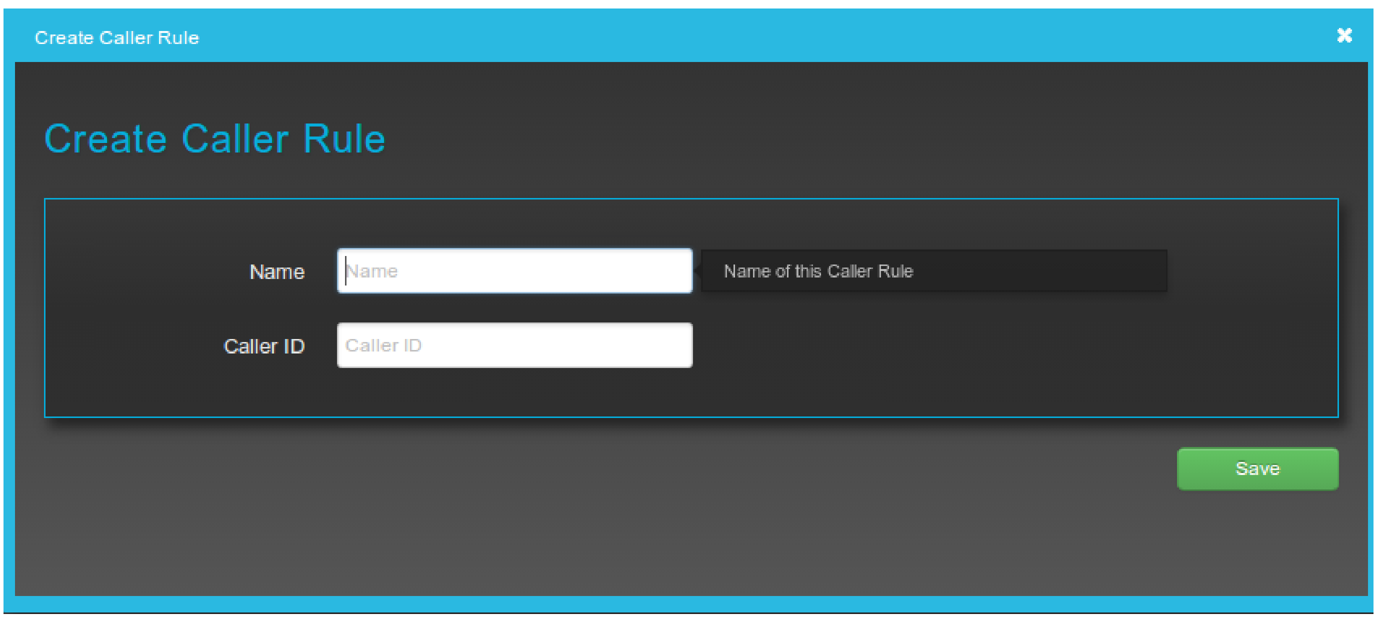


We can use an existing source or create a new one.

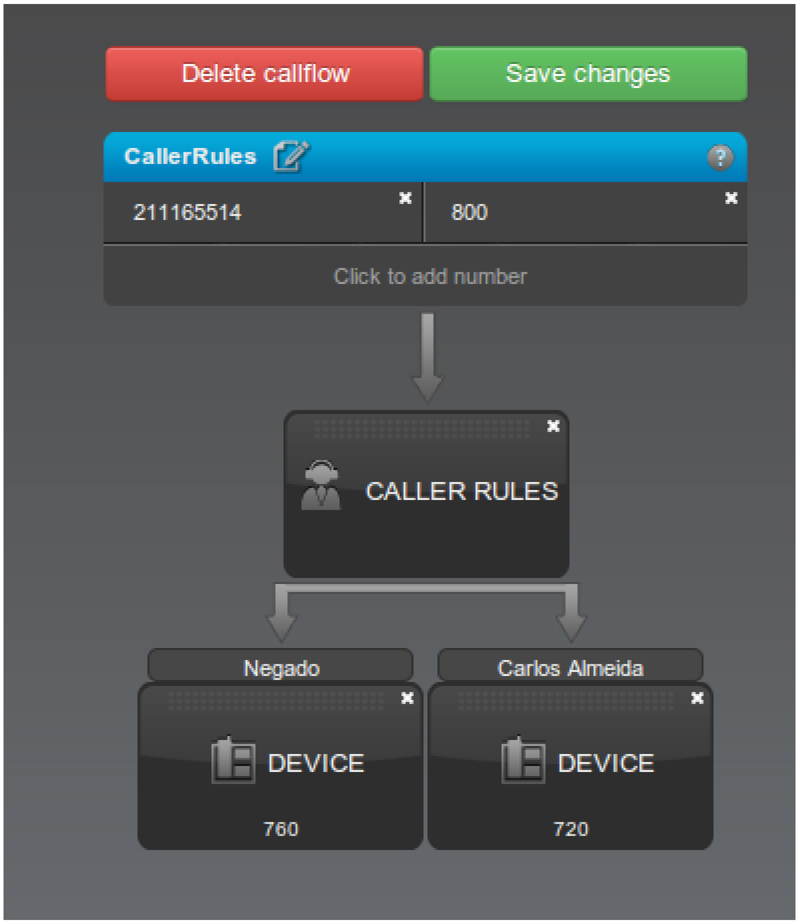
Create a new (**Add caller rule**) or edit (**Edit caller rule**)

Caller Rule Definition

Name	Value
Name	Name of the Caller Rule
Caller ID	Regular Expression based on regex (1)



After setting all the destinations we get a **Callflow** looking like this.



The **Caller Rule** may be placed anywhere in a Callflow.

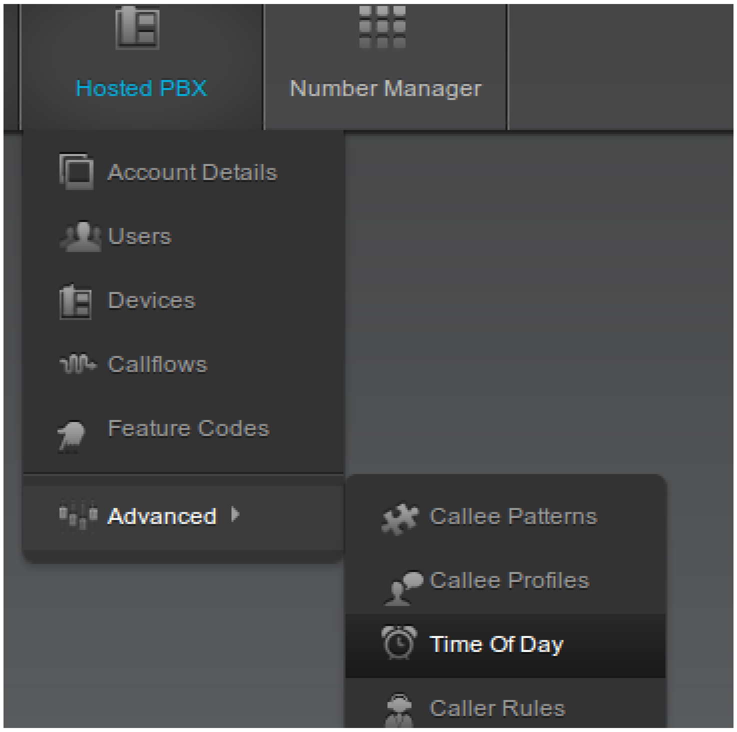
(1) http://pt.wikipedia.org/wiki/Express%C3%A3o_regular

ENG - 7.15 - Time Conditions

To create CallFlow with Time Conditions, we have to create a temporal rule

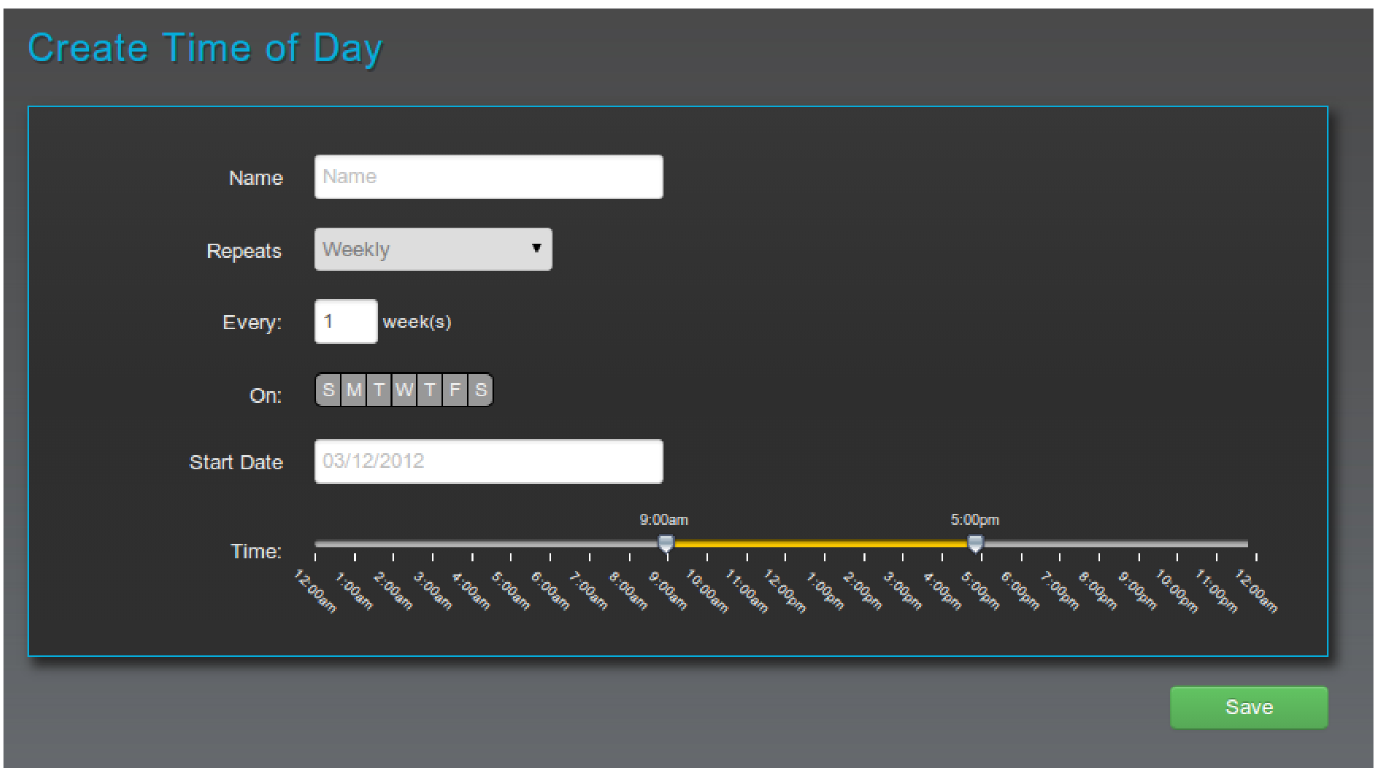
We choose the option **Time of Day**

in Hosted PBX > Advanced > Time of Day



To add a Time of the day, choose "**ADD TIME OF THE DAY**"

We have to name the time of the day of the configuration option "**Time of the Day**"



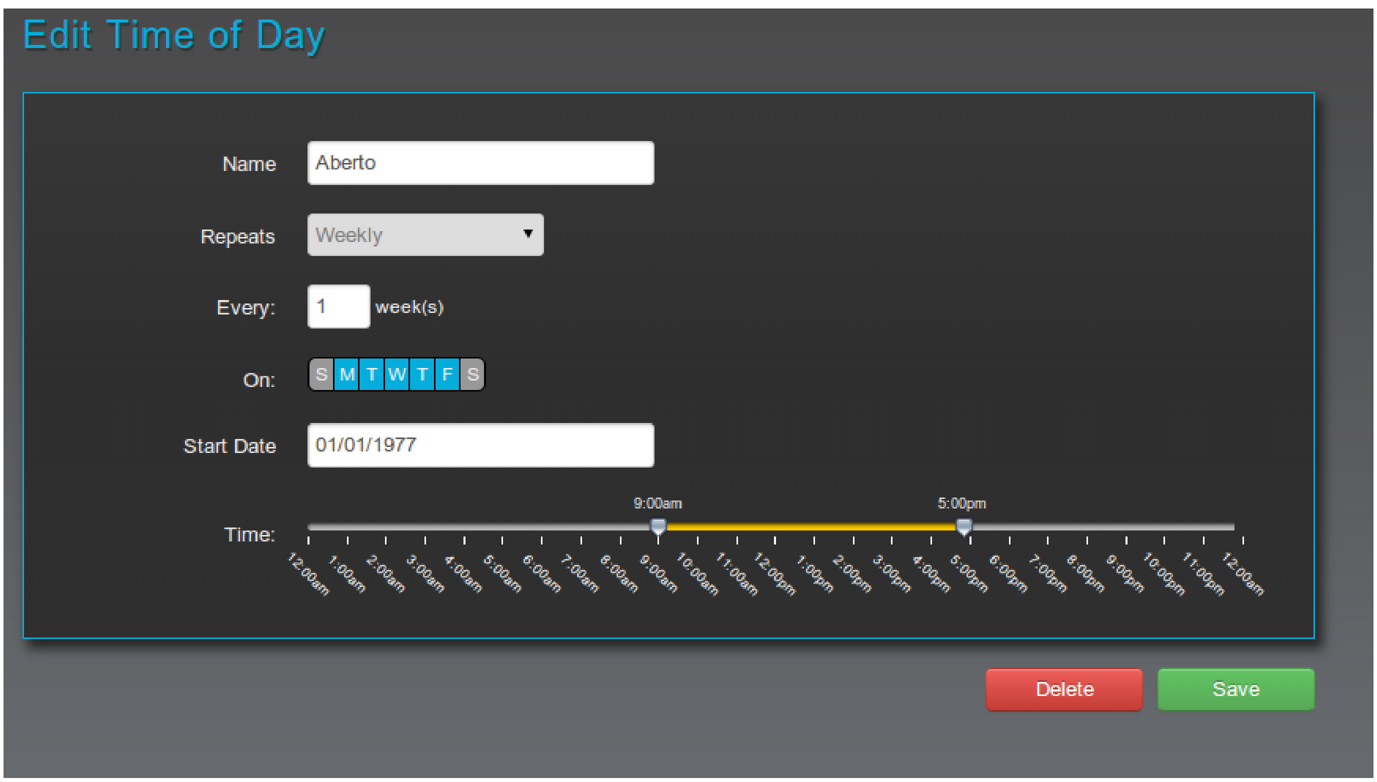
The configuration options are:

Option	Value	Description
Name	Name	Name
Repeats	weekly day	When the pattern is repeated, if all week, every month or year.
Every	1-4 weeks	Allows to choose the week of the month
On	SMTWTFS	Are the days of the week.
Start Date	dd/mm/aaaa	The day when Time of the day becames active
Time	start: end	Interval when the Time of the day becomes active (start time, end time).

Once created the Time of the day that can be used in a Callflow

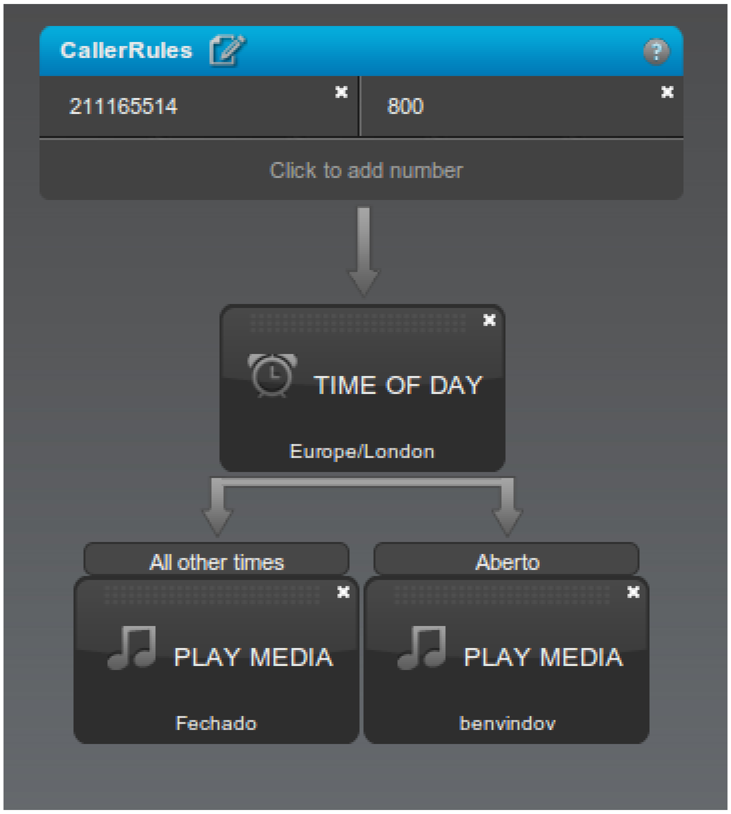
Example:

Time of the day



In this "Time of the Day" we indicate that the active period is all the working days from 9:00 to 17:00.

Callflow



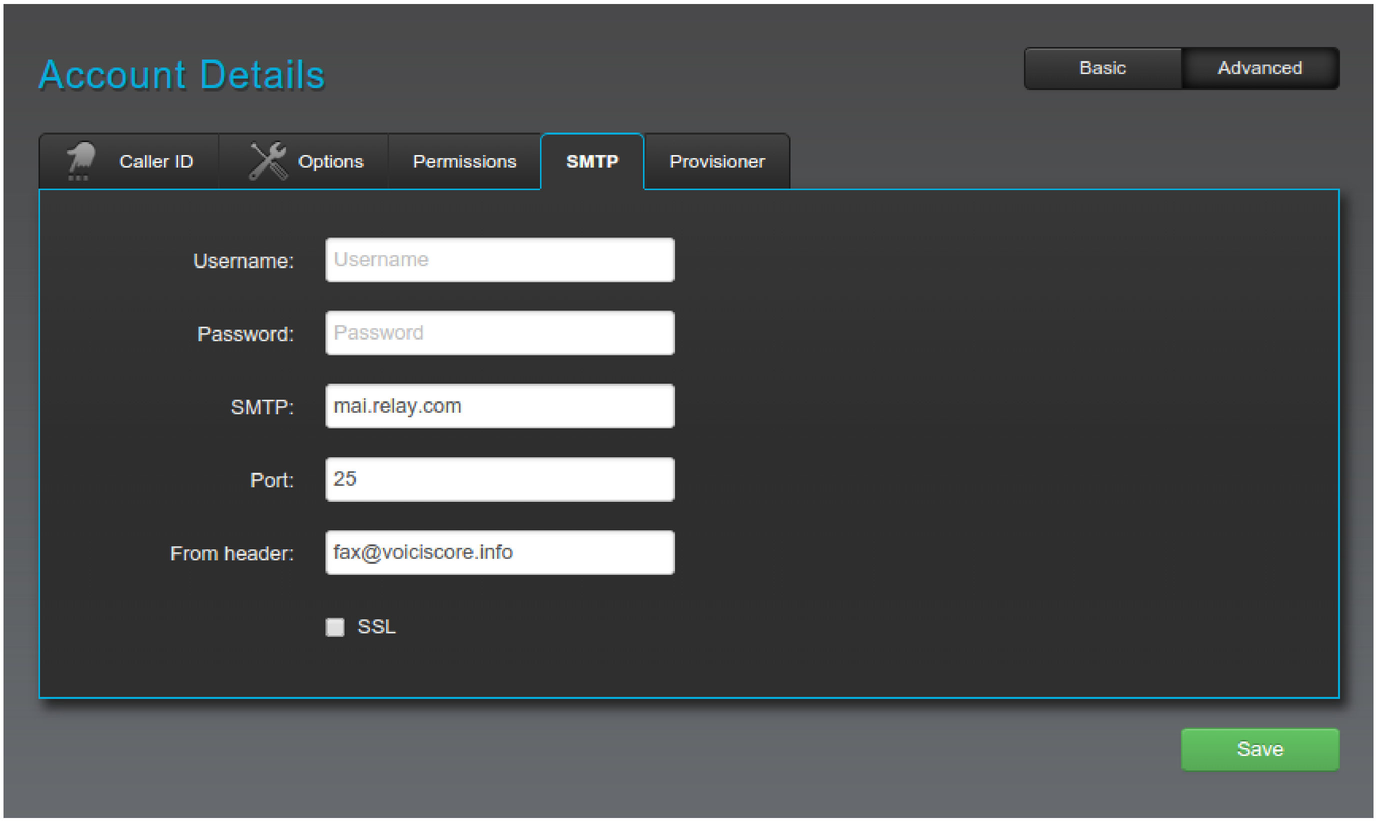
In this case the schedule associated with the "**Time of the Day**" open plays a welcome message, in every other case it plays the message closed.

ENG - 7.16 - Fax to Email

This function allows you to receive a fax on a mail e-mail box.

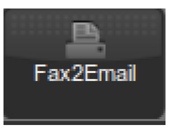
MTA definition

First you need to define the MTA server in account details, in **Advanced TAB**.



Callflow

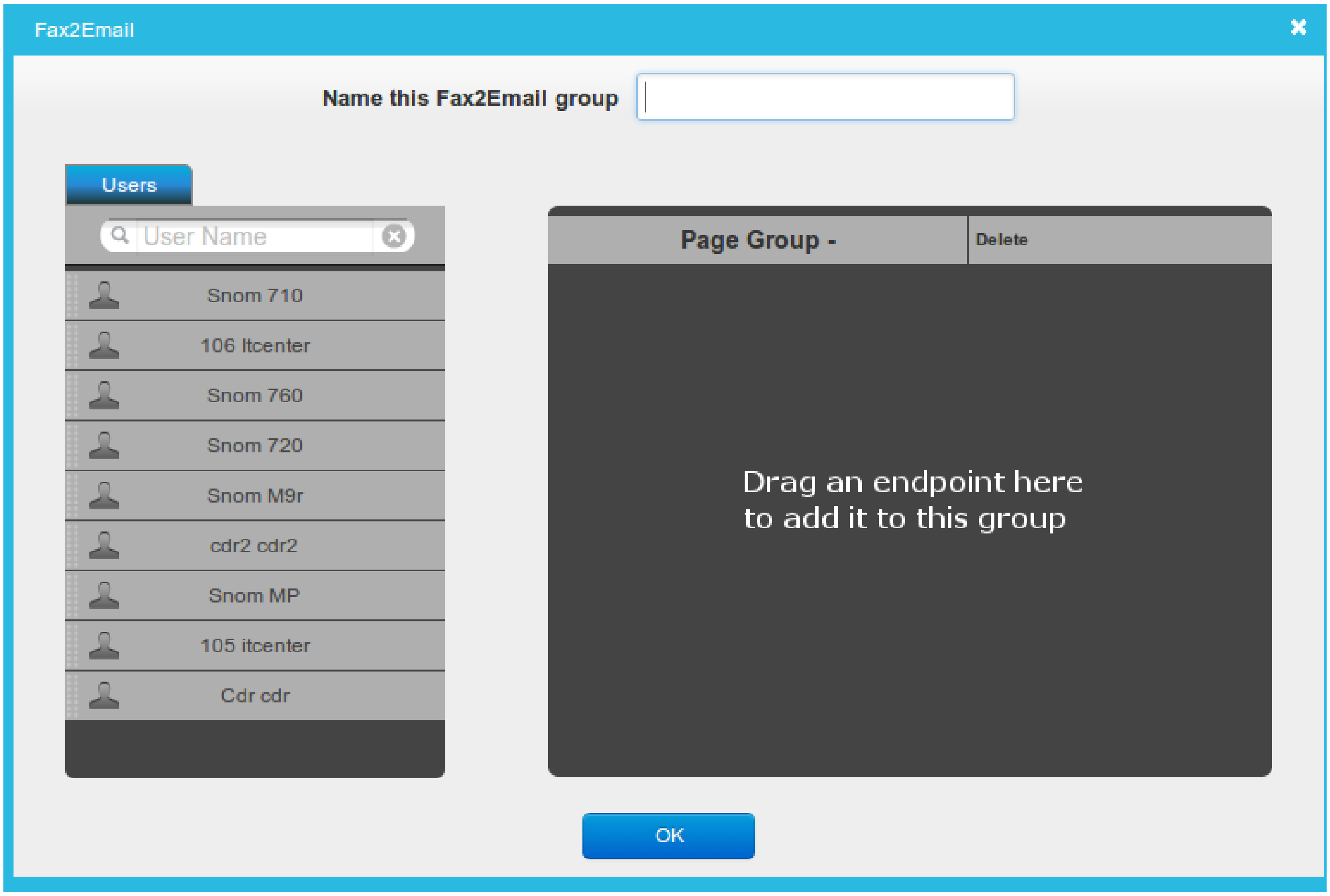
Icon Fax to e-mail



Definition of Fax2email

Name	Description
Name this Fax2Email group	Name of the fax group
Page Group	Users associated to fax

The fax is sent to the user's mailbox.

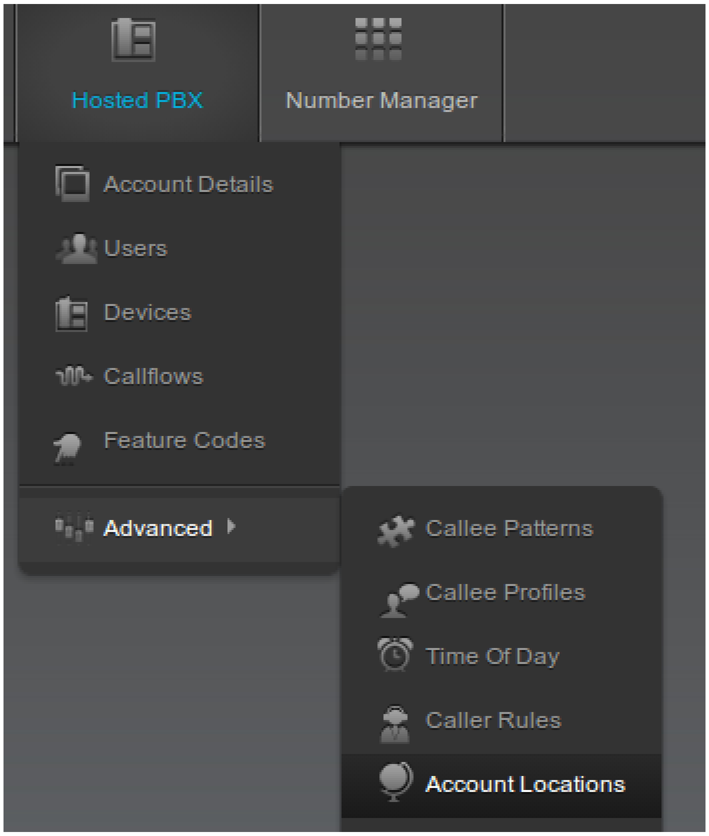


CallFlow with **Fax to Email**

ENG - 7.17 - Max Call per Site

With this feature we can set the maximum number of simultaneos calls that a local account can do to and from the outside.

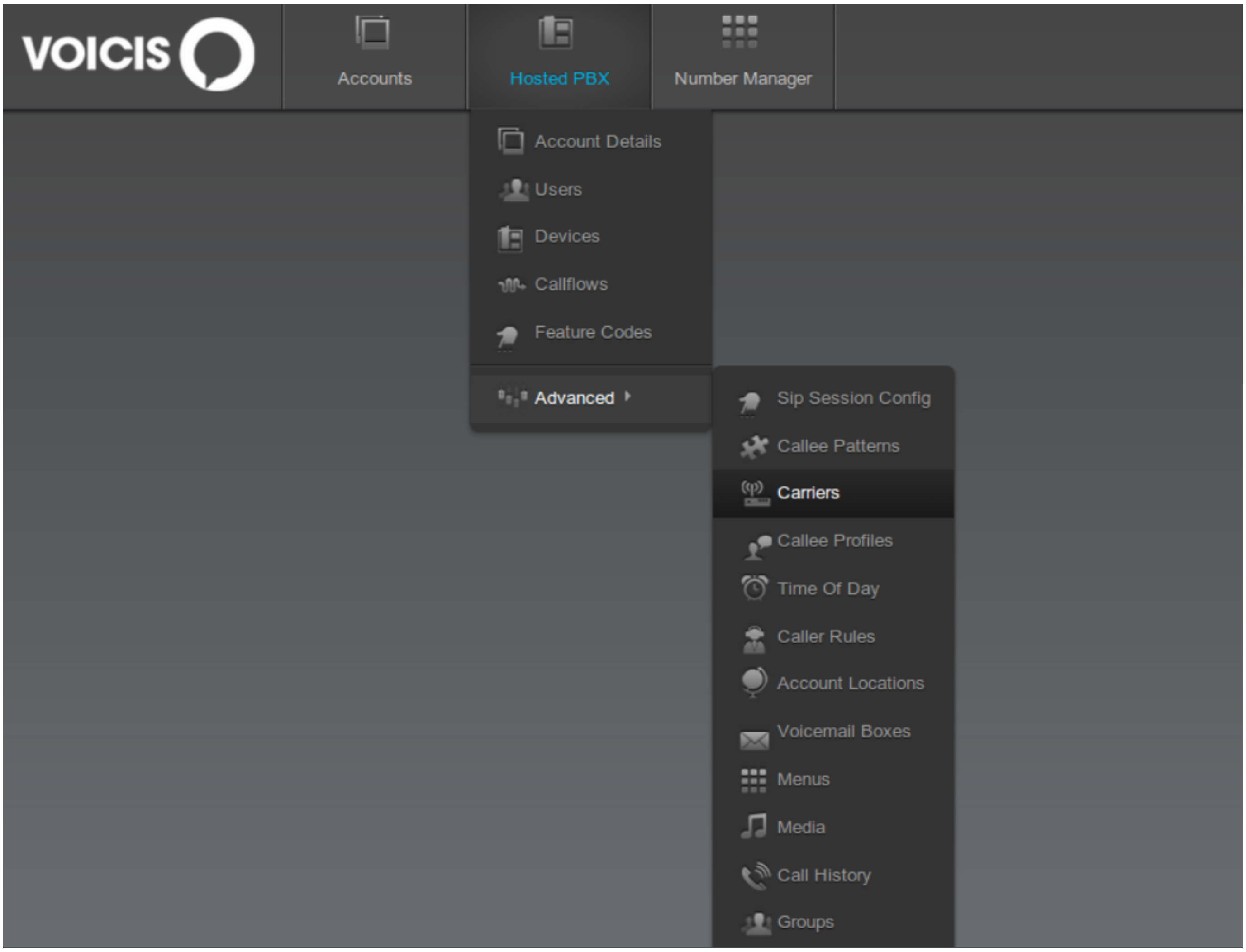
Select **Hosted PBX** > **Advanced** > **Account Locations**



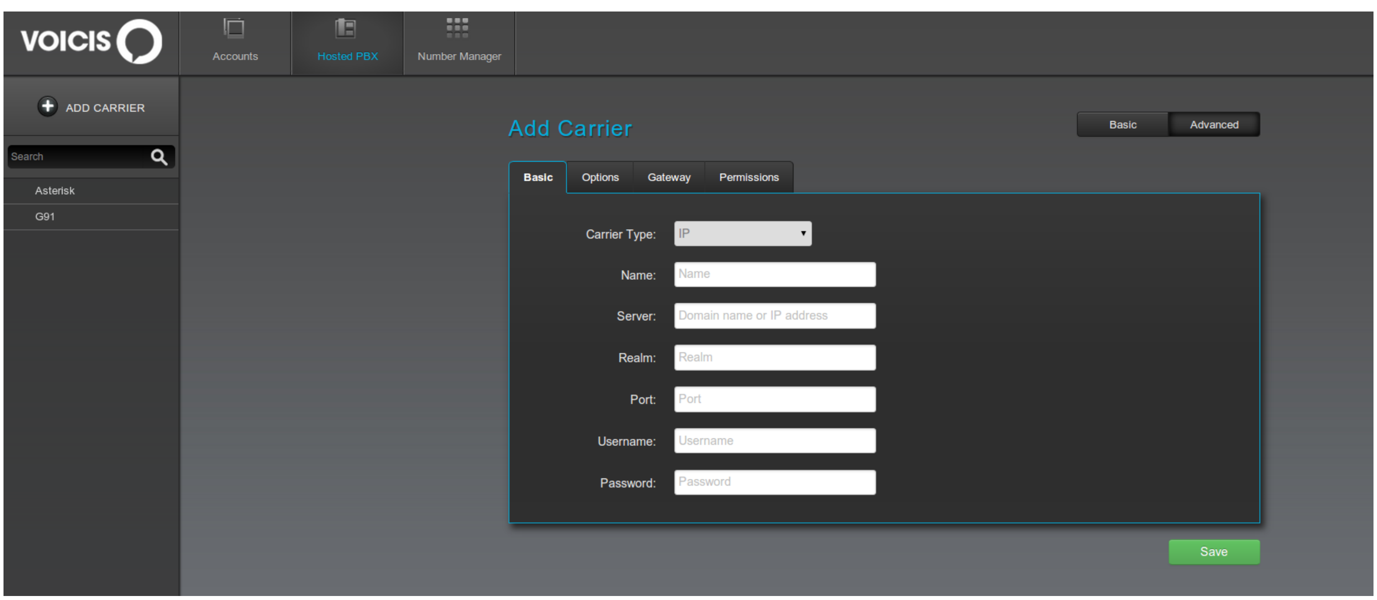
ENG - 7.18 - Carrier

Carrier Configuration

In order to configure a carrier, fist you need access to Hosted PBX menu then Advanced and select Carriers option, as you can see on image bellow:



Add a new carrier selecting the option on left side called "ADD CARRIER":

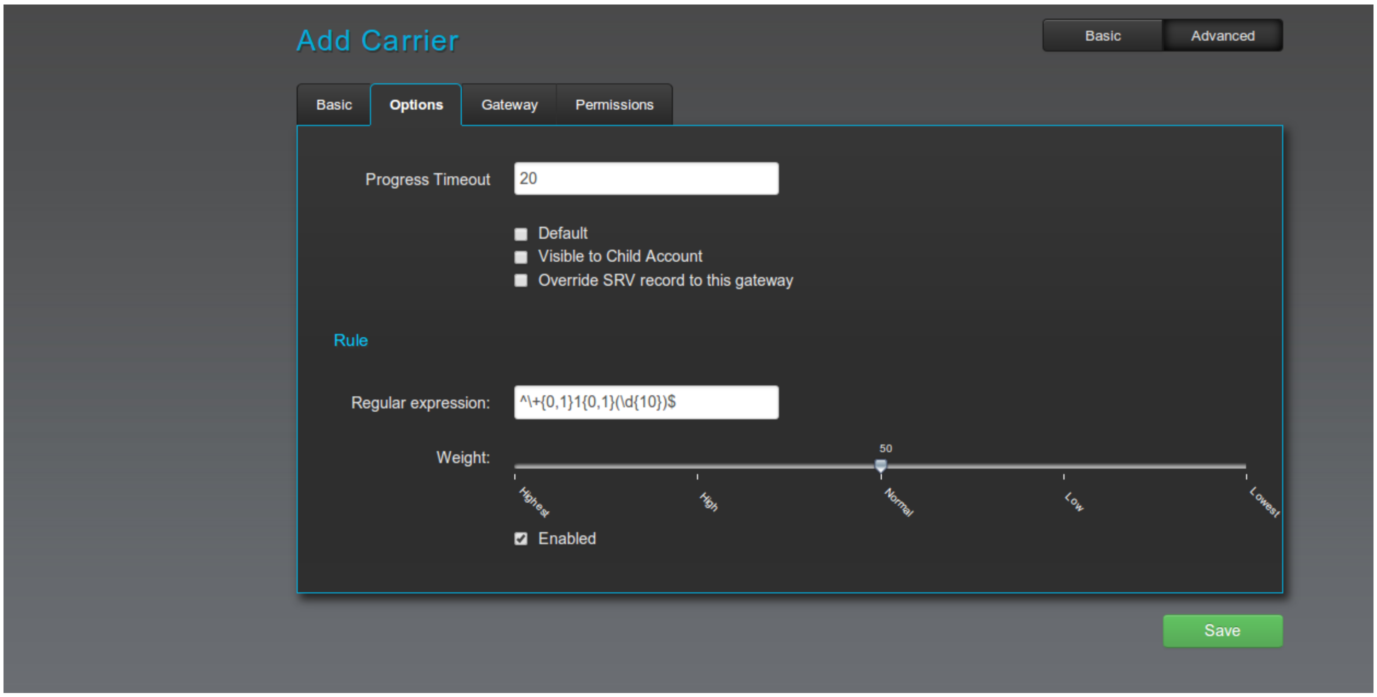


There is 3 types of carriers supported by Voicis:

- IP -> this type of carrier will be controled only by IPADDR,
- INBOUND_REGISTER -> this type of carrier needs register against VOICIS with username and password, in order VOICIS save AOR of the carrier, to knows its location, all invites sent from this type of carrier will be challenged for authentication.
- OUTBOUND_AUTH - > this type of carrier hasn't REGISTER method, but the remote server will challenge VOICIS on INVITES sent from VOICIS for authentication.

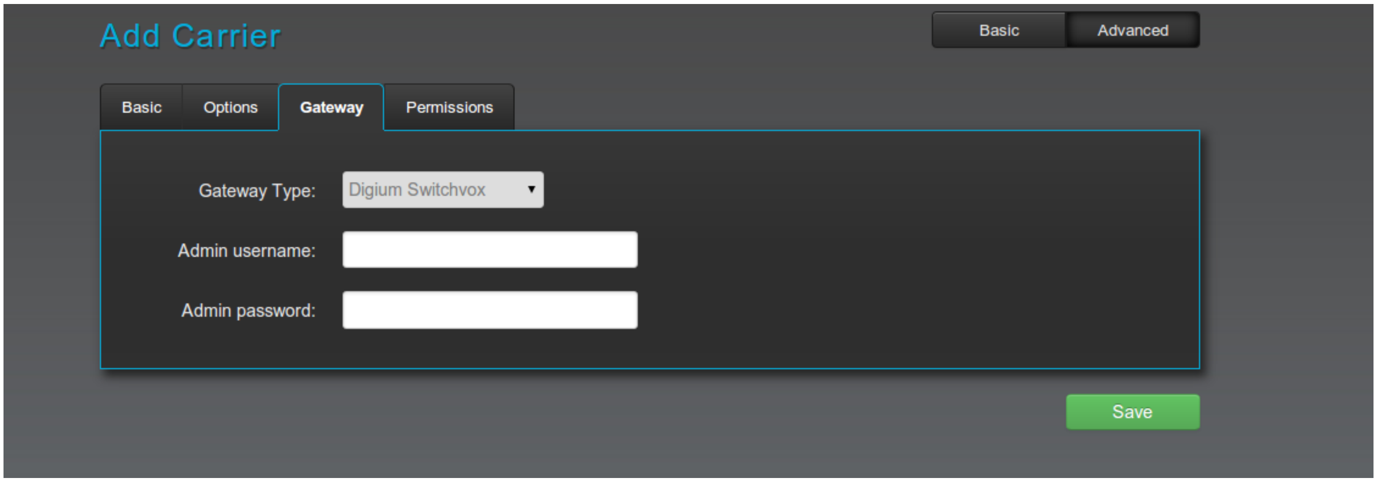
carrier Options tab:

- Progress timeout -> this option is for configure the seconds that carrier will take to reply to Voicis when Voicis send an INVITE to there.
- Default - > This option is to create a default carrier within account, note: you can only have one default carrier per account
- Visible to child account -> this option is to allow child accounts use this carrier in their callflows
- Override SRV Record to this gateway - > this option is for configure dns srv in order to phones register against this gateway instead of voicis.
- Regular expression and weight -> this 2 options are needed to configure LCR. Note: Only Gateways with type IP and OUTBOUND_AUTH will be used on LCR List

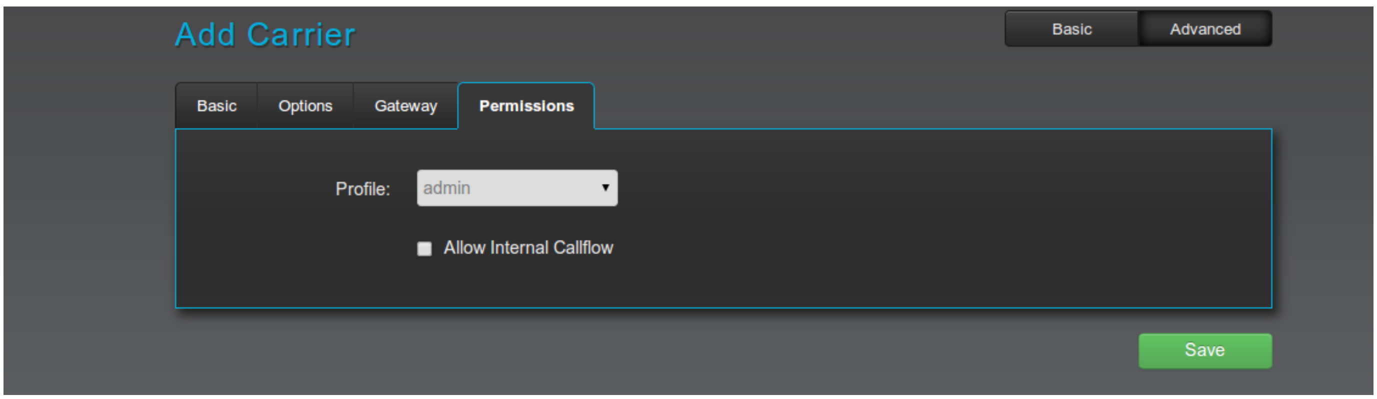


carrier Gateway tab, this is to use carrier API in order to configure this carrier remotely, like devices, permissions, etc...:

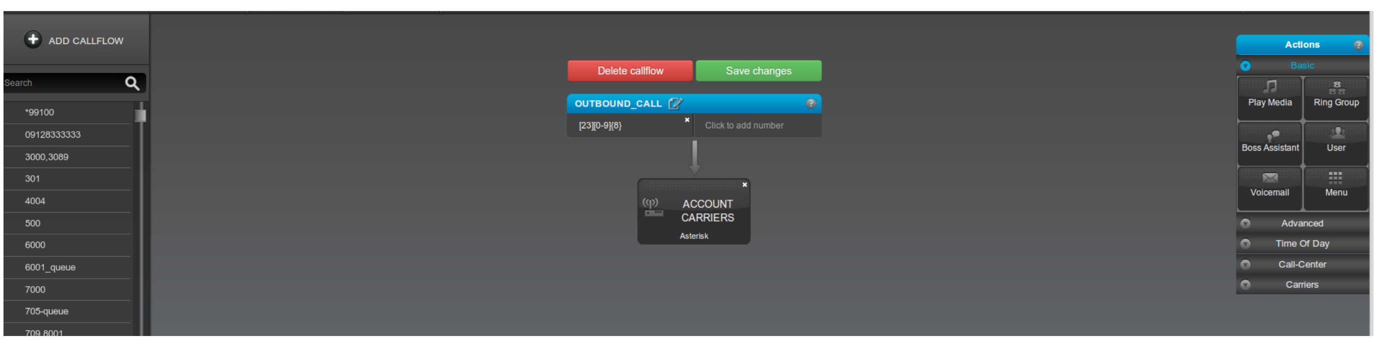
- Gateway type - > type of gateway API, ex: Digium switchvox
- Admin username / Admin Password : credentials of Carrier API



Carrier permissons tab, this is to allow/disallow access to internal callflows from carrier, and set profile id



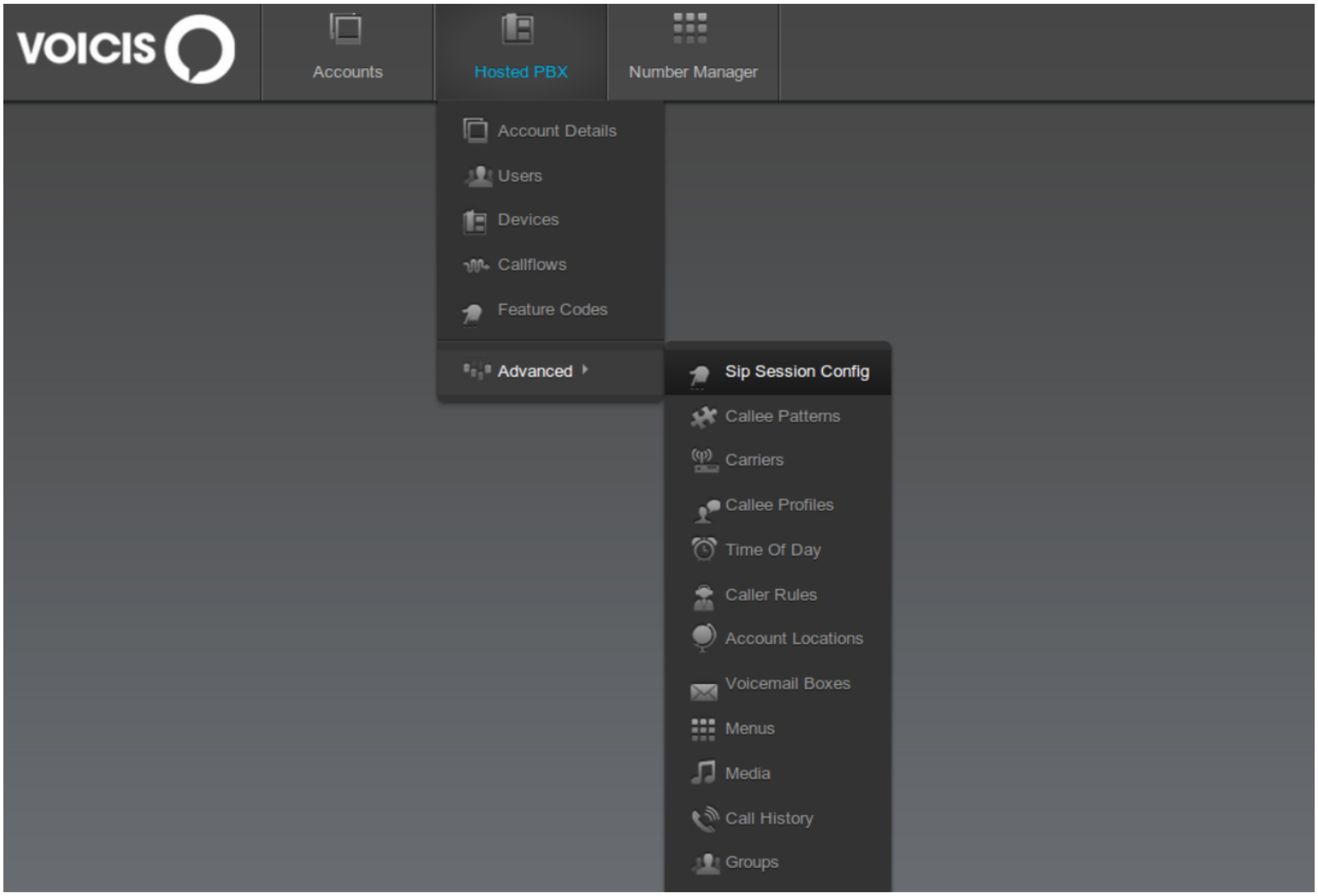
Now we are able to use this carrier in callfow, Create a new call flow and define some regexp in "extension" configuration, then select carrier object, as you can see on image bellow:



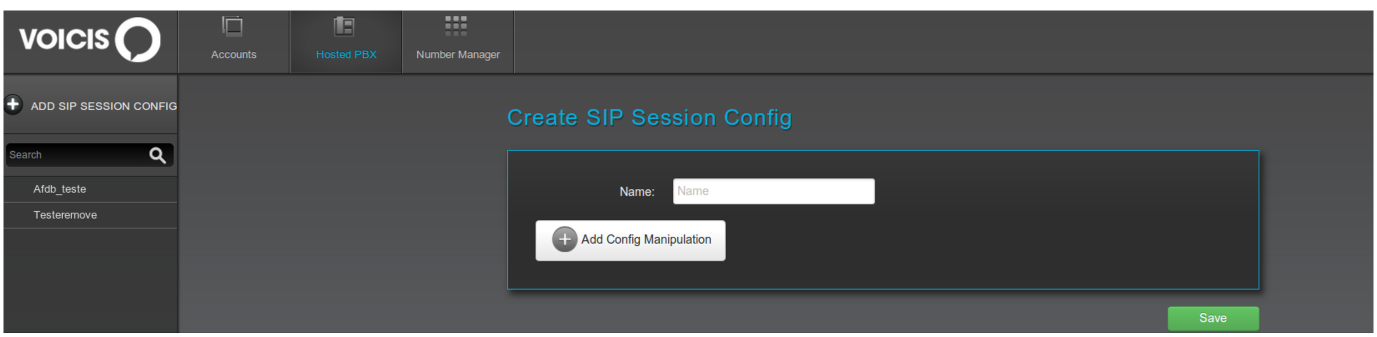
ENG - 7.19 - Session Config

Sip Session Config is used to manipulate sip messages such as request uri, from uri, codecs.

In order to configure a sip session config, fist you need access to Hosted PBX menu then Advanced and select sip_session_config option, as you can see on image bellow:



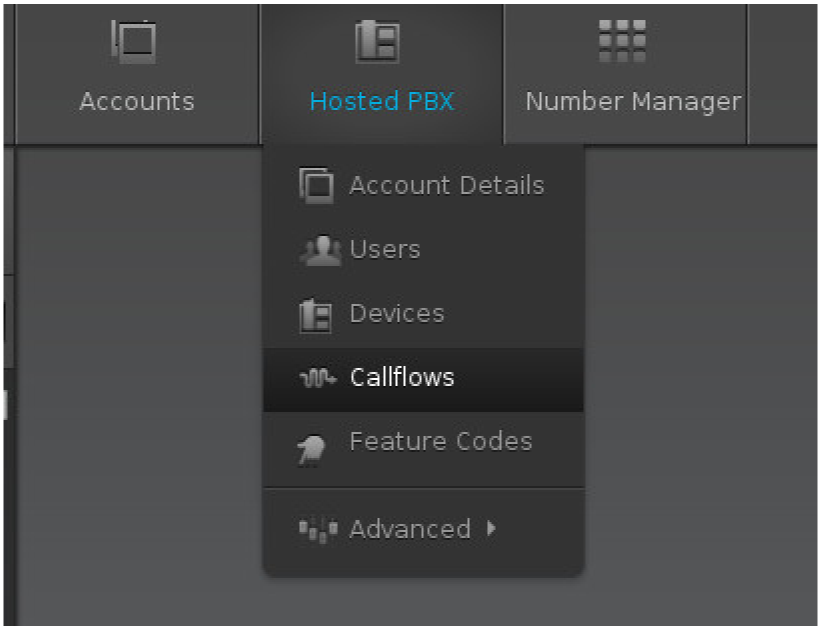
Add a new sip session config selecting the option on left side called "ADD SIP SESSION CONFIG":



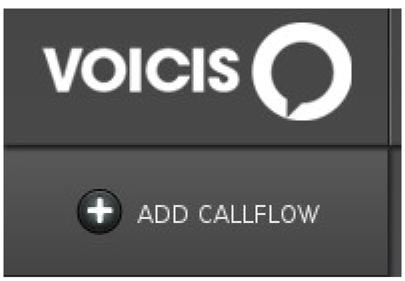
ENG - 7.20 - ENUM

ENUM Configuration

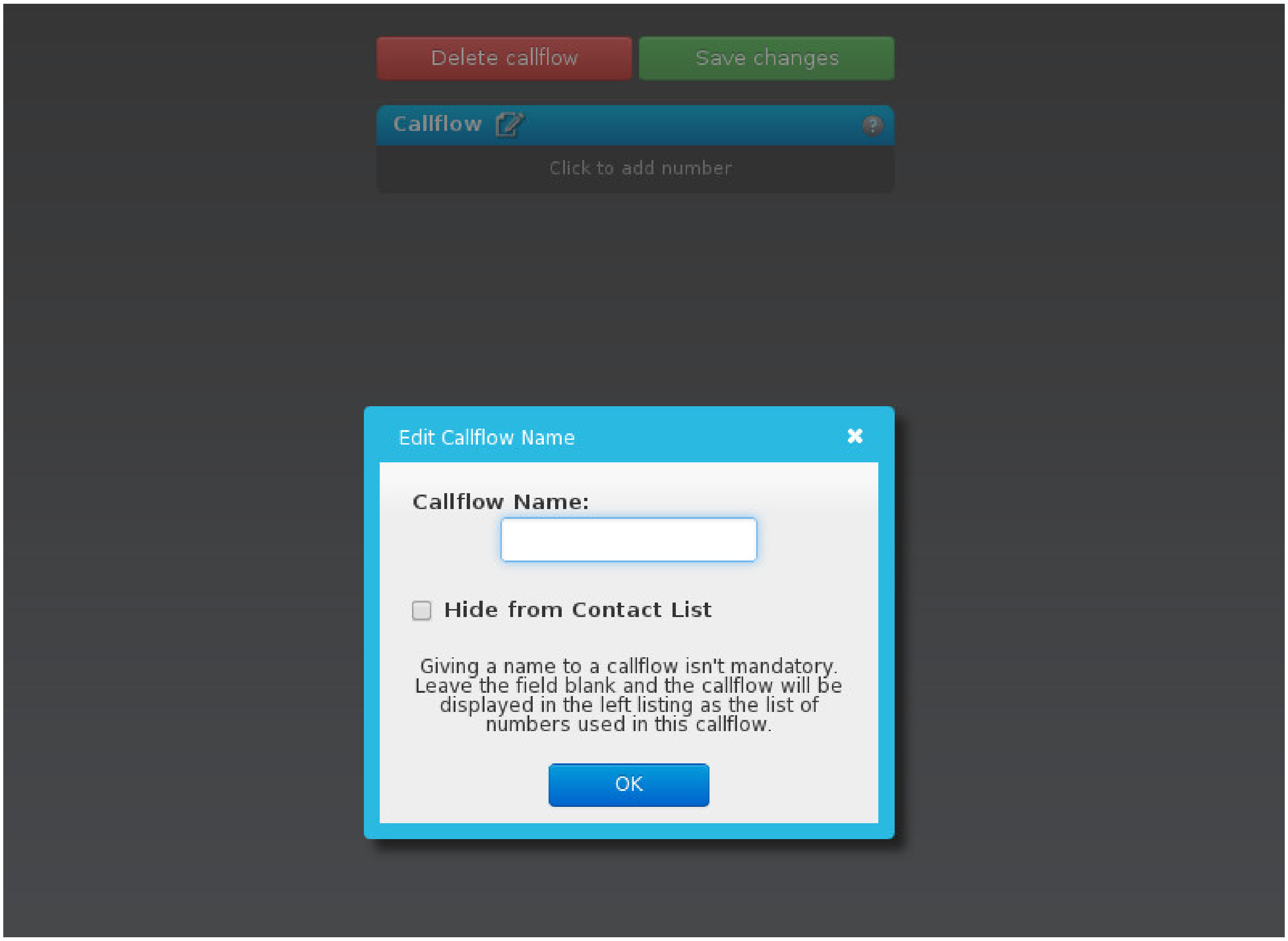
In order to configure a carrier, fist you need access to Hosted PBX menu then select Callflow option, as you can see on image bellow:



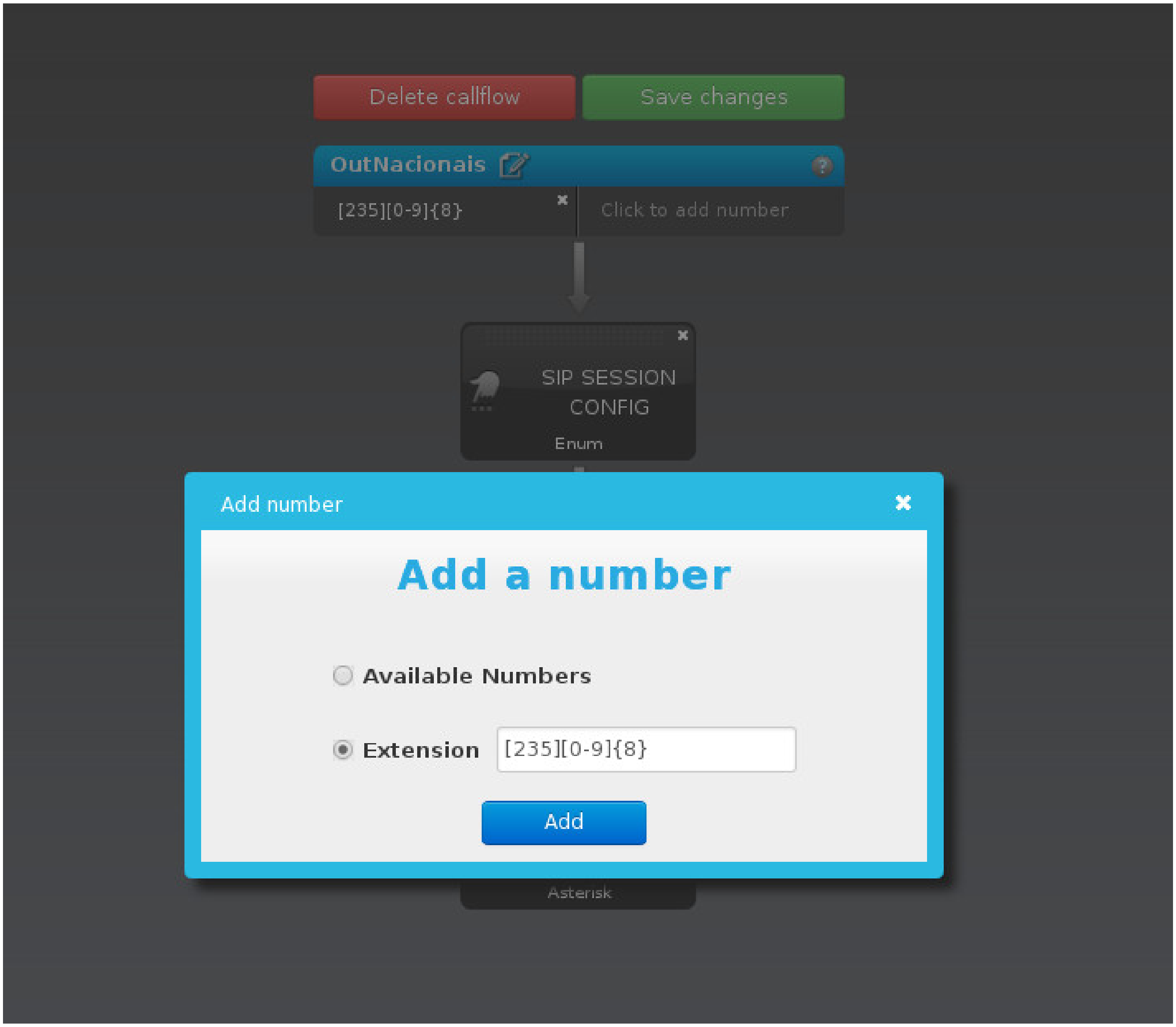
Click Add Callflow



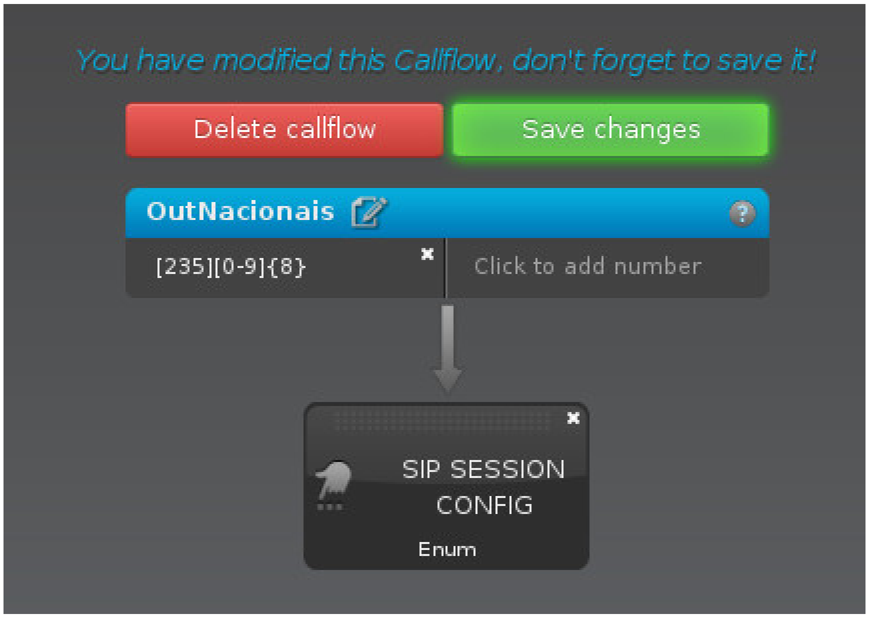
Define callflow name



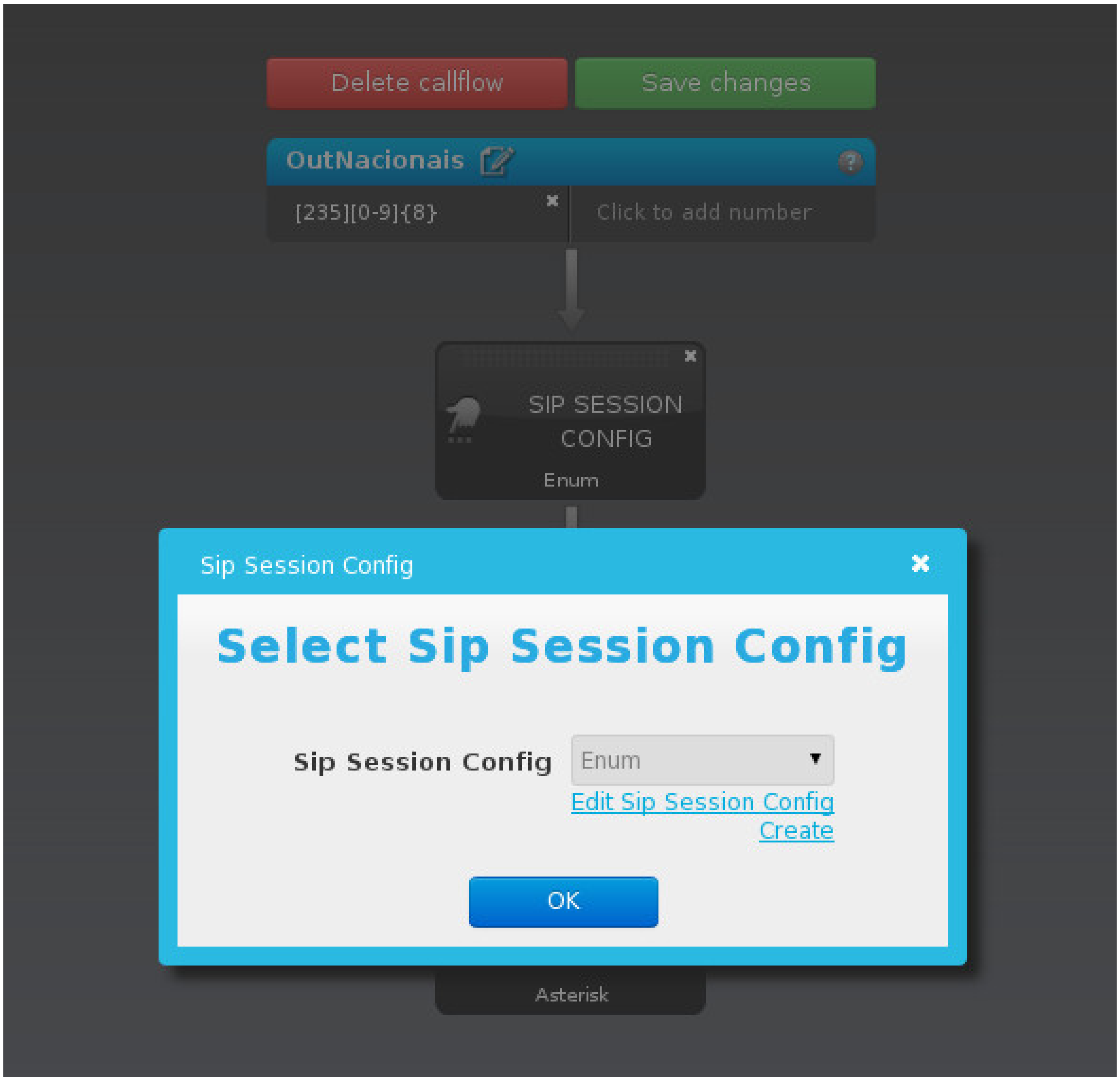
Click "Add Number" to add regular expression to match prefix of outgoing ENUM Calls



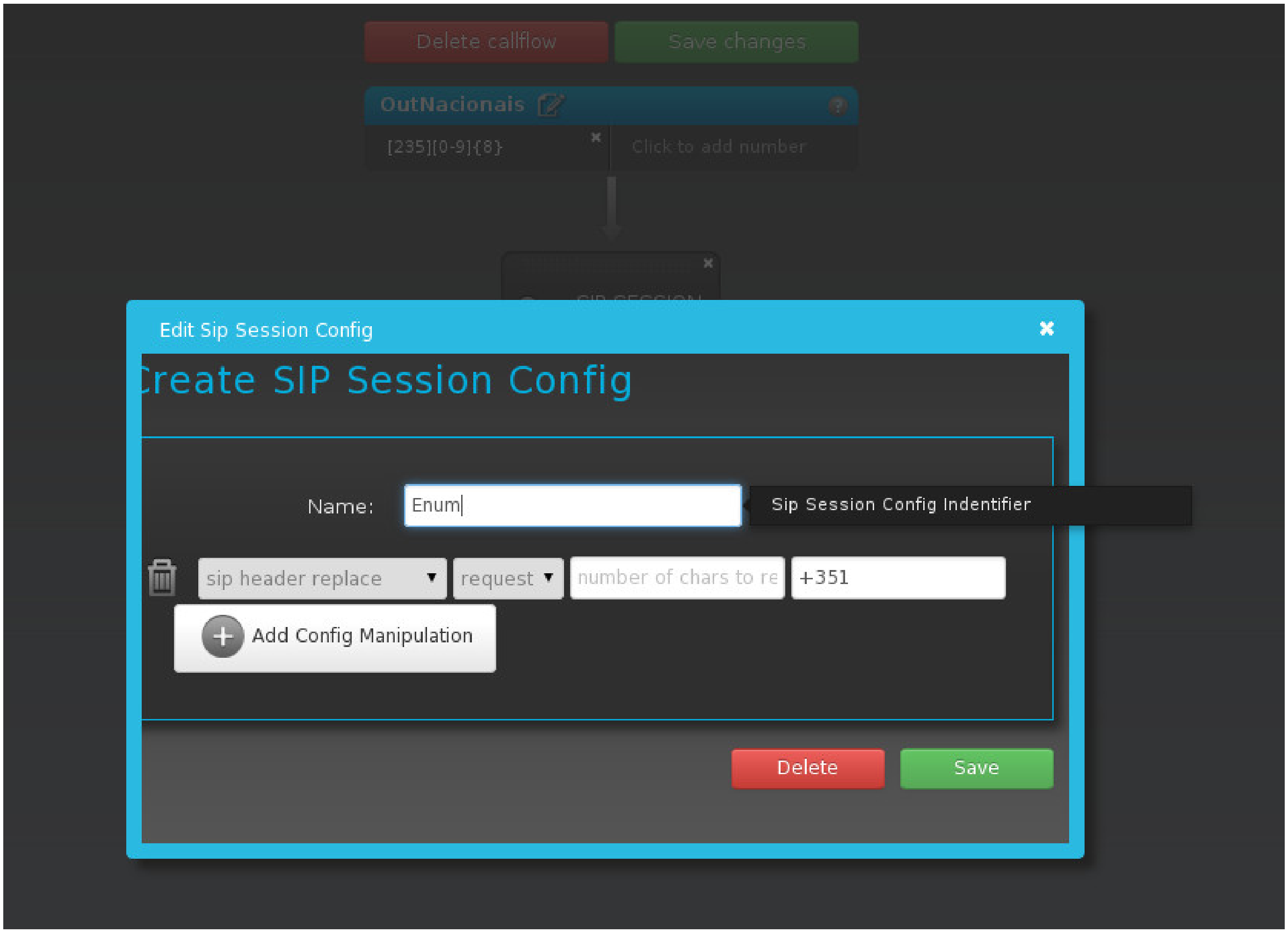
Add Sip Session Config object



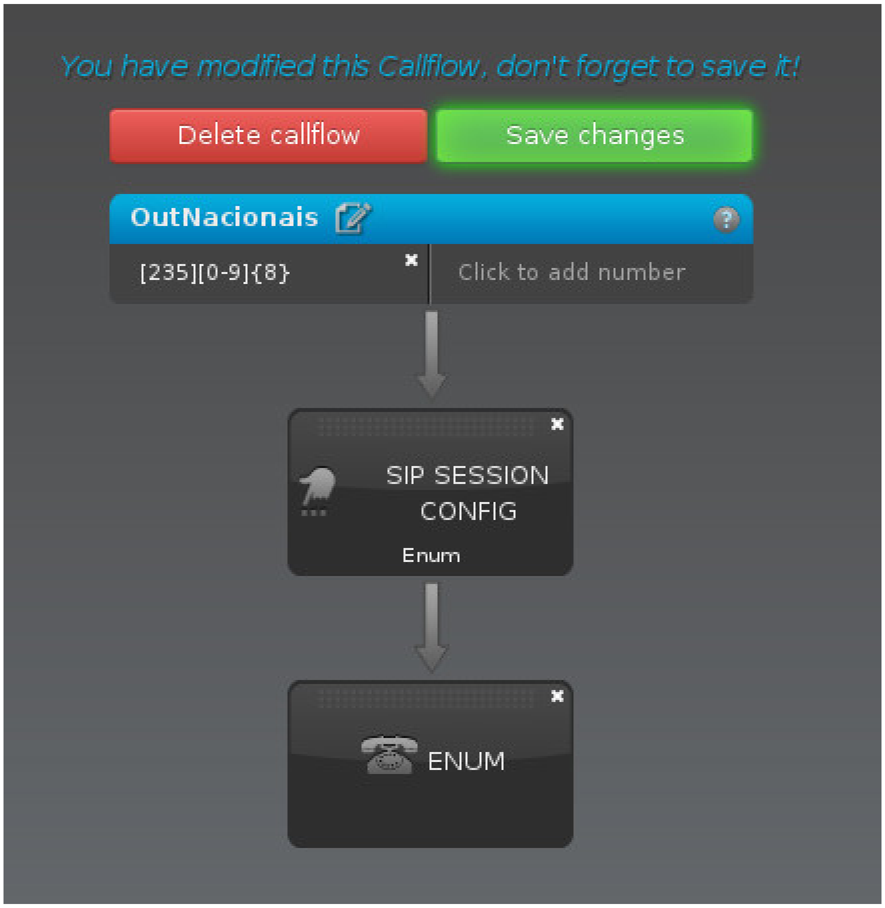
Create new Sip Session Config



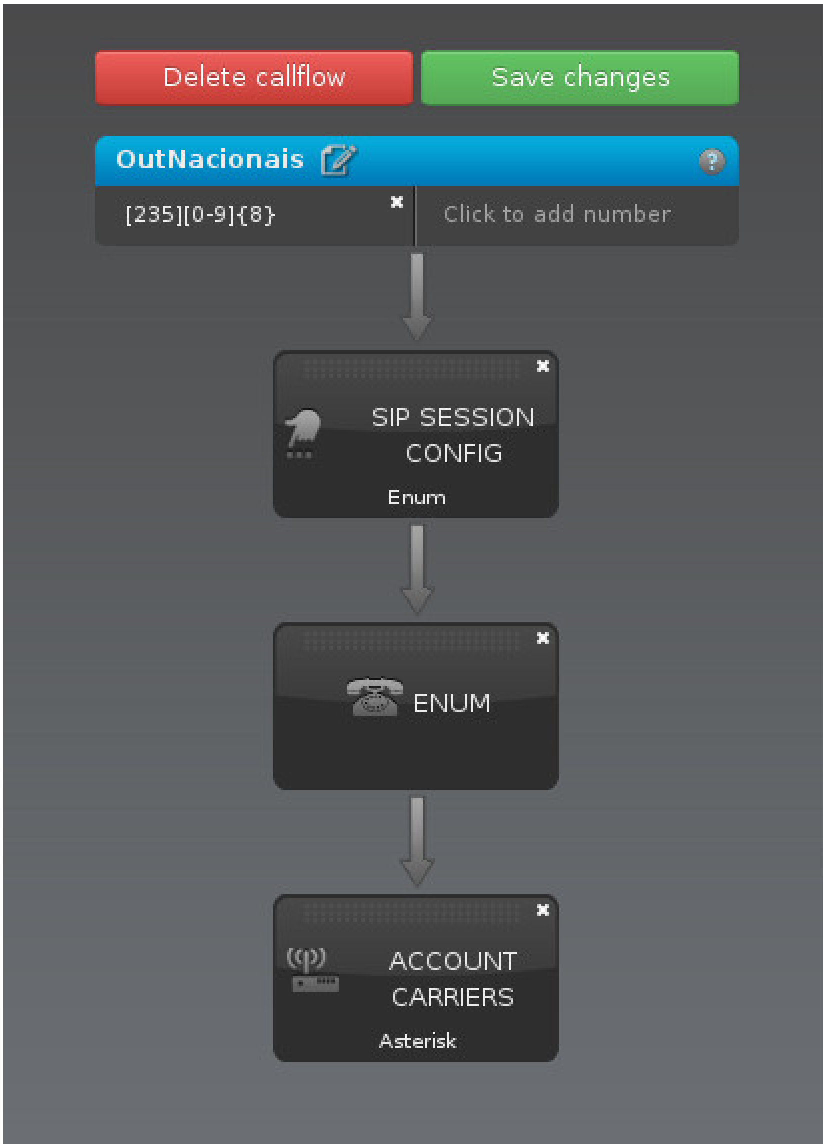
Define Sip Session Config



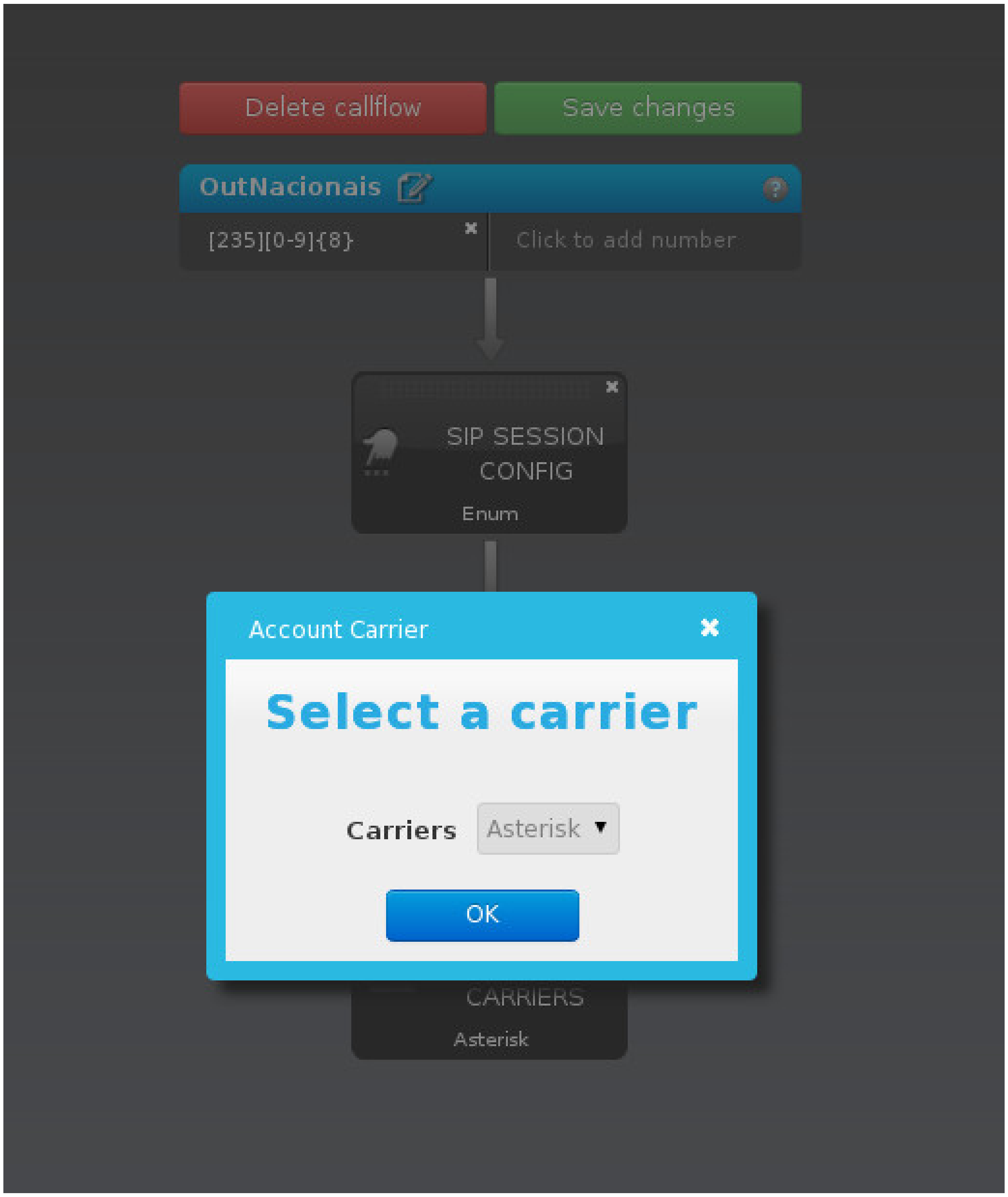
Add ENUM Object



Add Account Carrier



Define Carrier



ENG - 8 - Outbound Management

ENG - 8.1 - Local Gateway

Each account manager can one or more local gateway.

We can define 3 types of local gateway:

Type	Characteristic
IP	We use the IP to built the trunk
Registered in	The remote gateway register in Voiciscore with user and pasword
Registered OUT	Voiciscore register with user and pass inot the remote gateway

This local gateway only existe into the local account.

ENG - 8.2 - Global Gateway

The global account manager can one or more global gateway.

There is several type of gateways that are listed in the following window.

The definition of global gateway depends on the type of global gateway.

The global account manager can then provides this outbound gateway to the account manager.

ENG - 8.3 - LCR

The Global account manager can create LCR rules based on severals definitions.

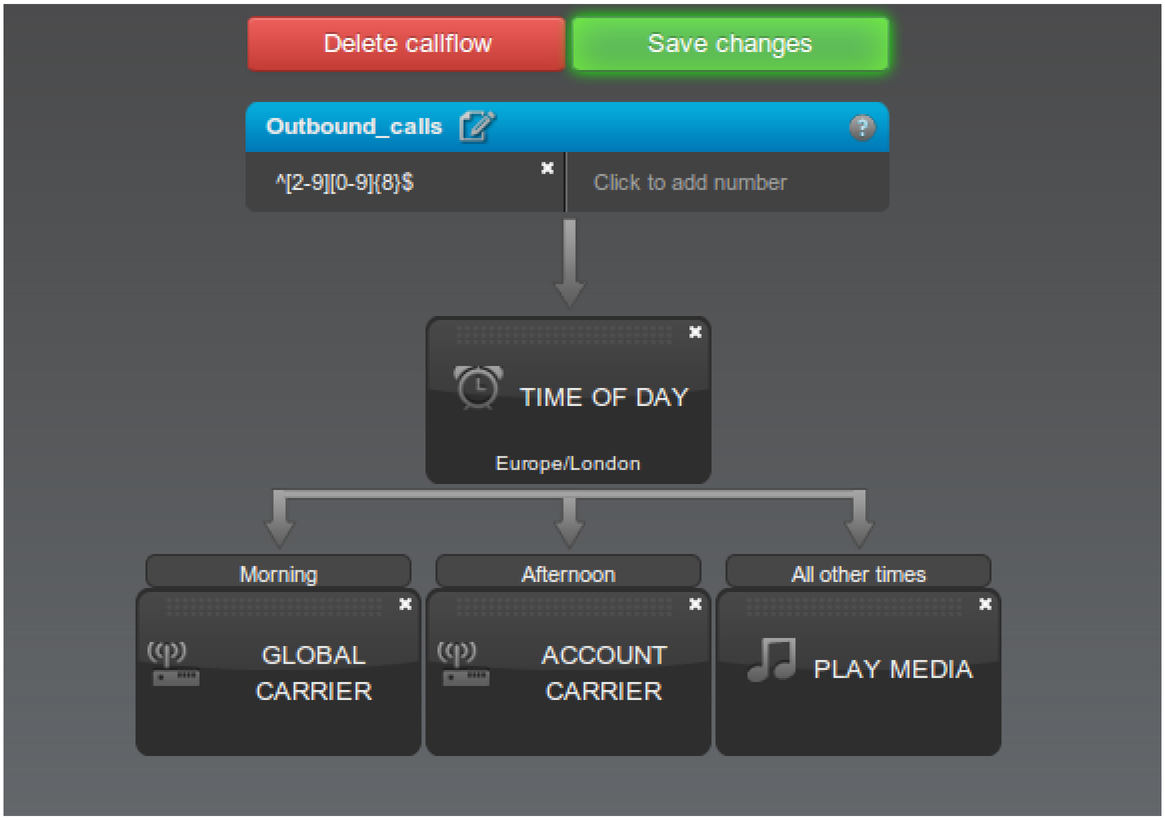
The Global account manager provides then a global geteway to the accounts.

ENG - 8.4 - Outbound Callflow

Using Local Gateway and Global Gateway the account manager can create a complex outbound callflow based on the calling number or the called number.

For Called number the criteria is a regex.

Example for outbound routes



ENG - 9 - Information flow

ENG - 9.1 - Phone Provisioning

Phone Configuration

```
Phone                DNS                 API              SIP Proxy
  |   Get API Address |                   |                   |
  |------------------>|                   |                   |
  |   API IP Address  |                   |                   |
  |<------------------|                   |                   |
  |  Get Phone Config |                   |                   |
  |-------------------------------------->|                   |
  |      MAC*.xml     |                   |                   |
  |<--------------------------------------|                   |
  |    Get SIP Proxy  |                   |                   |
  |------------------>|                   |                   |
  |   SIP IP Address  |                   |                   |
  |<------------------|                   |                   |
  |                   |                   |                   |
  |                   |                   |                   |
  |      REGISTER     |                   |                   |
  |---------------------------------------------------------->|
  |       200 OK      |                   |                   |
  |<----------------------------------------------------------|
  |                   |                   |                   |
```

Phone Configuration Change

```
API                SIP Proxy            Phone
 |                     |                   |
 | Update Phone Config                     |
 |------------------>|                     |
 |                     | Update Phone Config
 |                     |------------------>|
 |     Get MAC*.xml    |                   |
 |<--------------------------------------|
 |                     |                   |
```

Phone Register

```
Phone                DNS                SLBx                 CCP
CPD
  |                   |                   |                   |
|
  |      DNS Query    |                   |                   |
|
  |------------------>|                   |                   |
|
  |      REPLY SBx    |                   |                   |
|
  |<------------------|                   |                   |
|
  |      REGISTER     |                   |                   |
|
  |-------------------------------------->|                   |
|
  |  401 Unauthorized |                   |                   |
|
  |<--------------------------------------|                   |
|
  |      REGISTER     |                   |                   |
|
  |-------------------------------------->|                   |
|
  |                   |                   |      REGISTER     |
|
  |                   |                   |------------------>|
|
  |                   |                   |                   |  User
Rigth Check |
  |                   |                   |
|------------------>|
  |                   |                   |                   |      SQL
Query    |
  |                   |                   |
|------------------>|
  |                   |                   |                   |
SQL       |
  |                   |                   |
|<------------------|
  |                   |                   |       200 OK      |
|
  |                   |                   |<------------------|
|
  |       200 OK      |                   |                   |
|
  |<--------------------------------------|                   |
|
  |                   |                   |                   |
|
```

ENG - 9.2 - Call Flow

Call Between two endpoints (in this case Switchbox)

```
Switchvox1              DNS             Voiciscore          Switchvox2

     |                   |                   |                   |
     |      DNS Query    |                   |                   |
     |------------------>|                   |                   |
     |     DNS Answer    |                   |                   |
     |<------------------|                   |                   |
     |       INVITE      |                   |                   |
     |-------------------------------------->|                   |
     |                   |                   |       INVITE      |
     |                   |                   |------------------>|
     |                   |                   |       200 OK      |
     |                   |                   |<------------------|
     |       200 OK      |                   |                   |
     |<--------------------------------------|                   |
     |         RTP       |                   |                   |
     |---------------------------------------------------------->|
     |         RTP       |                   |                   |
     |<----------------------------------------------------------|
     |                   |                   |                   |
```

Call between Phone Registered on Voiciscore and PSTN

```
Phone                SLB                 MGW               PSTN
  |                   |                   |                   |
  |       INVITE      |                   |                   |
  |------------------>|                   |                   |
  |  401 Unauthorized |                   |                   |
  |<------------------|                   |                   |
  |       INVITE      |                   |                   |
  |------------------>|                   |                   |
  |                   |       INVITE      |                   |
  |                   |------------------>|                   |
  |                   |       200 OK      |                   |
  |                   |<------------------|                   |
  |       200 OK      |                   |                   |
  |<------------------|                   |                   |
  |         RTP       |                   |                   |
  |-------------------------------------->|                   |
  |                   |                   |         RTP       |
  |                   |                   |------------------>|
  |                   |                   |         RTP       |
  |                   |                   |<------------------|
  |         RTP       |                   |                   |
  |<--------------------------------------|                   |
  |                   |                   |                   |
```

ENG - 9.3 - Load Balancing

Phone Registration Load Balancing

```
Phone1              Phone2                DNS                 SLB1
SLB2
   |    Get SIP Proxy  |                   |                   |
|
   |-------------------------------------->|                   |
|
   |                   |    Get SIP Proxy  |                   |
|
   |                   |------------------>|                   |
|
   |       Proxy 1     |                   |                   |
|
   |<--------------------------------------|                   |
|
   |                   |       Proxy 2     |                   |
|
   |                   |<------------------|                   |
|
   |                   |                   |                   |
|
   |                   |                   |                   |
|
   |      REGISTER     |                   |                   |
|
   |---------------------------------------------------------->|
|
   |       200 OK      |                   |                   |
|
   |<----------------------------------------------------------|
|
   |                   |                   |                   |
|
   |                   |      REGISTER     |                   |
|
   |
|---------------------------------------------------------->|
   |                   |       200 OK      |                   |
|
   |
|<----------------------------------------------------------|
   |                   |                   |                   |
|
```

ENG - 10 - DHCP

The phones will receive the DHCP configuration to function properly.

a) DHCP Server

b) Cisco IOS Router

```
!
ip dhcp pool phone
  network 10.0.0.0 255.255.255.0
  default-router 10.0.0.1
  dns-server 213.58.213.229 213.58.213.245
!
```

c) Linux dhcpd

```
# dhcp to issue IP's to other computers in your network (separate from the phones)
# specify your network and subnetmask
subnet 10.100.32.0 netmask 255.255.255.0
{
option routers 10.100.32.1; # specify default gateway
option subnet-mask 255.255.255.0;
option domain-name "voicis.local"; # default DNS domain name
option domain-name-servers 213.58.213.231,213.58.213.232; # DNS Server
option time-offset -21600; # Central Standard Time
range 10.100.32.70 10.100.32.99; # if you want dhcp to issue ip's to other

pool {
allow members of "poly550";
range 10.100.32.60 10.100.32.69;
}

# --- End Of File ---#
```

In case it is not possible to configure the phone through DHCP. Must be set the DNS servers to point to the voiciscore.

Nome	IP Address
DNS1	213.58.213.229
DNS2	213.58.213.245

ENG - 20 - Firmware and Software Version

ENG - 20.1 - SNOM

Firmware versions used for the POC.

Model	Firmware Release	Remarks
710	8.7.3.25.5	
720	8.7.3.25.5	

760	8.7.3.25.5	
821	8.7.3.25	
M9 (Portable)		
Meeting Point	8.7.3.25	

ENG - Troubleshooting

Capturing the user's screen, direct access to telephone, from behind NAT.

http://iphone_address/screen.bmp

Example:

Call from 100 to 102:

Telephone 100:

Telephone 102:

Access to Kibana

http://213.58.213.241