USDigitalPhone
Call Anytime 731-681-2681

Hosted PBX User Manual

For More Information Visit

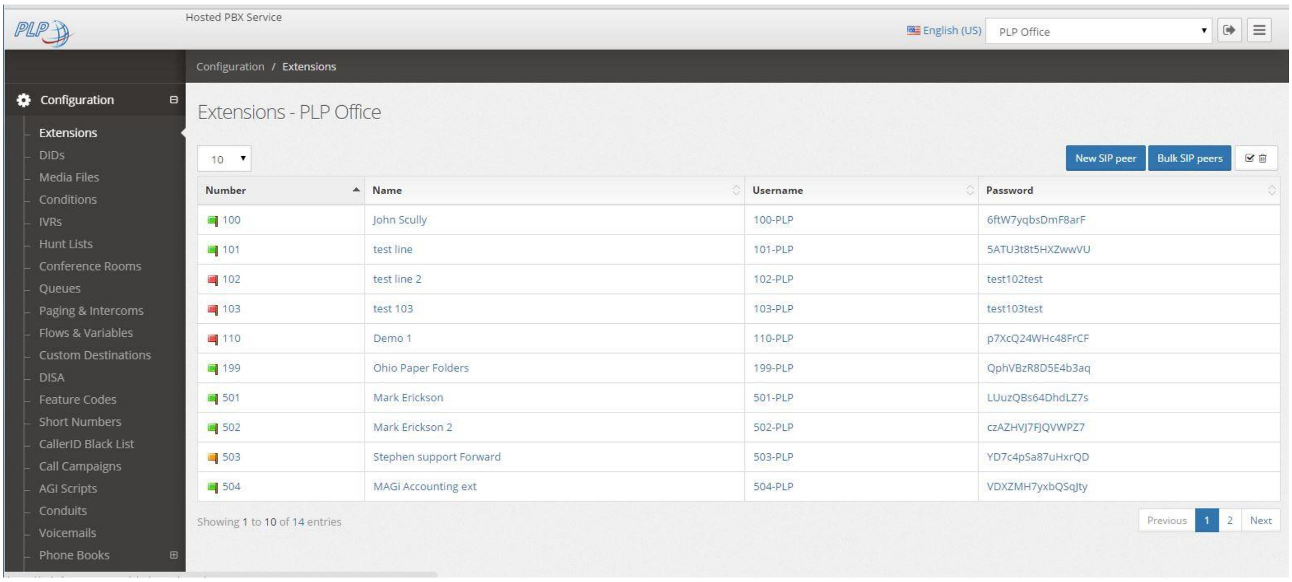
www.usdigitalphone.com

02/14/2015

Work Area

In the top right corner you can identify in order the *Language Selection Menu*, the *PBX Selection Menu*,

the *Exit Button* and the *Menu Display Toggle Button*.



On the left there are the *Configuration*, *Status* and *Admin Sections* to configure, check and manage your system. The menu available can be configured to show only the one interested for the user, by customizing the user profile in the Admin/User Profiles menu.

Language Selection Menu

Using this menu you can choose the language used by the interface. Please note only the English language is provided. If you want to use another language, you need to provide the translation by yourself using the Admin/Translations menu. More language hooks can be provided. If you don't want to display all the current language hooks available, you can disable them in the Admin/Settings. The language, if available, is automatically chosen based on the language accepted by the browser.

PBX Selection Menu

Using this drop down box, the PBX you choose to work on can be chosen. Please take in mind only the authorized PBXs are shown. Even if you are an admin, you can see only the PBXs enabled on your account. If you want, admin users can automatically see all the PBXs by enabling the “Admins see all PBXs” checkbox in the Admin/Settings menu.

Menu Display Toggle Button

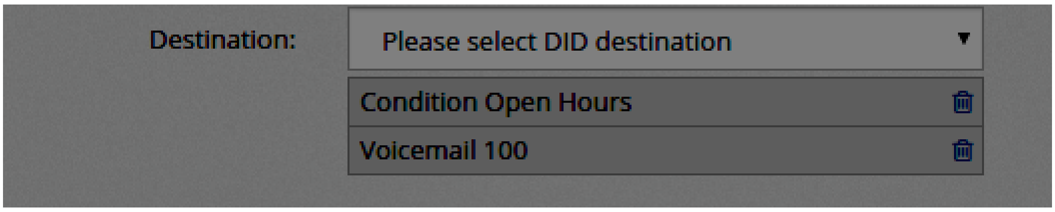
The website is responsive, so it will resize based on the actual screen size. If you need more space you can toggle the left menu display.

Mobile version

The website can be comfortably used also from mobile devices with limited screen size.

Multiple select

In almost every place of the interface, when a select box is shown, multiple options can be picked up,



ordered or deleted. There is no limit about the number of options to be selected.

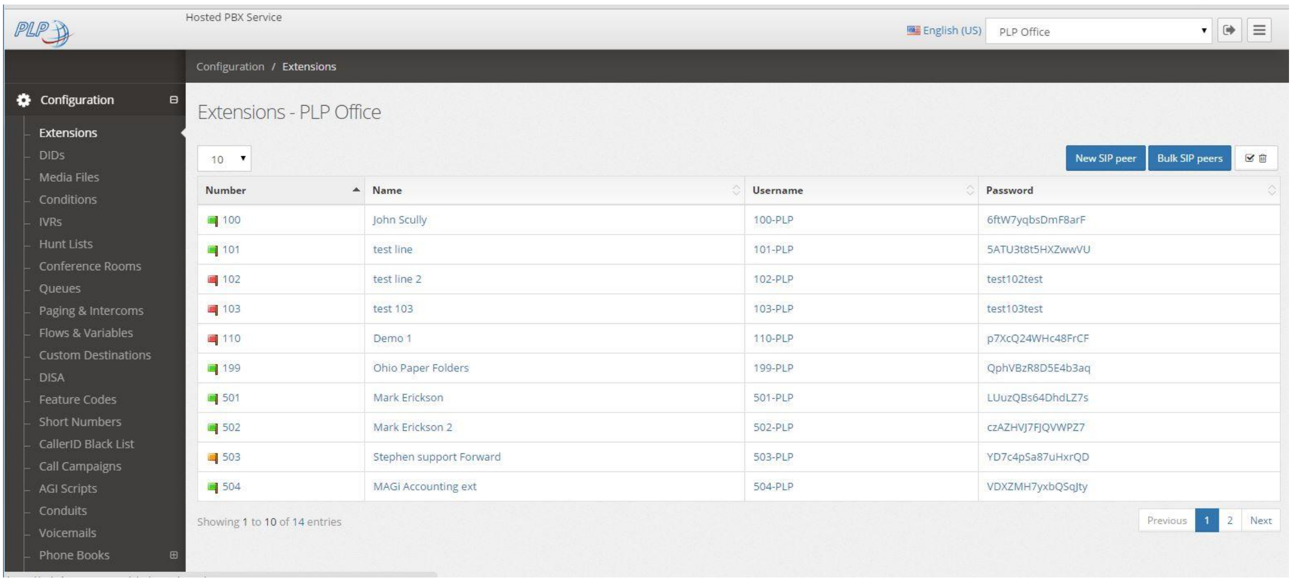
Configuration Section

The Configuration Section is used to configure every working aspect of the PBX. It can used by admin and non-admin users. The options selected within this menu are applied only to the selected PBX, using the “PBX Selection Menu”.

Extensions

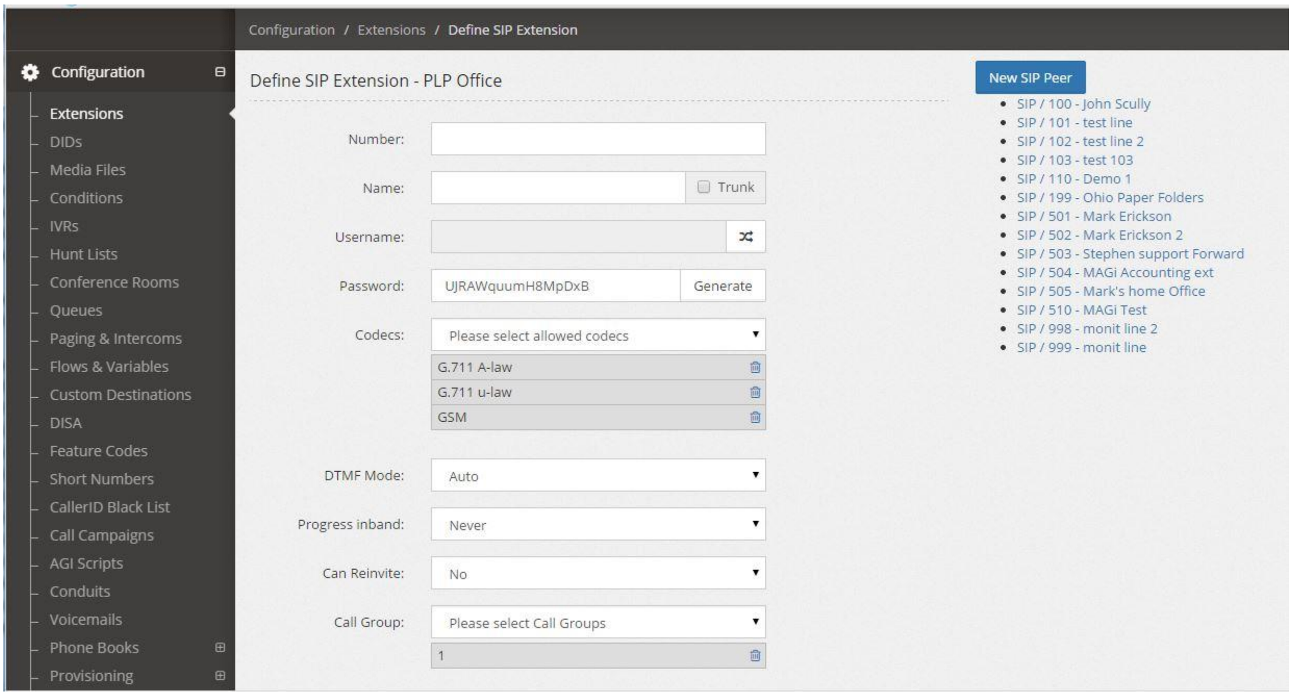
The lists of extensions defined for the selected PBX are shown along with the Caller ID, username and password associated. You can create one or multiple new extensions, using the buttons on the top right.

For each extension, a small icon will display the status; green for registered, red for off line, yellow for not yet registered.



New SIP Extension/Define SIP Extension

The definition of an extension is comprised of multiple sections. The most important one is the general



one, where you can define the internal number for the extension and the password.

The password is automatically generated – please DO NOT change this to an “easy” password, as the user will never need to know it – it is automatically placed into the configuration for the IP Phone.

The username is automatically generated based on the extension number and the code assigned to the PBX. Extension number and SIP accounts are tied together. The username format <extension>-<PBX code> is mandatory (see later for exceptions) and cannot be changed.

The name provided will be used as Caller ID for internal calls. This means the Caller ID on the phone will be overwritten with the one specified here. If you don't want to have the Caller ID associated to the one configured, but you would rather use the Caller ID coming from the phone (for example because the extension is not assigned to a single phone, but because it is assigned to another PBX with multiple extensions) you can set the “**Trunk**” checkbox. The trunk setting will also affect incoming calls to the phone (or PBX). If the “Trunk” checkbox is set, the SIP INVITE sent to the account will include the number dialed.

Username is automatically generated by adding the PBX code to the number provided. The format used by default is the “-” sign. But some phones have been found to not accept the minus (“-“) sign. The joining character can be changed by pressing on the double arrows. Take in mind the usage of “_” sign is discouraged and needs to be used only when really needed.

Password can be auto generated by clicking on the “Generate” button. A new password is generated every time using the random number generated. It is highly advisable to use long and completely random passwords.

Codecs: Every extension can have a broad range of codecs allowed. Please remember that if you use the G.729/723.1 codecs, even if they are listed in the system, you may have to pay royalty fees to the G.729/723.1 patent holders for using their algorithm.

DTMF Mode is selectable between auto, info, inband and RFC 2833. Please check the phone configuration and the provider support for choosing the right DTMF setting. The most widely accepted format is RFC 2833.

Progress inband forces the system to generate ringing tones.

NAT setting is important when the phone is behind a NAT. Use *force_rport, comedia* in almost all the cases. If you experience one way audio, then check the NAT setting.

The Can reinvite feature, allows two endpoints, like two phones, or the phone and the provider, to exchange the RTP data directly, without routing through the PBX. Usually if one party is behind NAT, you may experience one way audio. Usually set to No.

Qualify, allows the PBX to contact the phone periodically to check if it is still online. This has the benefit of “open” connection tracking on the firewall that would be between the phones and the Internet. Connections are made every second. If you have slow phones, you can increase the time so that it may have time to wait for an answer.

Call groups/Pickup groups define who is permitted to perform a pick-up for which calls. If call group and pickup group matches, then it is possible to pickup, using the specified feature code.

Note: you need to define the feature code to use.

Voicemail MWI allows you to assign the MWI on the phone for a voicemail.

Call Limit sets the max number of channels allowed to be used by a phone. Setting it to 1 doesn't generally allow it to transfer calls.

Do Not Disturb (DND) sets the extension in DND mode. This is a server assisted DND. It doesn't affect the phone DND set.

Inbound Dial Timeout sets the time in seconds. One extension has to ring before going to the additional “No Answer” destination. You can avoid setting a Dial Timeout value, allowing the default value will be used.

Recording

Always Record Sets the recording preference for the extensions. If they are set to “Yes”, all the phone calls made by the extensions are recorded. If set to “Yes, but allows stopping” or “No, but allows starting”, then the recording can be respectively turned off or on, by using the predefined #0 and #1 DTMF sequence while on the phone. The recorded file will be available for download in the Status/Call History menu.

Email Recordings Allows to set an email address to send the recordings, once the call completes.

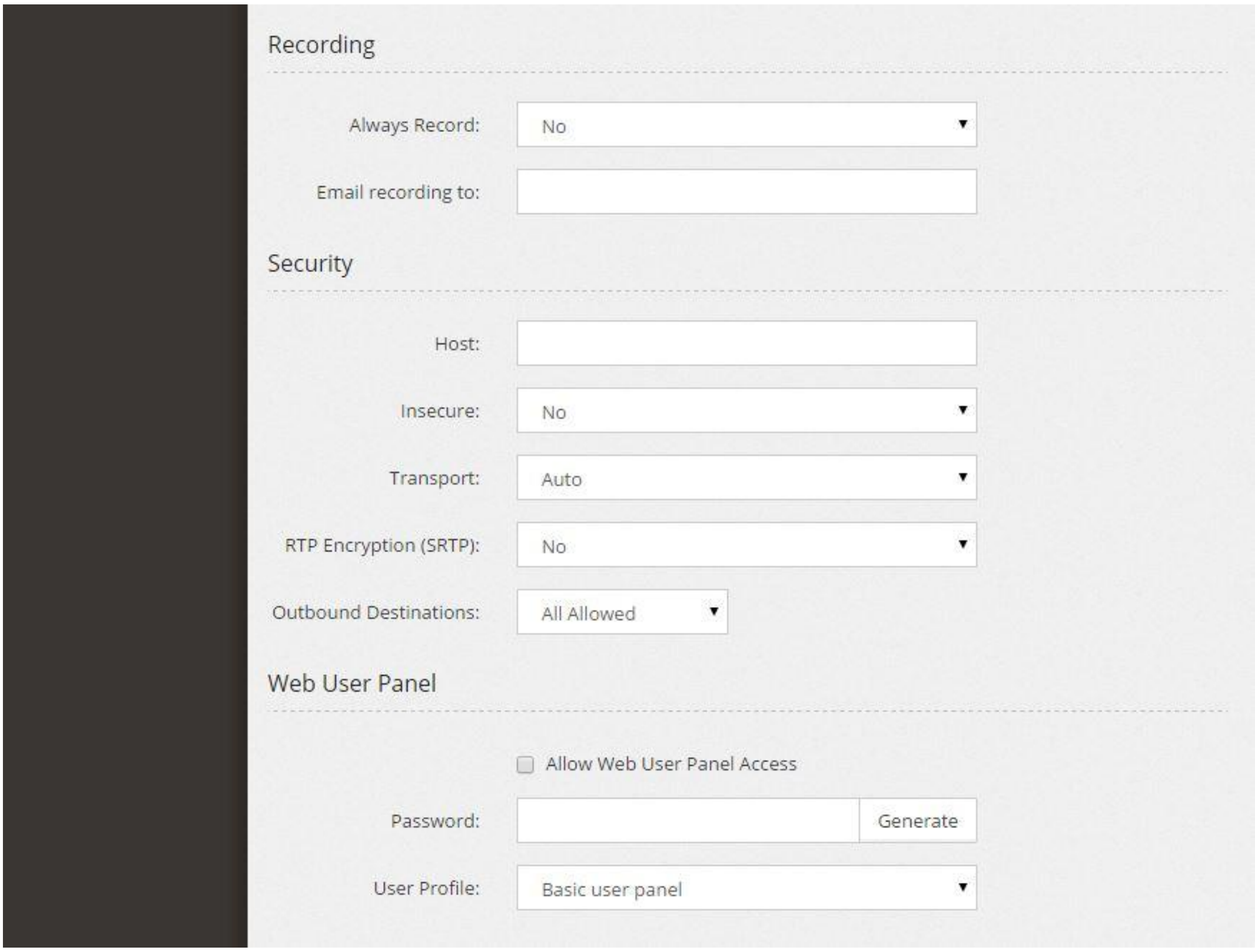
Security

Host can be “dynamic”, accepting registration from any IP. Or it can be assigned to a specific IP address. In this way, no registration is needed.

Insecure allows the peer to be authenticated using the IP.

Transport permits to use a different transport for the signaling. If TLS is selected, it is needed to add a certificate to Asterisk. Please check the Appendix for special configuration.

RTP Encryption provides encryption to the RTP (audio) part. The key is transmitted over the SIP channel, so it will be useless to use it, without setting the transport to TLS



Outbound Destinations permits restriction of numbers the extension can dial. In other words, certain call types can be restricted from designated phones. For example, the phone placed in the kitchen of the office, can be restricted to place international calls.

The Outbound destination can be:

All Allowed: Allowing every number

All Prohibited: The phone cannot place outbound calls

Allowed if matches: The call is allowed if the number dialed matches the Regex associated

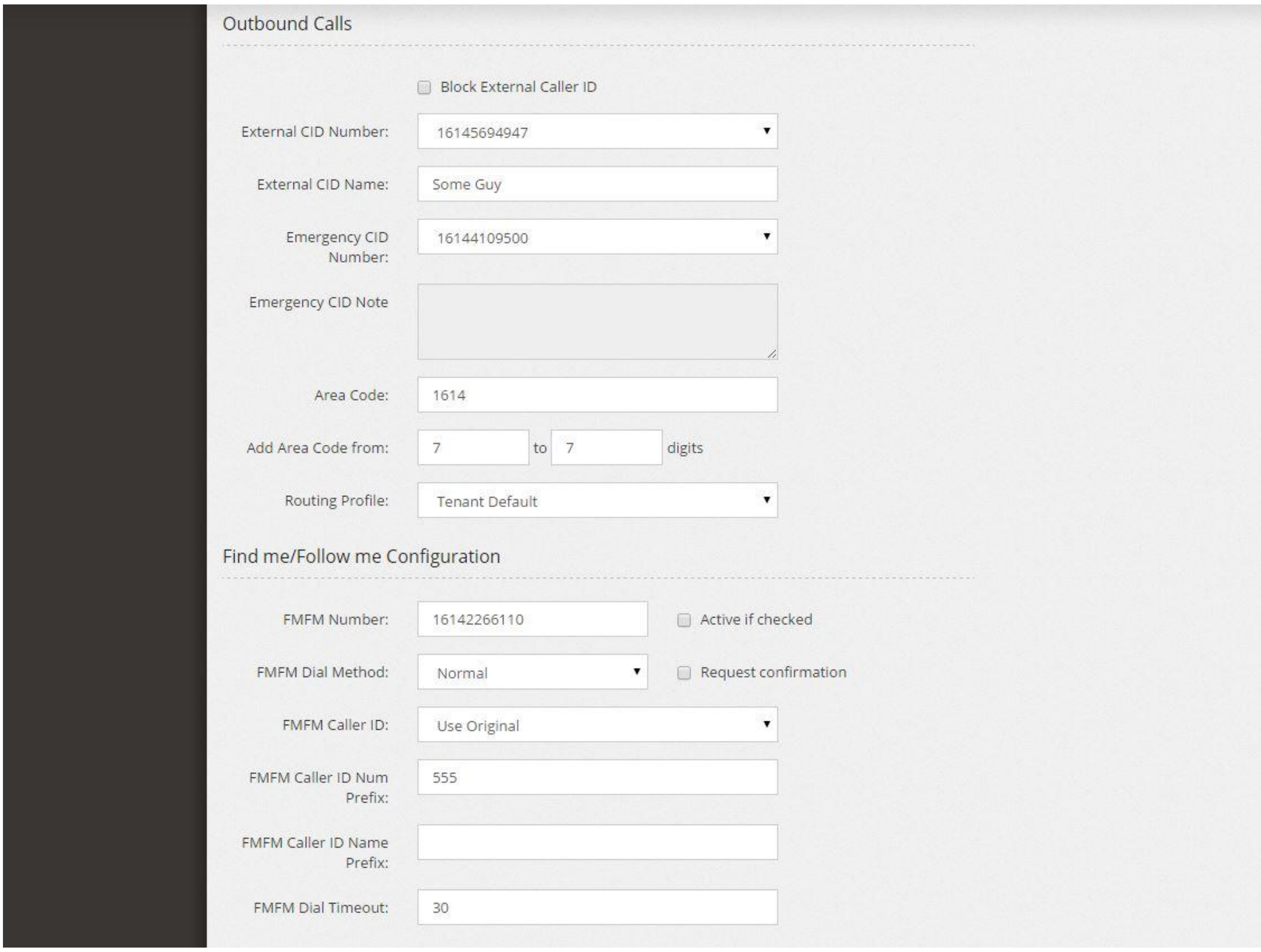
Prohibited if matches: The call is prohibited if the number dialed matches the Regex associated

Web User Panel

Enabling the web user panel allows login to the web interface, providing the extension username, and the provided web user panel password. It is not possible to use the SIP password to login.

User Profile defines the user profile assigned to the user connecting to the web user panel.

Outbound Calls



This section allows configuring how the call is managed when dialing out the local pbx. Use of the Caller ID can be blocked by selecting the "**Block External Caller ID**" checkbox. This checkbox can be selected/unselected using a feature code.

The **External CID number** can be chosen among various formats, usually resembling the E.164 standard (15552011234). The External CID number can be chosen only among the DIDs assigned to the PBX.

External CID name allows defining the alphabetic part of the Caller ID.

Emergency CID number allows you to define the Caller ID number to use when an emergency route is used to dial out. This can be chosen among the DIDs marked as "emergency". The location of the DID is shown if entered.

Area Code allows specifying a prefix to add to numbers when the number of digits entered is between the number of digits specified next, inclusive. In North America, where local numbers are always 7 digits, put the area code in the "area code" box, and "7 to 7" in the add area code from boxes. This tells the system to add the area code when a 7 digit number is dialed.

Routing Profile permits assigning the extension a different routing profile than the one assigned to the PBX.

Find me/Follow me Configuration

This feature allows defining a simple “next hop” for calls, when the dialed number is busy, or not available. The FMFM configuration needs to be enabled using the relative check box. It is possible to use a feature code to enable or disable it.

FMFM Number is the number to dial when the extension is busy or not available.

FMFM Dial Method Allows a choice between two dialing methods; “normal” when the FMFM number is dialed after the “Inbound dial timeout” for the extension, and “simultaneous” when the FMFM number is dialed together with the extension number.

Request Confirmation allows asking to the callee to accept the call, playing the standard message or a custom message. If the callee refuses the call, the call is managed as he was BUSY.

FMFM Caller ID allows choosing which Caller ID to display to the called number. Two special options are available:

Use Original will use the Caller ID

Use Incoming DID will use the DID receiving the call as Caller ID

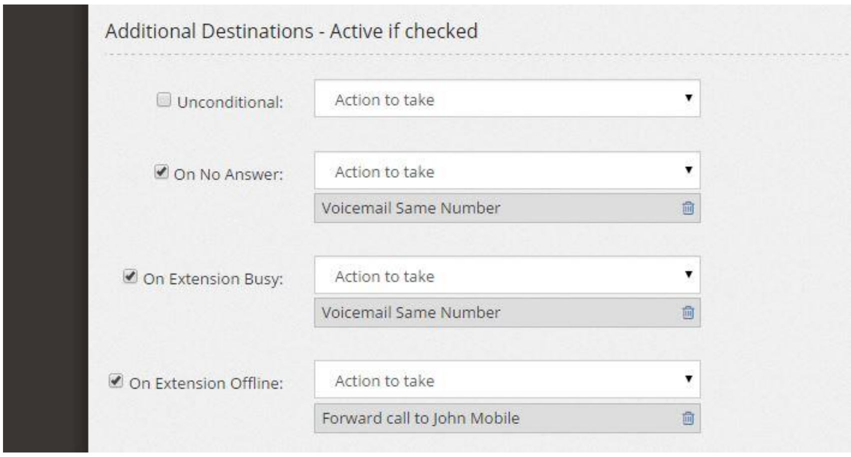
FMFM Caller ID Num and Name prefix defines a prefix to add to the Caller ID chosen when dialing the FMFM number

FMFM Dial Timeout defines the second to dial the FMFM number before going to the Additional Destination, if defined

Additional Destinations

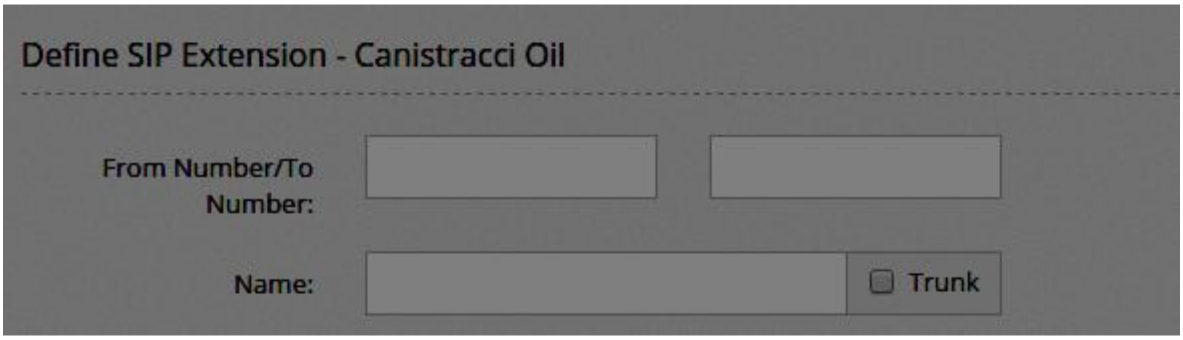
They specify the destination of the call when the extension is not answered, Busy, or Offline. A special destination feature called, “Unconditional”, allows redirecting the phone calls to another destination. Every kind of Additional Destination can be enabled or disabled using Feature Codes.

When defining the Additional Destinations, you may choose a special destination, usually not available, “Voicemail Same Number”. This destination will automatically create a voicemail with the same number as the extension, and a random PIN. If the voicemail was already available, it will be just assigned to the destination-



Bulk extension creation

It is possible to create multiple extensions at once by pressing the “Bulk SIP peer” button. The definition



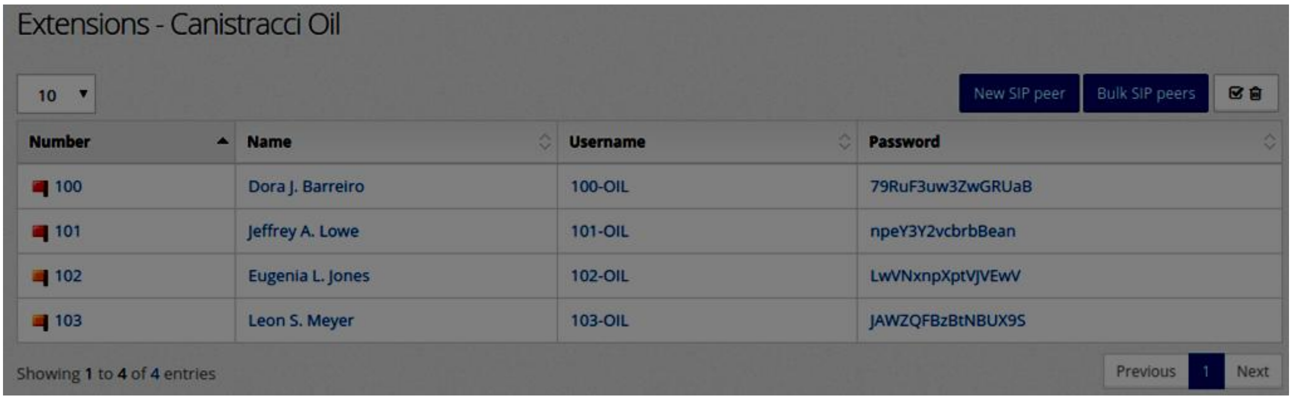
web page will be the same except for the number range requested.

Delete of Extension

To delete an extension, just press the delete button at the end of the extension definition. A message will request confirmation. Deleting the extension will unregister and clean it from the asterisk peer cache, denying further operation for the deleted extension.

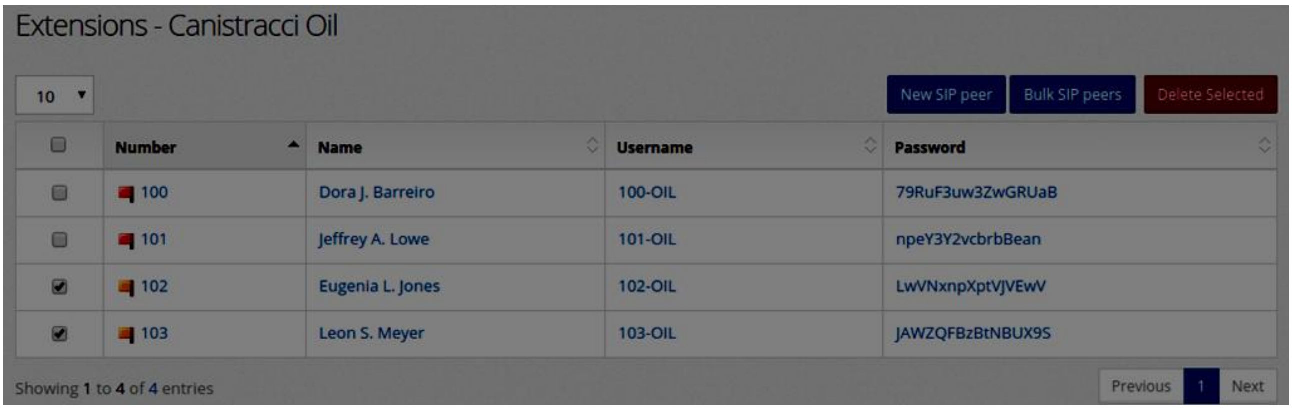
Multiple Delete of Extensions

From the extensions list is possible to delete multiple extensions at once. On the top right corner you



can locate a small garbage icon.

When pressed, a new column will appear on the left, allowing for selecting the extensions to delete, Do this by pressing the newly appeared button “Delete Selected”.



Extensions - Canistracci Oil

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New SIP peer | Bulk SIP peers | Delete Selected

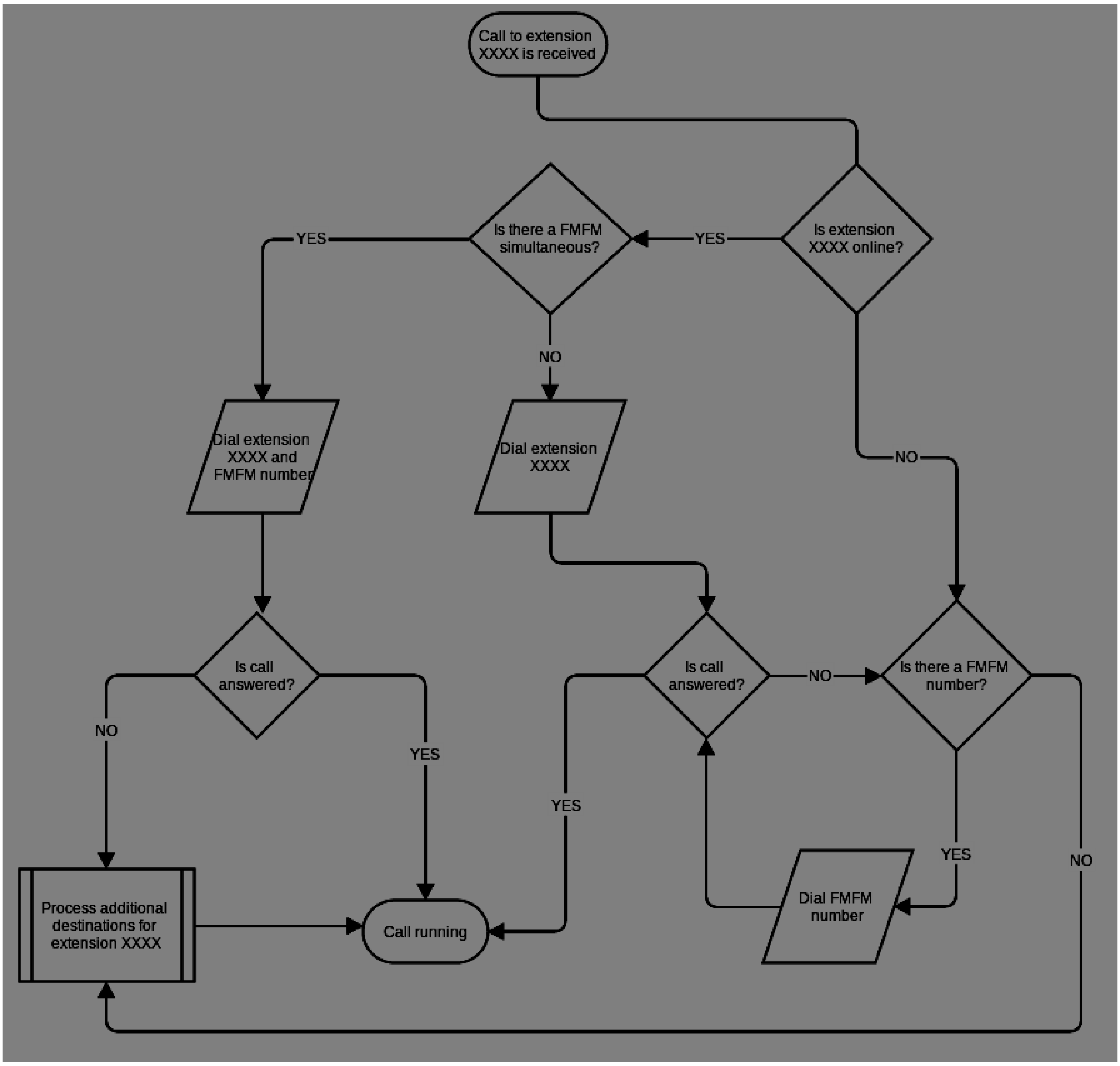
	Number	Name	Username	Password
	100	Dora J. Barreiro	100-OIL	79RuF3uw3ZwGRUaB
	101	Jeffrey A. Lowe	101-OIL	npeY3Y2vcbrbBean
	102	Eugenia L. Jones	102-OIL	LwVNxnpXptVJVEwV
	103	Leon S. Meyer	103-OIL	JAWZQFBzBtNBUX9S

Showing 1 to 4 of 4 entries

Previous 1 Next

How dialing works

Dialing an extension follows a series of steps based on the extension status and its configuration.



Here are some examples:

Online Extension without any FMFM number defined and without any additional destination defined:

Extension is called. Nobody answers. After the “Inbound Ring Timeout” the call drops.

Offline Extension without any FMFM defined and without any additional destination defined:

Extension is called. Call drops immediately

Online Extension with an FMFM number defined and without any additional destination defined:

Extension is called. Nobody answers. After the “Inbound Ring Timeout” the FMFM number is dialed for the FMFM Dial Timeout

Online Extension without FMFM number defined and with an additional destination to Voicemail for the No Answer:

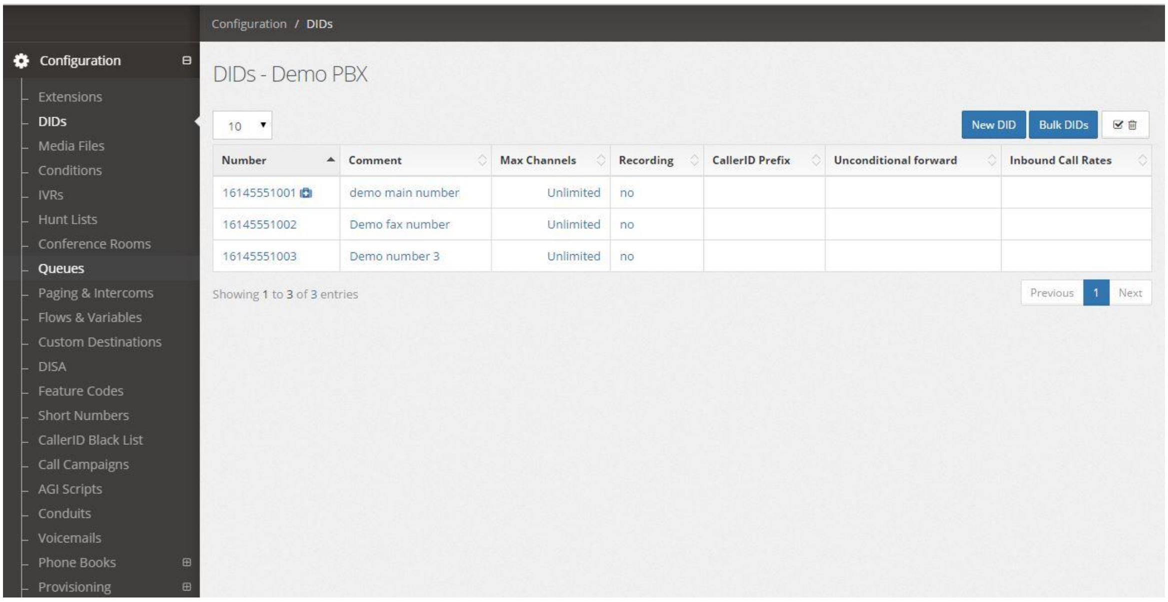
Extension is called. Nobody answers. After the “Inbound Ring Timeout”, voicemail answers

Online Extension with FMFM number defined and with an additional destination to Voicemail for the No Answer:

Extension is called. Nobody answers. After the “Inbound Ring Timeout”, FMFM number is called. Nobody answers. After the “FMFM Dial Timeout”, voicemail answers

DIDs

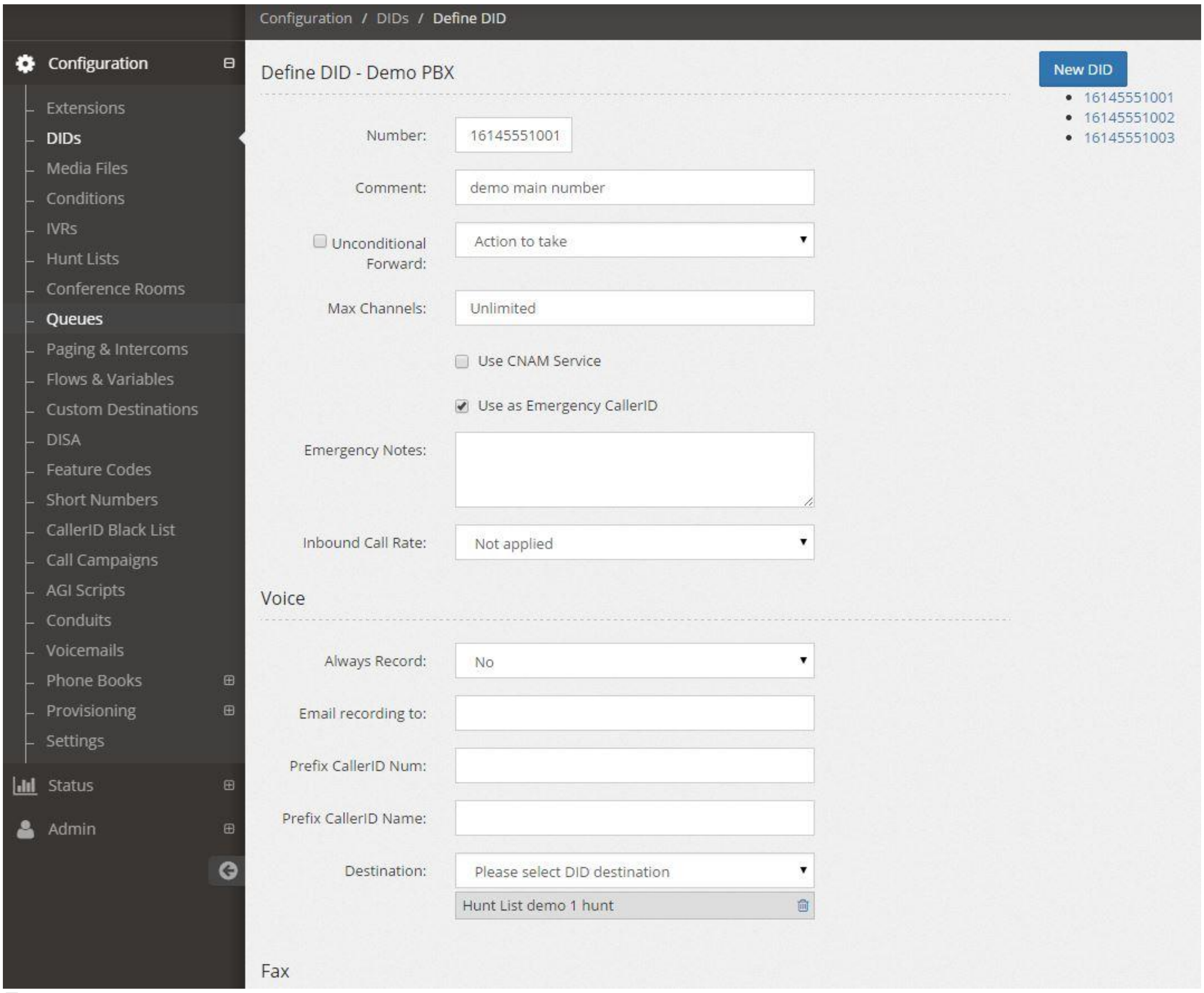
Every PBX can be assigned a group of DIDs, numbers to be dialed from the outside. Each DID can be configured to reach a certain extension or Queue, or Hunt Group, or IVR ,or any combination of the elements defined for the PBX.



New/Define DID

A DID can be configured to accept voice fax, or try to guess the calling party (if voice of fax). Auto detection relies on signaling at the start of the call and cannot be always reliable. The time to detect the fax tone can be set in the Admin/Settings page.

Comment is just a comment and is only used for you to “name” the DID in the displays.



Unconditional Forward is a destination that can be set on the DID to send the call to a particular destination. It is enabled/disabled by the checkbox. That checkbox can be easily controlled by a feature code. This is not the destination to be set for common usage; you need to use the one in the Voice section.

Max channels allows you to set the maximal number of channels available on the DID. Any additional call received will be refused with a busy signal.

Use CNAM service allows you to assign the Caller ID Name based on this popular service offered in most countries. You can configure the service in the Admin/Settings page.

Use as Emergency Caller ID permits you to mark this DID as one of the available to be used when an emergency call is dialed. Due to the fact that emergency numbers are not standard in all countries, it is your duty to assign the "emergency flag" to the dialing rules for emergency numbers. When a call is dialed and it is using a dialing rule marked as "emergency", the Caller ID number used for the extension is the one defined as "Emergency", among the ones with this flag set.

Inbound Call Rate allows you to define a call rate for inbound calls.

Voice section allows defining how to route the call when a voice call is received.

Always record forces the recording of the call and future calls. Recordings will be available through the Call History menu.

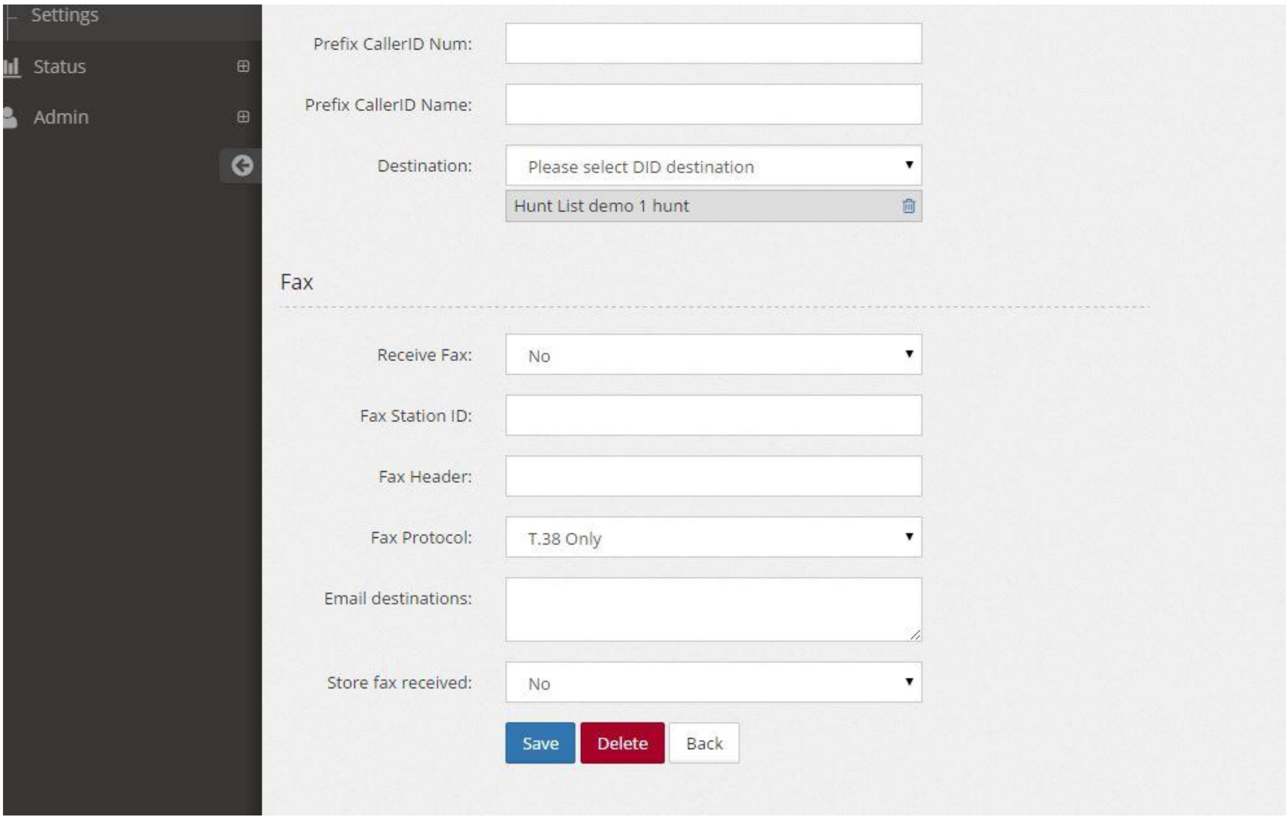
Email recordings to defines if and to who the recordings need to be sent once the call is over. Multiple destinations email can be entered with any delimiter, like space, comm and point and comma.

Using the **Prefix CallerIDNum** is possible to define a string to be added to any Caller ID number received to identify. For example, the DID the call is coming from. The same can be achieved by defining a Custom Destination.

Prefix CallerID Name is the same as above, but for the Caller ID Name.

Destination multiple drop down allows defining the list and the order of the objects receiving the call.

Fax section defines what to do with the call if a fax is received on the number defined. Receiving Fax over the Internet has often a low success rate, even if T.38 protocol is used.



Receive fax can be used to select if autodetect, force or disallow the reception of a Fax over the current DID. NOTE: Autodetect makes the PBX listen for one full ring cycle in order to determine if this is a fax or voice call. For dedicated fax numbers, always choose “force”.

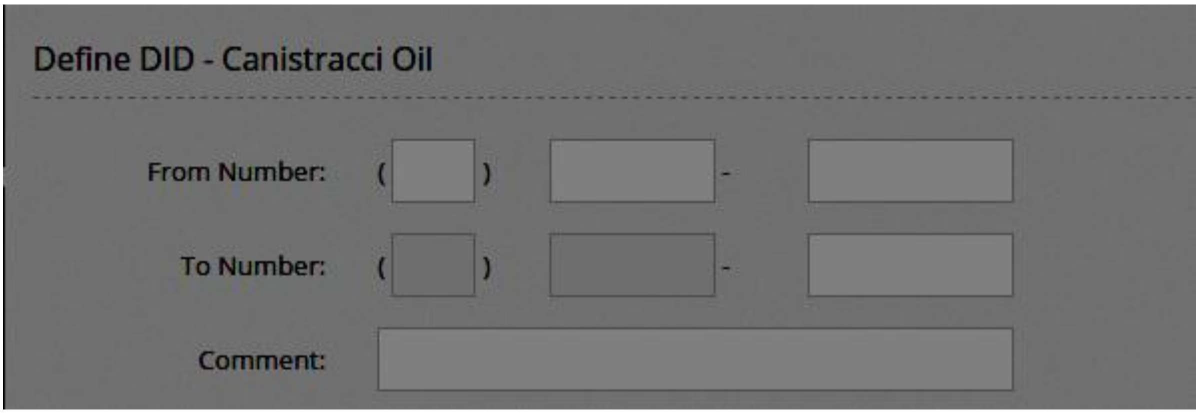
Fax Station ID and **Fax Header** allows you to customize your virtual fax.

Fax Protocol can be selected among T.38 (the preferred way to receive faxes), T.38 with fallback to G.711 or only G.711.

Email destinations box permits to enter one or multiple comma delimited email address to forward the received fax. Received faxes are sent in PDF format. Partially received faxes are sent in the same way.

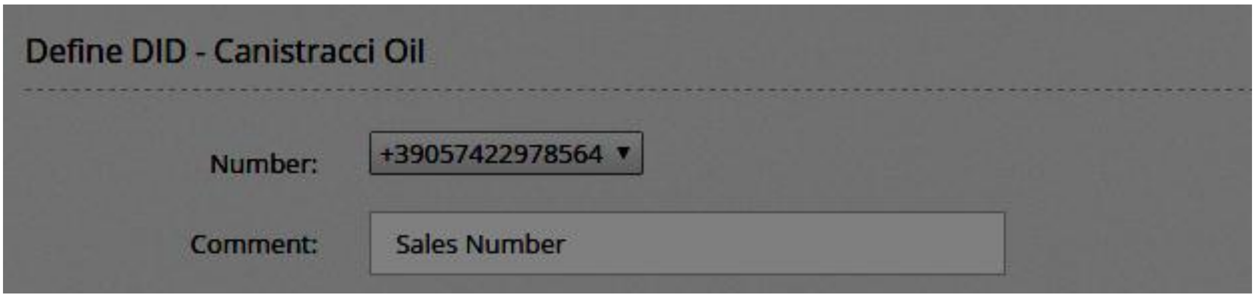
It is possible to store the fax received for later reviewing using the **Store Fax Received** control.

Bulk DIDs creation



It is possible to create multiple DIDs at once with the Bulk DIDs creation button. In this way, a range of numbers are requested. All the numbers generated are configured in the same way.

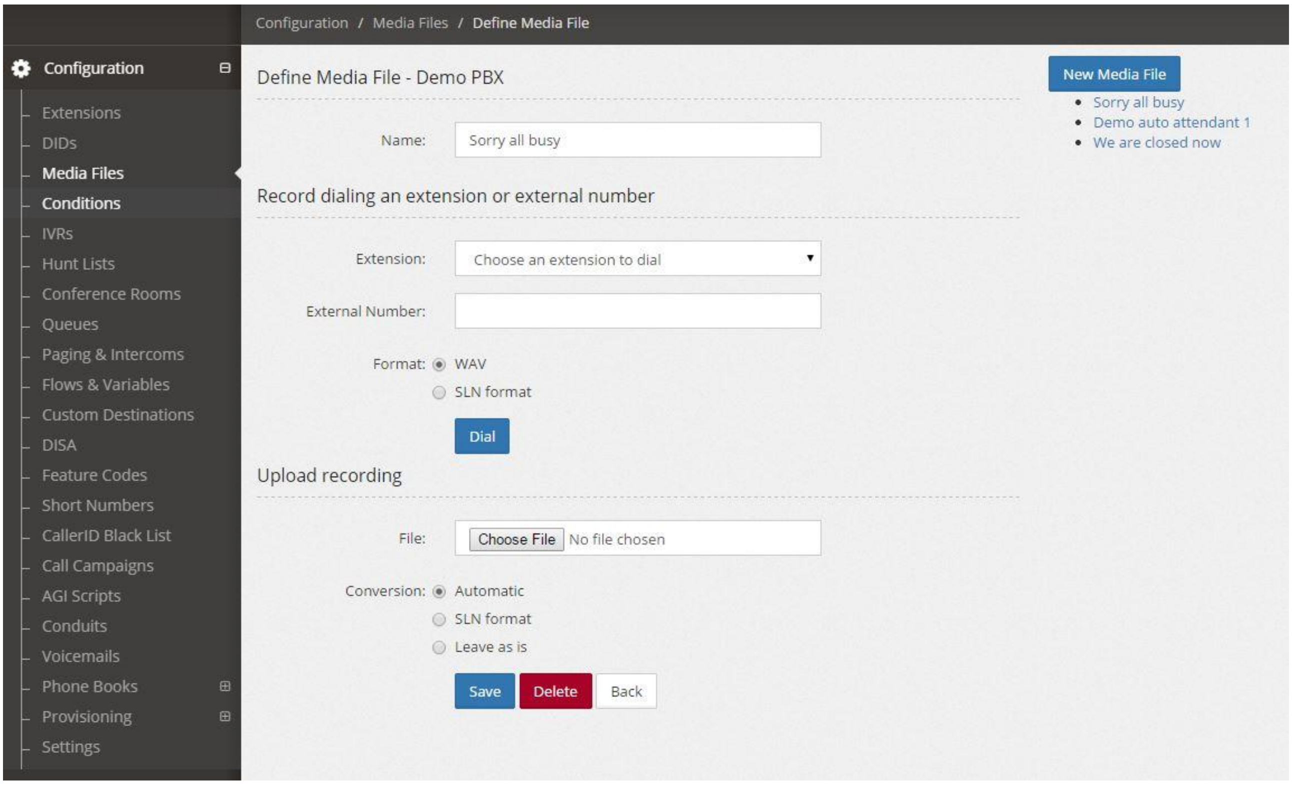
Use DIDs storage



In the Admin/Settings page it is possible to choose to use the DIDs storage. To do this, the DIDs need to be entered by the Admin/DIDs List menu and can be later chosen using a drop down box from the Define DID page. DIDs cannot be deleted, but they are just not assigned to any PBX.

Media Files

Media files are used for music on hold, welcome messages and for every kind of message played to the user.



Each media file is internally addressed by its MD5 sum and a local copy is stored on each node to reduce the load on the database server. A name can be assigned to the media file to easy identify in the system. To reduce the asterisk load when playing, it can be automatically converted into slin format (16 bit Signed Linear PCM).

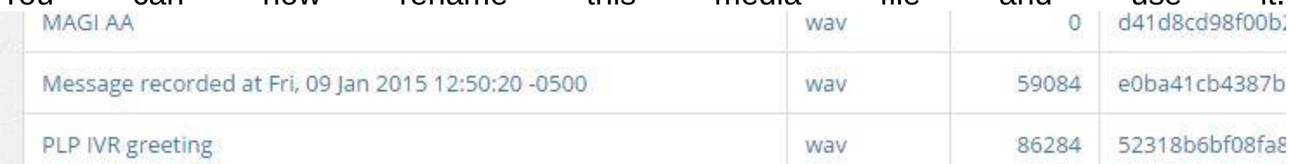
To create a new media file:

Enter a name for the recording – something like “Holiday message”

Either:

- Have the system call you at an extension (It will walk you through recording, confirm/rerecord/save the message).
- Have the system call you at an external number (It will walk you through recording, confirm/rerecord/save the message)
- Upload an existing WAV file.

One additional option is to add a feature code pointing to feature “Record a message”.
Example: You select configuration->feature codes, click new feature code and add *1234 pointing to feature “record a message”.
When any user dials *1234 the system will ask them to record a message.
This new recording will be in the media files list with name “Message recorded at …”
You can now rename this media file and use it.



MAGI AA	wav	0	d41d8cd98f00b
Message recorded at Fri, 09 Jan 2015 12:50:20 -0500	wav	59084	e0ba41cb4387b
PLP IVR greeting	wav	86284	52318b6bf08fa8

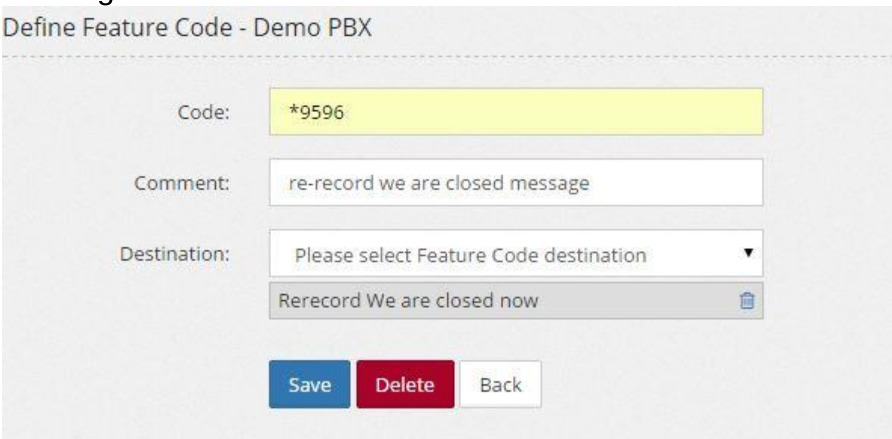
To re-record an existing media file you may:

Either:

- Have the system call you at an extension (It will walk you through recording, confirm/rerecord/save the message).
- Have the system call you at an external number (It will walk you through recording,

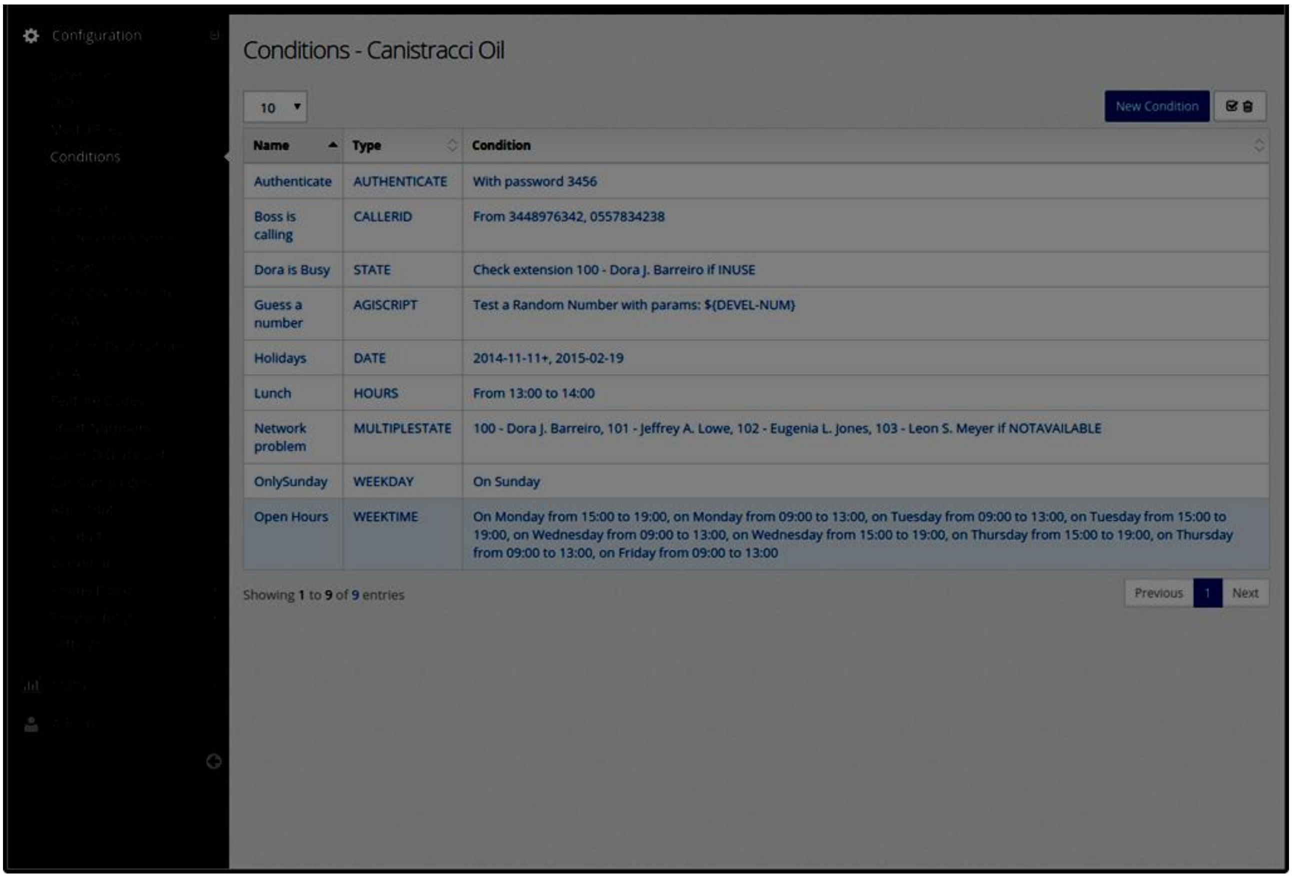
confirm/rerecord/save the message)

- Upload an existing WAV file.
- Or add another feature code pointing to feature “re-record media file…”. When a user dials this feature code, it will walk them through re-recording this specific message



Conditions

Conditions allow managing the call flow, playing for example, a different messages or rou based on hours, days, or calling party.



Several type of Condition can be configured:

Weektime a complete week planner allows to easily identify in which day/hour to trigger the destination.

Caller ID the routing decision is made using the Caller ID of the call. For example, allowing coworkers calling the main number to directly reach the support staff without waiting in the Queue.

Weekday allows routing calls based on the day of the week

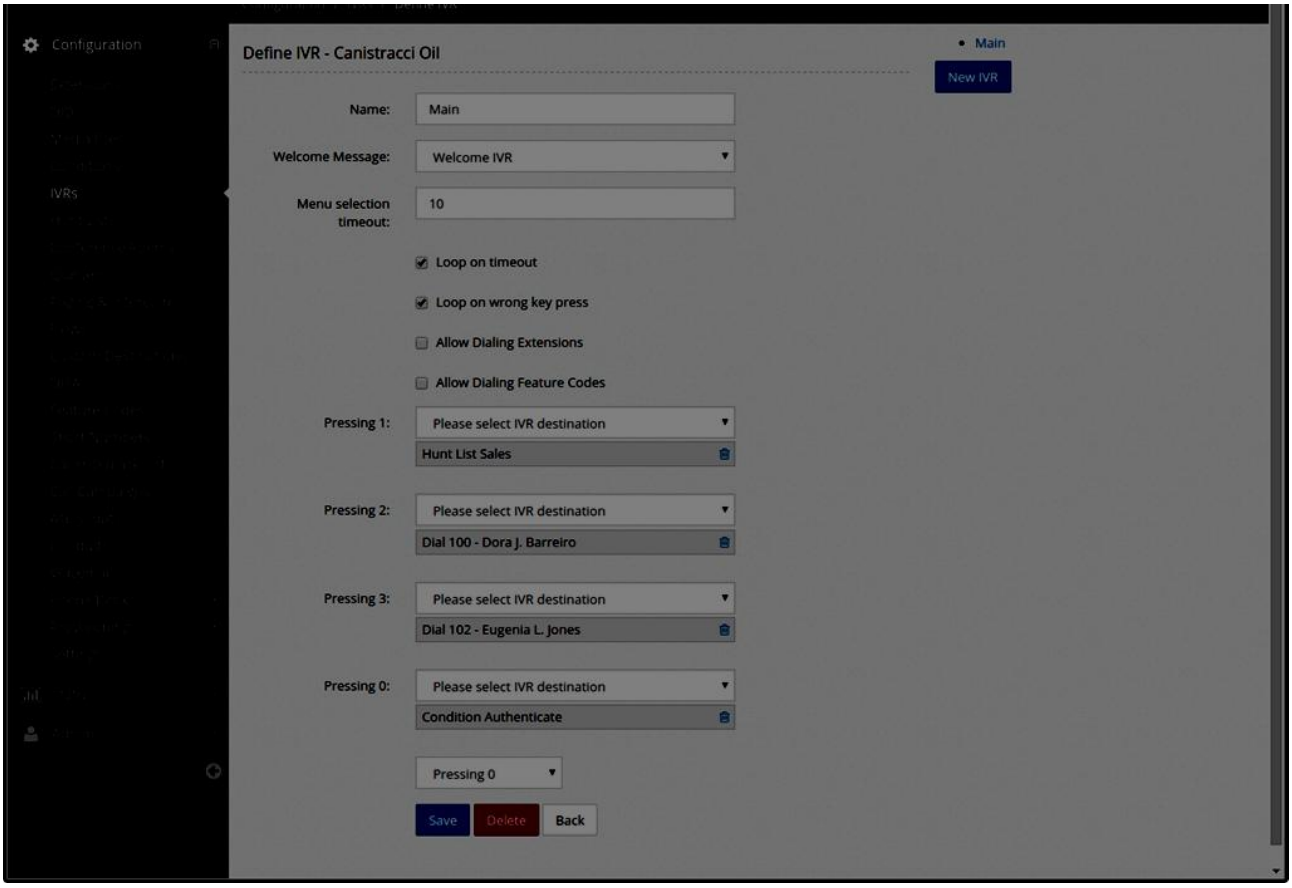
Date routes the calls based on specified dates. A date, like Christmas, can be made “recurring”, so it will trigger every day, regardless the year.

Extension Status the routing decision is made based on the status of an extension. This condition is really powerful when connected to the custom setting of extension status.

AGI Script execs an AGI script and check the variable AGIRESULT. If set to true, the condition is matched, otherwise the “not match” condition is followed.

IVR

IVR defines Interactive Voice Response to manage voice menus.



Welcome Message is the media file to play to the calling user while waiting for the user choice

Menu selection timeout is the time in seconds to wait for the user choice before going to the “On timeout” destination.

Digit timeout is used when the “Allow Dialing Extensions” or the “Allow Dialing Features Code” is selected. This determines the amount of seconds to wait before considering the number entered as “complete”

Loop on timeout allows the continuation of playing the welcome message, and the wait for the selection every time the Menu selection timeout expires.

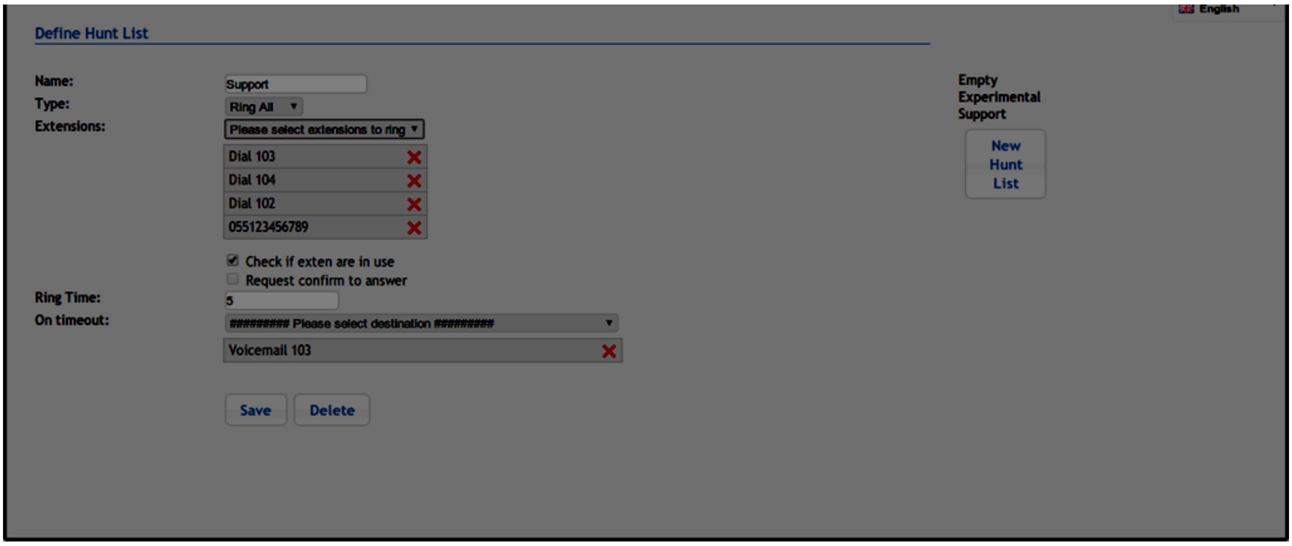
Loop on wrong key press allows the restart of playing the welcome message and the wait for the selection, if the user chooses an unsupported key.

Allow Dialing Extensions permits the calling user to dial directly to an extension instead of picking one of the digits.

Allow Dialing Feature Codes, like above, but for feature codes.

Hunt List

Hunt List defines a list of extensions or external numbers to dial at all once or in sequence.



The **Type** of the Hunt List can be:

Ring All – All extensions and external numbers can be dialed all at once

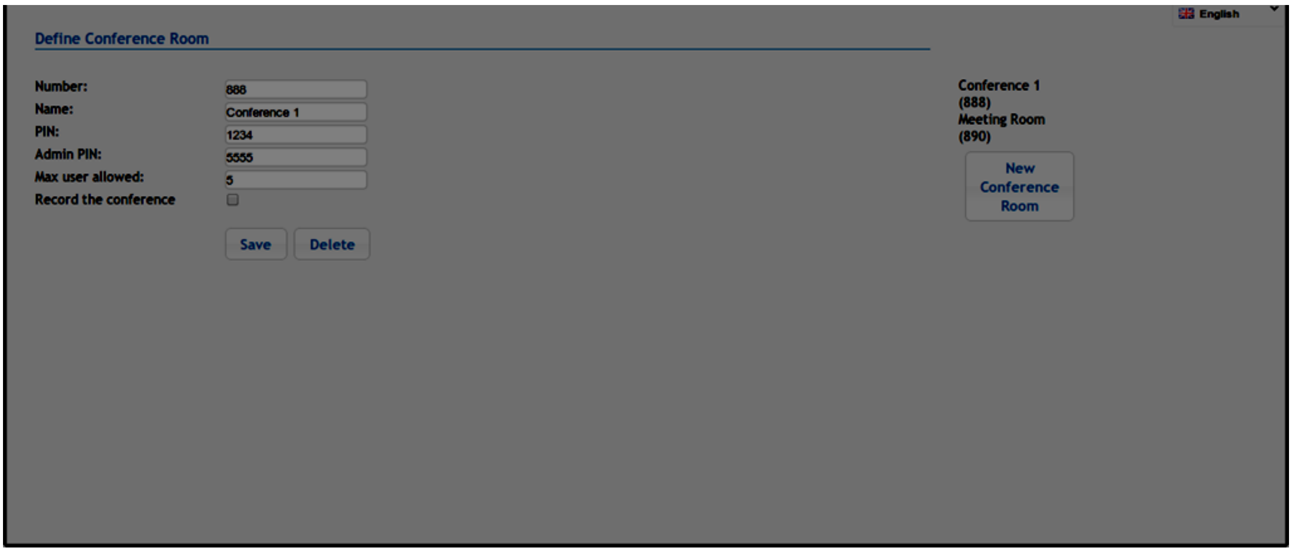
Cycle – Extension and external numbers are dialed in the order specified and once the bottom of the list is reached, the “hunt” starts over.

Sequence – Extensions and external numbers are dialed in the order specified. When the bottom of the list is reached, the destination specified in the “**On timeout**” is followed

Check if exten are in use allows user to skip dialing extension already in use

Request confirm to answer if set, requests the dialed user to accept or reject the call. If the call is rejected, the hunt list continues trying to locate a phone

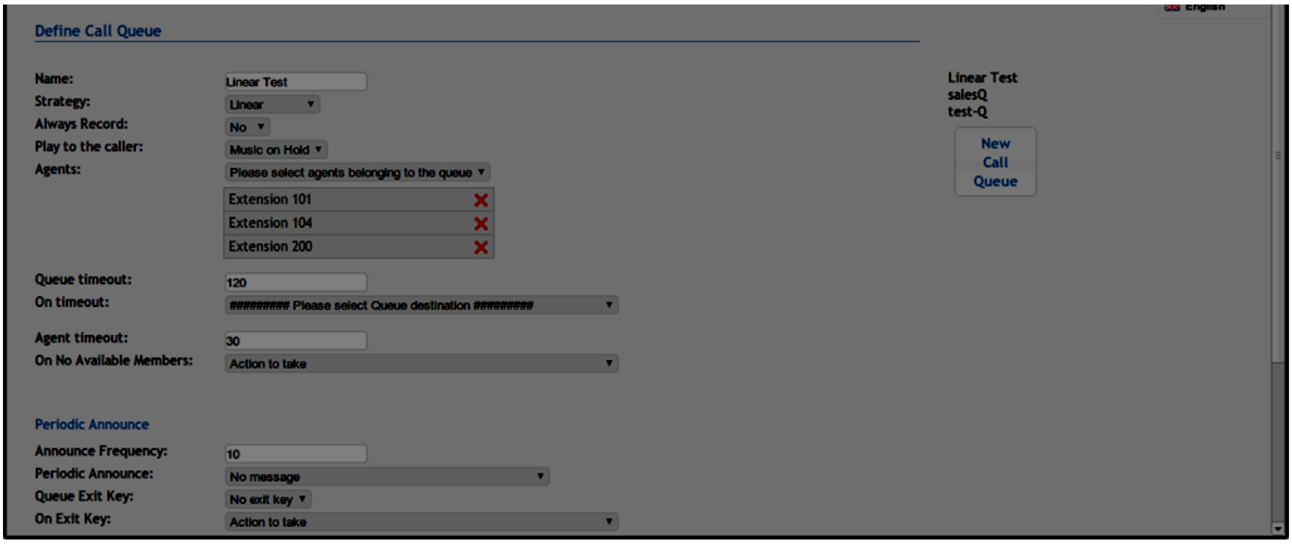
Ring Time allows specifying the time each extension or external number has to be dialed before skipping to the next item.



Conference Rooms

A conference room or meeting room is a virtual place where all phones dialing, are joined in a single conversation. Conference rooms can be protected by a PIN. A special Admin PIN is reserved to the administrator, so he can mute/unmute participants. The maximal number of users allowed in the conference can be set.

The conference can be recorded. The audio file is available through the Call History.



Call Queue

Call queue permits to hold all the incoming calls in a queue and distribute the calls to the agents available.

Strategy can be one of the following:

Ring All – All agents are rang at the same time

Round Robin – agents are rang, one at the time, in a round robin way

Random – agents are rang in a random order

Least Recent – The least recent agent is rang

Fewest Calls – The agent with the fewest calls is rang

Linear – agents are rang in the order specified. Due to some asterisk limitation, it is not possible to change the Strategy of an already defined Queue to “Linear”. The queue needs to be destroyed and recreated.

Always Record always allows recording the call. The call record is available through the Call History.

Play to the caller permits to define if playing the defined Music on Hold for the PBX or a generic Ringing

Agents list the agents in the queue. For each extension, two kinds of agents are available, the first is normal. The second is the “following to A.D.”; allowing the forwarding of the call to the Additional Destination defined for the extension.

Queue timeout is the amount of time the caller can be hold in the queue before being sent to the destination specified in the “**On timeout**”

Agent Timeout is the amount of time an agent is rung before moving on the next agent.

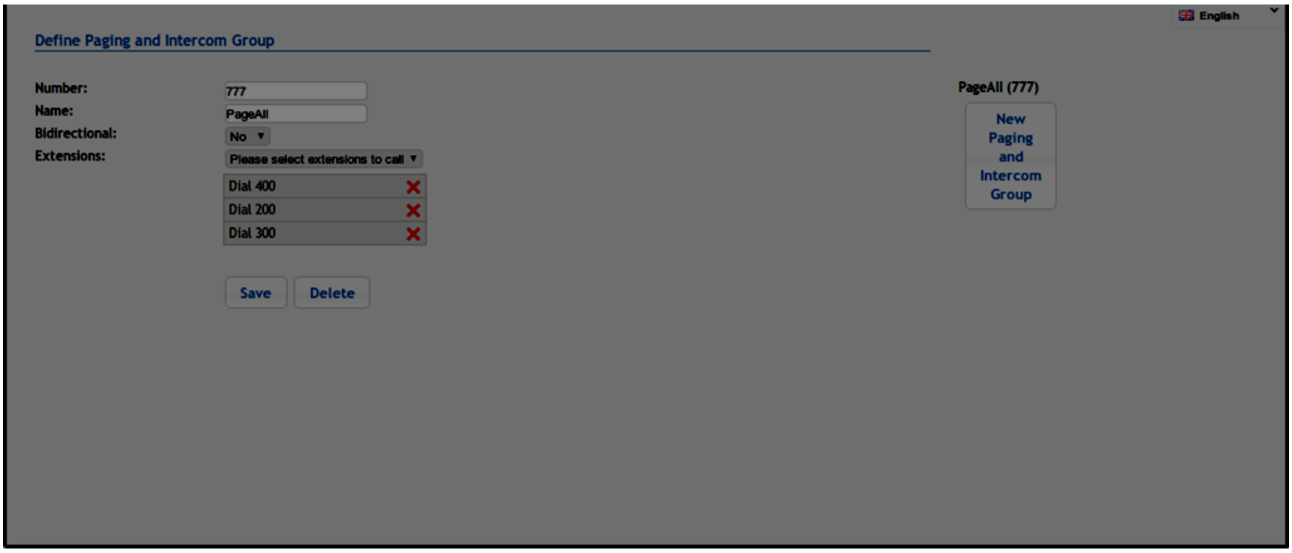
A special action can be configured when **No Available Members** are present in the queue.

Periodic Announce section allows definition of the announcment to play to the callers in the queue with the **Announce Frequency**. The **Periodic Announce** can be chosen between the media file loaded.

The user can exit the Queue by pressing the “**Queue Exit Key**” and he will be directed to the destinations chosen by the “**On Exit Key**”. A special destination can be selected, named “Exit the Queue and call back when it is your turn” allowing the caller to hang up the call and be called back when he is in front of the queue, ready to be served.

The position of the user can be played with the “**Announce Pos. Frequency**”, if different than zero.

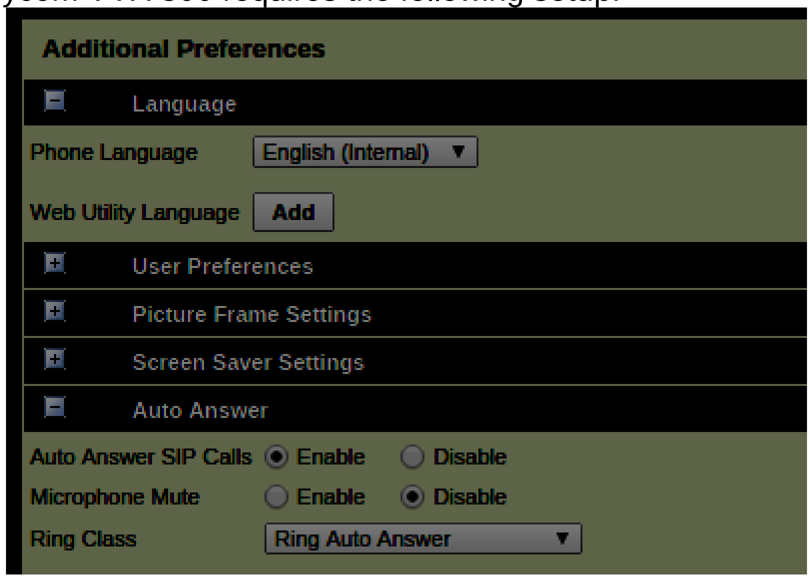
It is possible to redefine the messages usually played to the user choosing them between the media file uploaded



Paging and Intercom

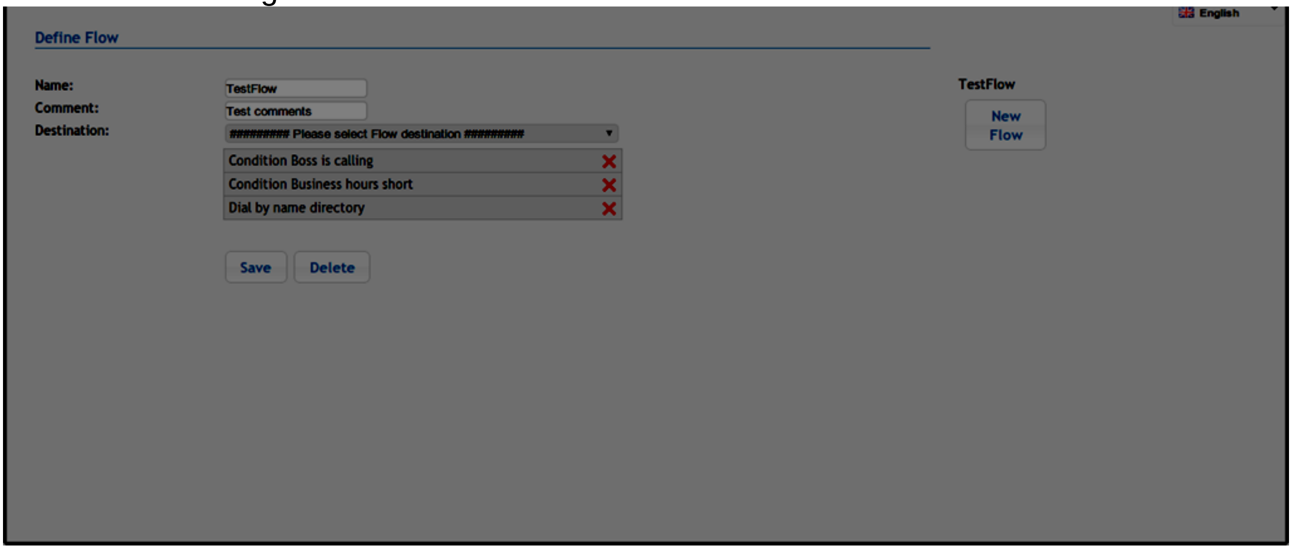
Almost all SIP phones allow paging by using them as an intercom; the ability to establish a mono directional or bidirectional communication without making them ring.
The **number** defined can be called directly by all extensions and it can be chosen to make a **bidirectional call** (intercom) or just use the service as paging device.

Some phones require adjustment to the configuration to allow them to auto answer. For example, the Polycom VVX 300 requires the following setup:

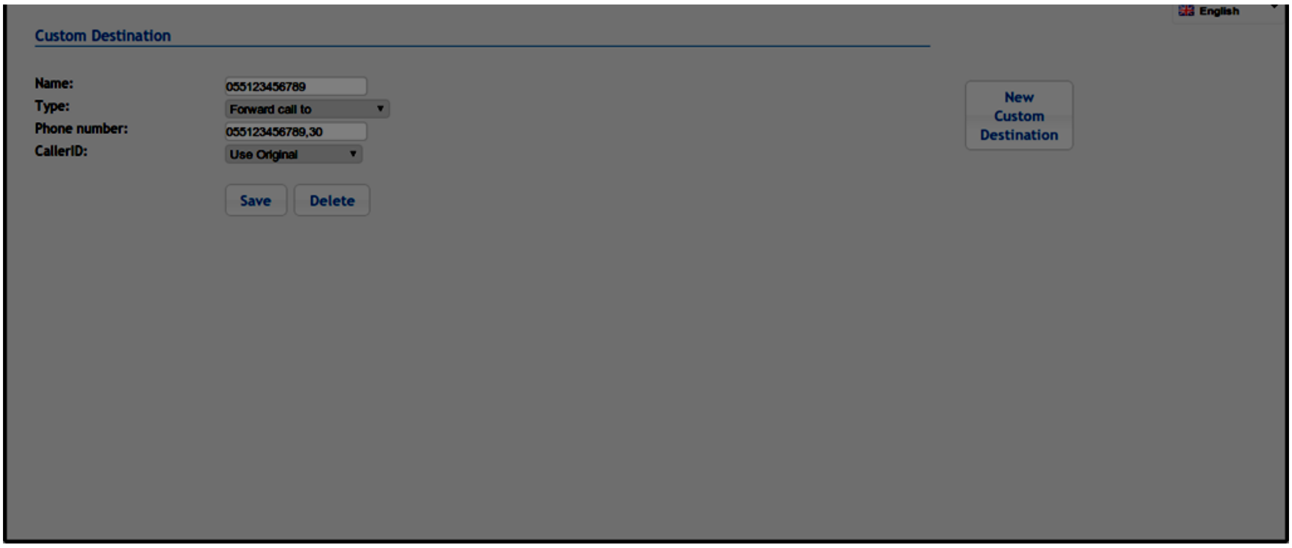


Flow

A flow is a predefined set of destinations that can be used in other Configuration directives. It is like a Macro, allowing to use the same steps of destinations in several places and maintained in a single location.



Custom Destinations



Custom Destinations allow the definition of custom destinations to be used in other Configuration settings.

There are several types of custom destinations:

Forward call to – will forward the call to an outbound number, using a **timeout** for the dialing. The dialing timeout has to be entered separated by a comma. The **CallerID** can be chosen among the usual Caller ID available and use the Original Caller ID of the call received.

Alter Caller ID to – will change the Caller ID to a custom one. Standard Asterisk variables, like ${CALLERID(num)} can be used.

Alter Caller ID Name to – will permit to change the Caller ID Name to a custom one. Standard Asterisk variables, like ${CALLERID(name)} can be used.

Custom Dial() with param – will use the **Dial Command** as argument to a generic Dial command

Use Feature Code – Like to "Forward call to", but a feature code can be used

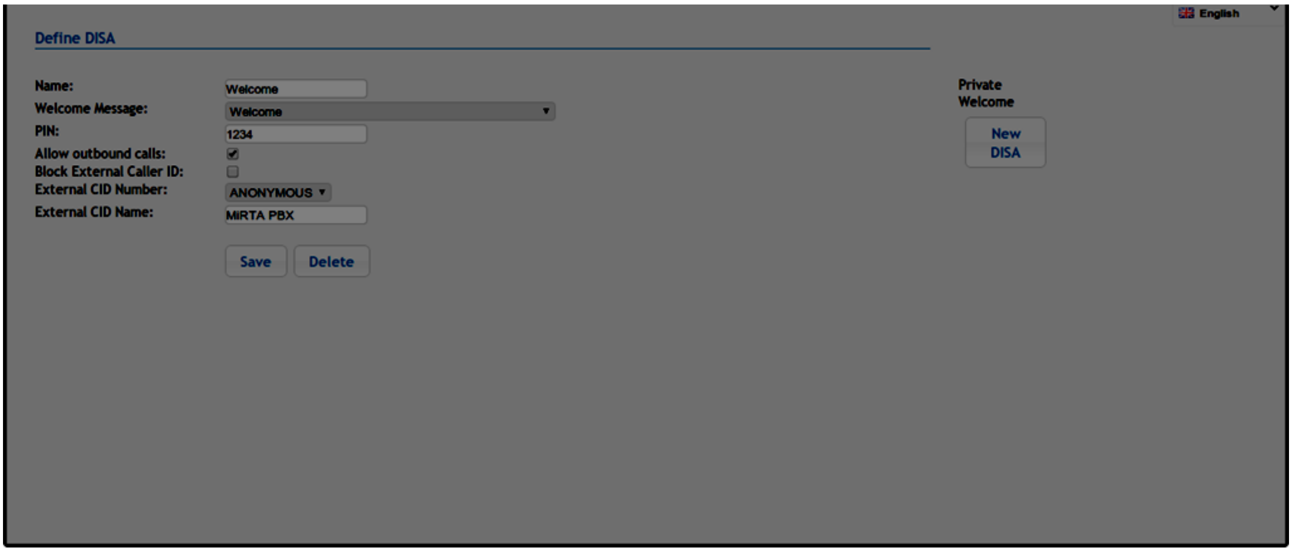
Set Extension to not in use: Set the extension state to "not in use"

Set Extensions to in use: Set the extension state to "in use"

Toggle Extension state: Change the extension state from “in use” to “not in use” or vice versa.

DISA

DISA stands for Direct Inward System Access and is a way to let inbound callers reach any internal extension. Once answered, the system will play a message inviting the caller to enter the extension number to be connected to. The system also allows dialing outbound numbers, usually protected by a PIN code.
The **Welcome message** is played to the calling party, usually asking to enter the extension to dial. If present, the **PIN** is requested to access this feature. It is highly advisable to set a PIN when outbound calls are allowed.



Feature Codes

Feature Codes are the key to access any aspect of the PBX, trigger special features or just performs tricky operations. Feature codes can start with the * (star) or with the # (sharp) and can be associated to a number of action from the following list. When requested, the special string [NUM] and [EXT] can be used in the feature code. The [NUM] will be replaced with the number dialed. For example, if a feature code is set to *62[NUM] for “Mask the Caller ID on calling [NUM]” then if the number *625558764 is dialed, and then the [NUM] is assigned to the number 5558764. The [EXT] instead will be replaced with an extension number. If a feature code is set to *8[EXT] to Pickup Extension [EXT] and you have defined extension 100, then dialing *8100, the extension 100 will be picked up.
Feature code *1 is reserved (due to asterisk limitation) to enable/disable recordings.

List standard feature codes:

Feature Code	Description
Answer the call	Usually not needed, it just answer the call
Barge with extension [EXT]	Barge with extension [EXT]
Dial by name directory	Access to Dial by name directory menu
Dial by name using the [NUM] dialed	Use the dialed [NUM] to call using dial by name

Feature Code	Description
Disable FMFM extension	Disable FMFM for the calling extension
Disable on busy forwarding for calling extension	Disable on busy forwarding for calling extension
Disable on no answer forwarding for calling extension	Disable on no answer forwarding for calling extension
Disable on offline forwarding for calling extension	Disable on offline forwarding for calling extension
Disable unconditional forwarding for calling extension	Disable unconditional forwarding for calling extension
Echo test	Perform an echo test, repeating all what is said
Enable FMFM	Enable FMFM for the calling extension
Enable on busy forwarding for calling extension	Enable on busy forwarding for calling extension
Enable on no answer forwarding for calling extension	Enable on no answer forwarding for calling extension
Enable on offline forwarding for calling extension	Enable on offline forwarding for calling extension
Enable unconditional forwarding for calling extension	Enable unconditional forwarding for calling extension
Force Recording of the call	Activate the recording for the call
Hangup the call	Hangup the call
Intercom with extension [EXT] (two way audio)	Perform an intercom (two way audio) with the extension dialed [EXT]
Login to all Queues	Login the calling extension to all queue
Logout from all Queues	Logout the calling extension to all queue
Mask the callerID on calling [NUM]	Activate the Privacy Mode while calling the [NUM]
Page extension [EXT] (one way audio)	Perform a page (on way audio) with the extension dialed [EXT]
Park the call	Park the call
Pickup Extension [EXT]	Pickup extension dialed [EXT]
Pickup Group	Pickup a call from the current group
Play Beep	Play a simple beep
Play the callerid of the calling party	Play the callerid of the calling party
Record a Message	Record a message. The message is recorded and added to the list of media files with the date and time of the recordings. Usually it is renamed and used in some menu
Retrieve the voicemail of the calling	Retrieve the voicemail of the calling

Feature Code	Description
extension	extension
Retrieve the voicemail of the [EXT] dialed	Retrieve the voicemail of the extension [EXT]
Say the parked calls extensions	List the parking lot used by calls parked
Set Extension [EXT] state to in use	Set the state of the extension [EXT] to “IN USE”
Set Extension [EXT] state to not in use	Set the state of the extension [EXT] to “NOT IN USE”
Set FMFM number to [NUM] and enable it	Set the number [NUM] dialed as FMFM for the calling extension.
Set on busy forwarding for calling extension to [NUM]	If the [NUM] dialed is not an internal extension, then the [NUM] is checked among the Custom Destinations. If one of the Custom Destinations is matching the [NUM], then it is used, otherwise a new Custom Destination is automatically created and set as Busy Forwarding.
Set on no answer forwarding for calling extension to [NUM]	If the [NUM] dialed is not an internal extension, then the [NUM] is checked among the Custom Destinations. If one of the Custom Destinations is matching the [NUM], then it is used, otherwise a new Custom Destination is automatically created and set as No Answer Forwarding.
Set on offline forwarding for calling extension to [NUM]	If the [NUM] dialed is not an internal extension, then the [NUM] is checked among the Custom Destinations. If one of the Custom Destinations is matching the [NUM], then it is used, otherwise a new Custom Destination is automatically created and set as Offline Forwarding.
Set unconditional forwarding for calling extension to [NUM]	If the [NUM] dialed is not an internal extension, then the [NUM] is checked among the Custom Destinations. If one of the Custom Destinations is matching the [NUM], then it is used, otherwise a new Custom Destination is automatically created and set as Unconditional Forwarding.
Spy extension [EXT]	Spy on extension [EXT]
Toggle state of extension [EXT]	Change the state of the Extension [EXT] from IN USE to NOT IN USE and vice versa
Unmask the callerID on calling [NUM]	Remove the Privacy Mode while calling the [NUM]
Voicemail for Extension [EXT]	Retrieve the voicemail for extension [EXT]
Wait five seconds	Wait five seconds

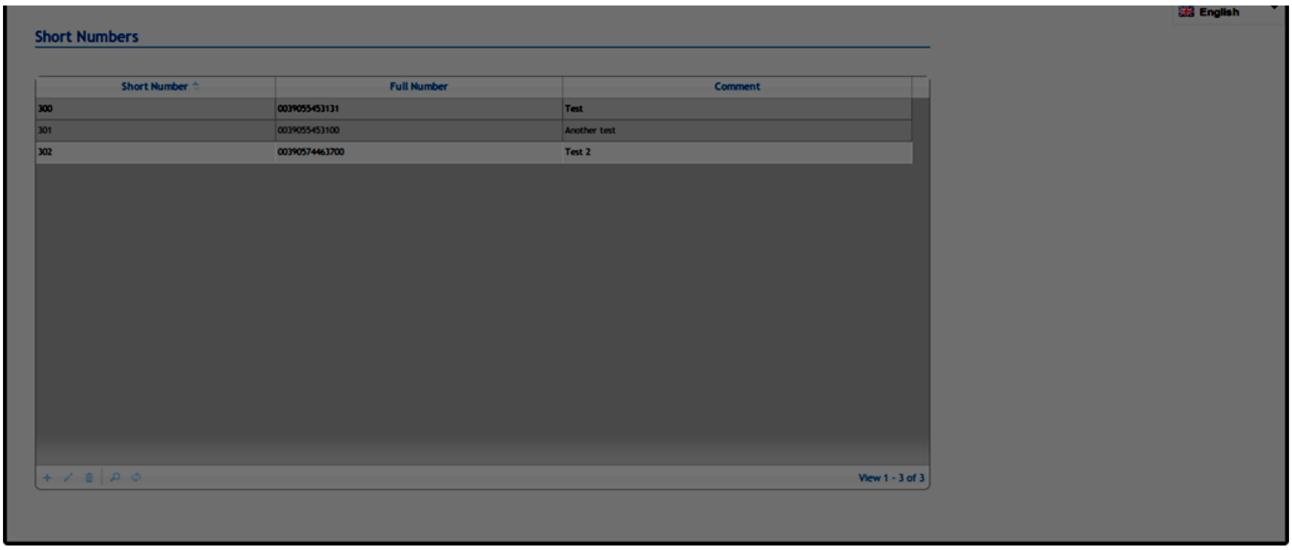
Feature Code	Description
Wait one second	Wait one second
Wakeup Alarm - Disable the time set	List all wake up alarms set for the calling extension and allow to delete one of them
Wakeup Alarm - Play the time set	List all wake up alarms set for the calling extension
Wakeup Alarm - Set the time from DTMF HHMM	Set a wake up alarm by requesting to enter the hour in the HHMM format

New Feature Codes are automatically created when any of the objects are created, as follows:

Object	Feature Code
Extensions	Allowing to dial the extension listed
Custom Destinations	Allowing to dial the destination defined
Media Files	Rerecord, Playback and Background the media file
Conditions	Check condition
IVRs	Execute IVR
Hunt Lists	Execute hunt lists
Conference Rooms	Access conference room
Queues	Access queue
Flows	Execute flows
DISA	Access DISA
Voicemails	Send the call to the voicemail
Whisper to extension [EXT]	Whisper to extension [EXT]

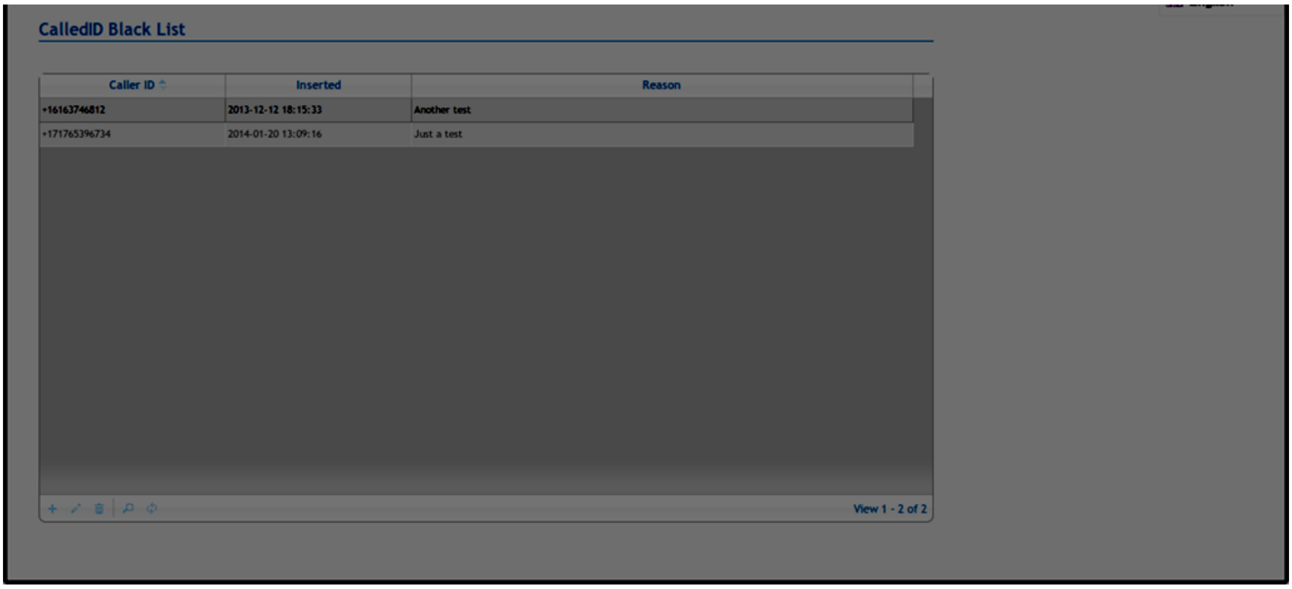
Short Numbers

Short numbers are a way to assign shortcut for dialing numbers.

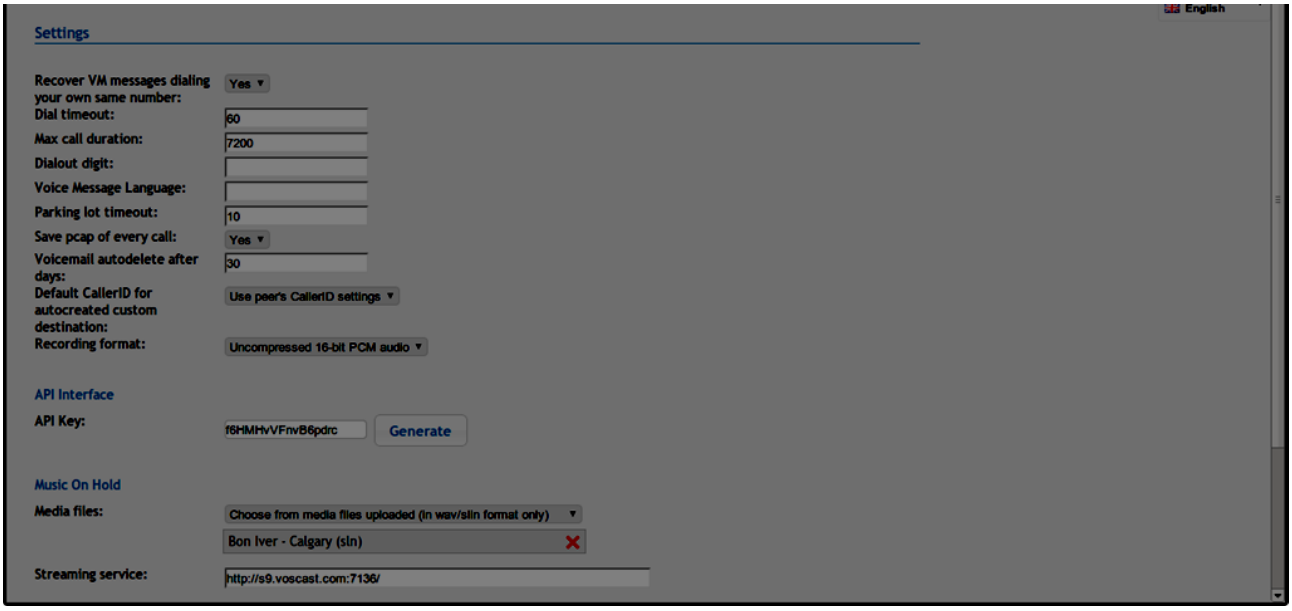


CallerID Black List

It is possible to avoid receiving calls from a list of caller IDs by entering them in this list. Calls from those numbers will be hang up directly.



Settings



Every PBX can have its own settings.

The following aspect can be customized:

Recover VM messages dialing your own same number. Dialing your own same number is not really useful, so why don't you assign this action to recover the VM messages?

Dial timeout identify the standard time the dialing command will ring an extension or an external number before reporting as “no answer”. Remember the dialing time can be adjusted for each extension.

Max call duration sets the maximal time, in second, a call can last connected.

Dialout digit allows you to define a digit to use for dialing out. If not set, all not local numbers are dialed outbound

Voicemail Message Language let's you specify the default language to use in the Voicemail. This value can be set for each voicemail.

Parking log timeout specifies how long in seconds a call can stay parked before returning to the parking extension.

Save pcap every call let's you capture the pcap of every call received by the PBX for debug purpose. Call logs are saved in the database for easy recovery. The tool used for capturing the traffic is Voipmonitor

Voicemail autodelete after days let's you decide how many days of Voicemail to keep online. Deleted messages cannot be recovered.

Default CallerID for autocreated custom destinations. This is the Caller ID to use when the system needs to generate a Custom Destination, for example, when an unconditional transfer is set using a feature code.

Recording format. The recording format for monitoring calls can be chosen between Uncompressed 16-bit PCM Audio in Wav container and MS GSM audio, still in Wav container.

API Key. The API Key is used for the proxyapi.php API script.

Music On Hold allows user to choose the files to use as Music On Hold for the given PBX. If no files are chosen, the standard asterisk Music On Hold files are used. The files need to be in wav or slin format. Alternatively, a binary source of music can be selected, like a public streaming service entering the URL in the “**Streaming Service**” field, like http://s9.voscast.com:7136/

Web calls are calls generated using an HTTP request. A simple proof of concept is supplied with the webcall.php script. The generation of calls can be restricted using a password or by IP. The webcall.php script can be run either from the command line or by invoking using GET or POST method.

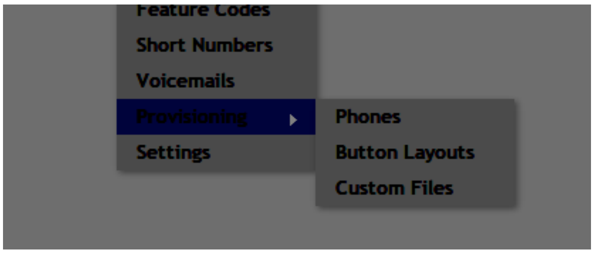
When run in CLI mode, the arguments are in order: source number, destination number, PBX code and secret key.

When run in GET/POST mode, the following variables needs to be assigned: source, dest, PBX, secret.

For example, getting the URL:

https://demo.plphonepbx.com/plphonepbx/webcall.php?source=104&dest=102&PBX=DEVEL&secret=H63JpSdPEWequMpr

Will make extension 104 to ring, once answered, extension 102 is dialed.

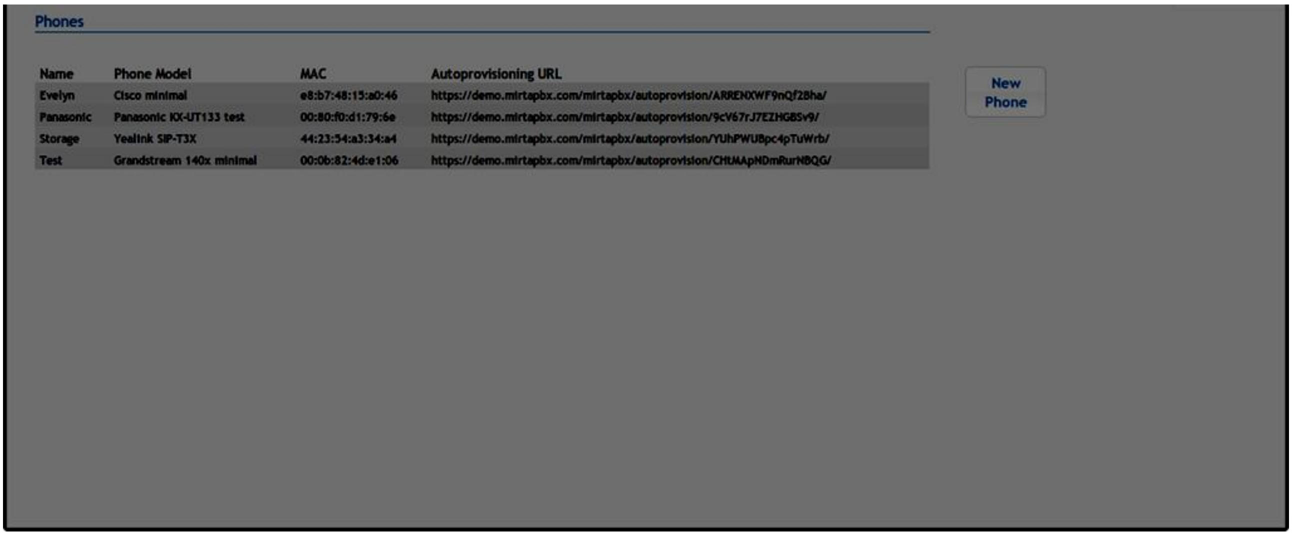


Provisioning

Provisioning is the action of configuring a phone automatically, by providing only basic information. PIphone PBX supports a wide range of phone provisioning with a general file format. New phones brand and model can be added using the Admin/Provisioning menu.

Phones

Files used for provisioning are usually text or XML file containing information like the user and password and the IP address or hostname of the SIP server. Information contained in the provisioning files need to keep confidential and the leakage of this information can lead to unauthorized usage of voice traffic. To avoid any snooping on provisioning content, usage of HTTPS is recommended. Be aware some phones require a valid SSL certificate to provision using HTTPS. The correct provisioning file is requested based on the MAC address of the phones. To avoid any brute forcing of the MAC address, a special password is needed in the URL to recover the file. Based on the model of the phone, a special string needs to be append to the end of the provisioning URL, like {mac}.cfg for Panasonic phones or {MA}.xml on Cisco phones. Check your phone manual for the right way to identify the MAC address in the provisioning URL.



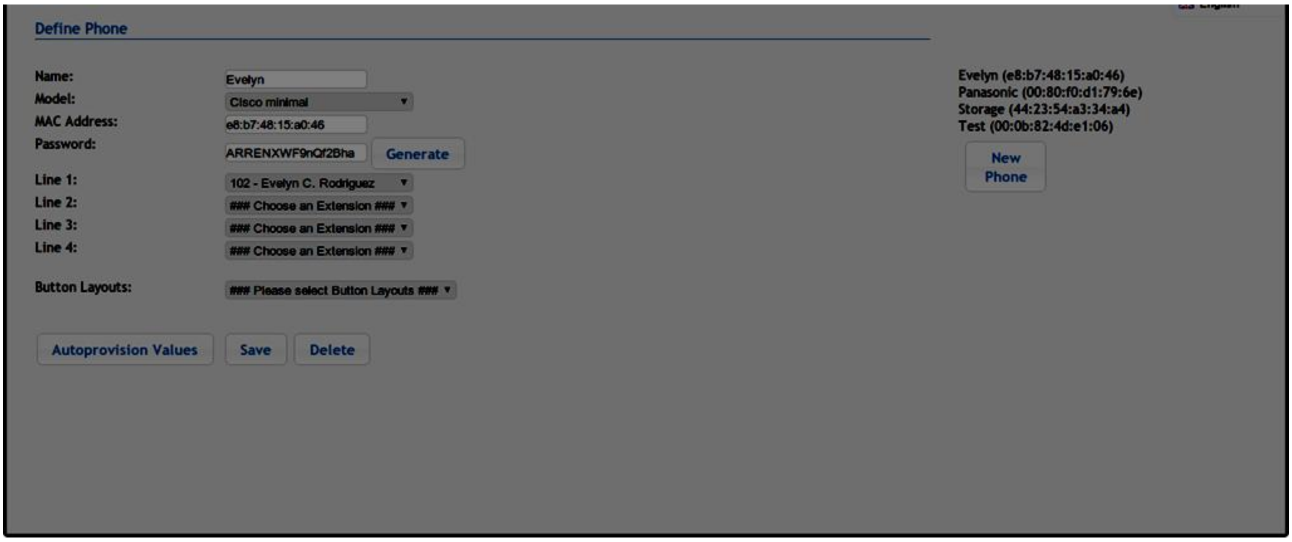
New Phone

Each phone can be named. The phone **name** is just used as reference and is not used anywhere else.

Model permits to select the provisioning template to use. Provisioning templates can be created using the Admin/Provisioning menu.

Mac Address is the key identifying the phone needing to be provisioned.

Password is normally set to your PBX CODE. This is NOT the authentication password of the extension, or a user password. It is a password added to the MAC address in order to

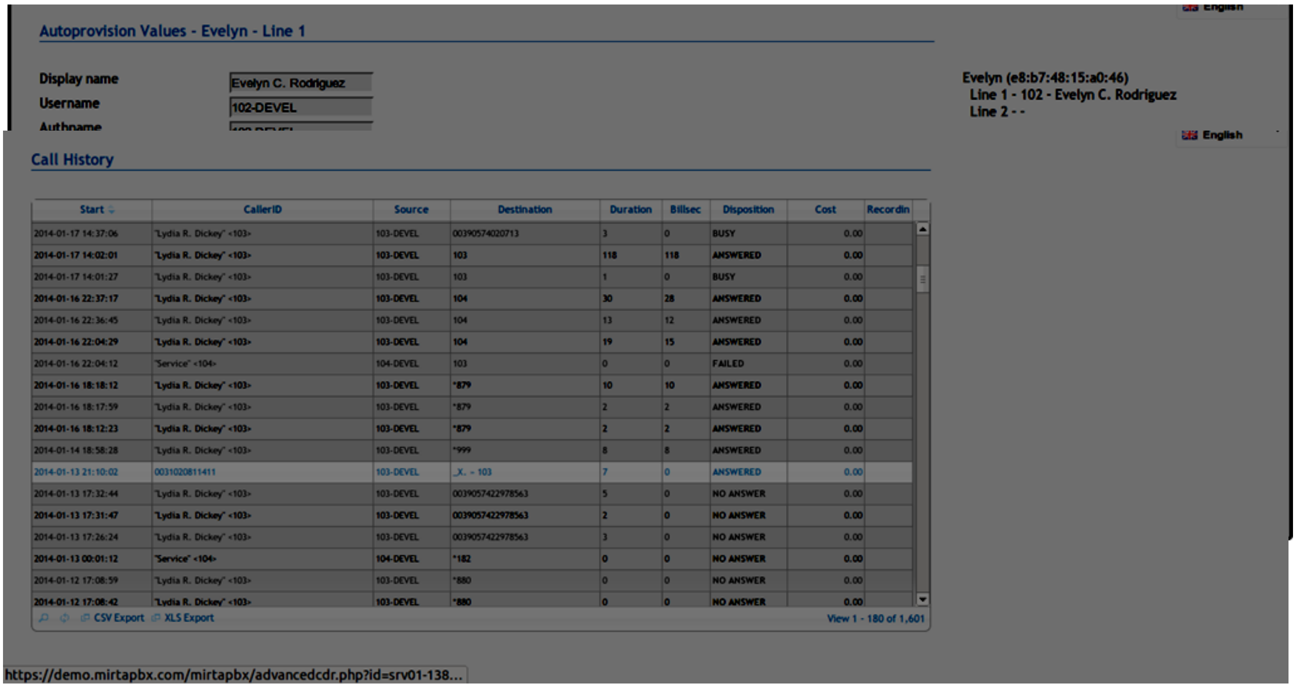


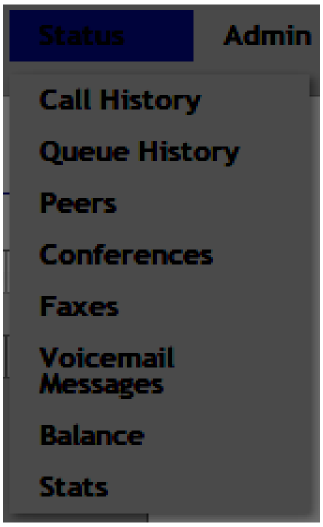
prevent brute force attacks.
Based on the definition of the phone model, one or multiple **lines** can be present, allowing selecting one or multiple accounts from the Extensions defined for the PBX.
One or multiple **button layouts** can be defined and assigned to the phone.

Autoprovision Values

When configuring the provision for a phone, you make a connection between the extensions in the pbx and the lines in the phone. Unfortunately it is not enough. The phone requests additional configuration you can enter by using the **Autoprovision Values** button.
Two kinds of autoprovision values can be needed:
The general one can be configured by clicking on the name and MAC address of the phone on the right and will be used to configure values defined in the template file applying on the whole phone.

The line related one can be configured by clicking on the Line desired and usually are already prefilled with values read from the extension configuration. Other values, like the server name and SIP port needs to be manually filled. Please refer to the Admin/Provisioning section on how to define predefined values and build custom templates.



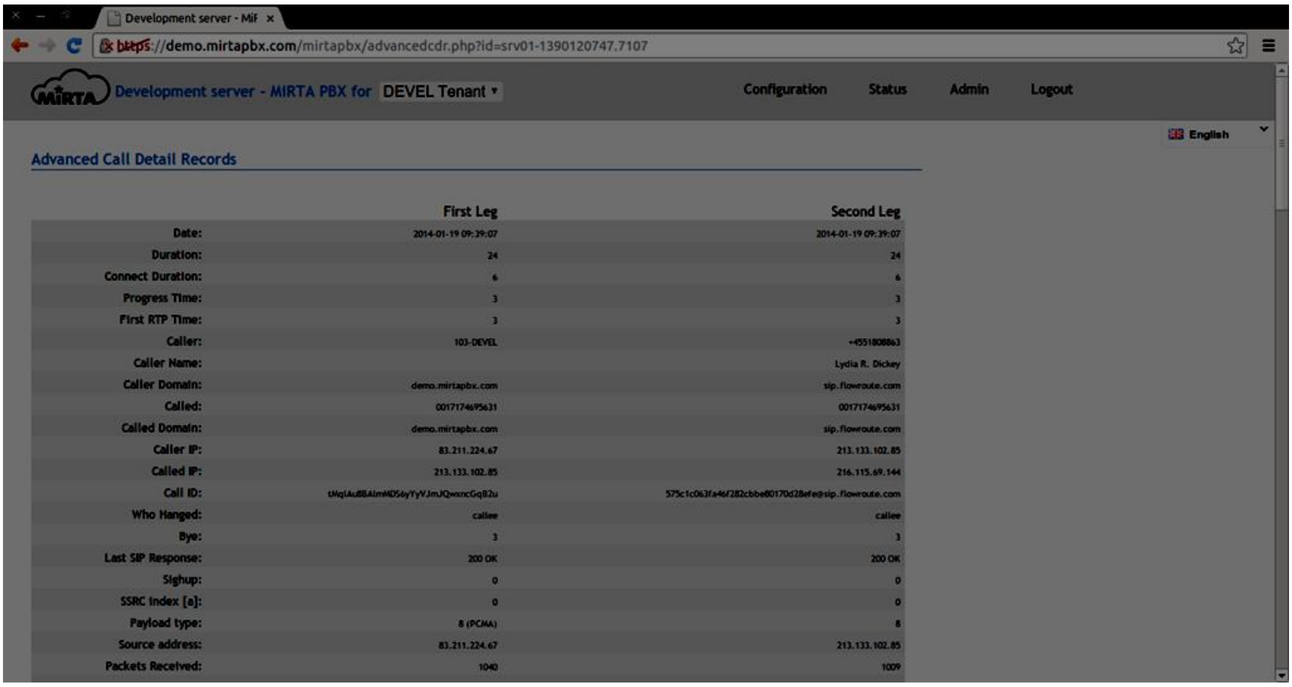


Status

Using the Status menu, a complete overview of the status of the system can be obtained. As other parts of the system, status is realtime. The info reported by the Status menu are related to the selected PBX and are available to the not-admin profile.

Call History

The info for every call, internal or external, inbound or outbound, is saved in the Call History along with basic information regarding the source and the destination.
For every call the info regarding the start time, the caller ID, the extension source of the call (if call is originated by the internal network), destination, duration, answered duration (billsec), disposition, cost and if any recording is available for the call.
Call History is retained for the amount of days configured in the Admin/Settings menu.
The disposition of the call is a link to access additional information gathered by the VoIPMonitor system.



Info like the progress time, who hang up, the codec used in the call, the IP of the caller and the called are easily shown a few seconds after the call has ended. A basic MOS calculator is included.

The packet jitter and the latency can be easily shown in the graphs following the raw data above.

A basic decoding of the call SIP message is available under the graphs.

First Leg SIP decoded Full PCAP dump

```
   1  0.000000 83.211.224.67 -> 213.133.102.85 SIP/SDP Request: INVITE sip:0017174695631@demo.mirtapbx.com, with session description
   2  0.000986 213.133.102.85 -> 83.211.224.67 SIP Status: 401 Unauthorized
   3  0.090018 83.211.224.67 -> 213.133.102.85 SIP Request: ACK sip:0017174695631@demo.mirtapbx.com
   4  0.110133 83.211.224.67 -> 213.133.102.85 SIP/SDP Request: INVITE sip:0017174695631@demo.mirtapbx.com, with session description
   5  0.113996 213.133.102.85 -> 83.211.224.67 SIP Status: 100 Trying
   6  2.939414 213.133.102.85 -> 83.211.224.67 SIP Status: 180 Ringing
   7  2.939876 213.133.102.85 -> 83.211.224.67 SIP/SDP Status: 183 Session Progress, with session description
1444  17.642834 213.133.102.85 -> 83.211.224.67 SIP/SDP Status: 200 OK, with session description
1466  17.863373 83.211.224.67 -> 213.133.102.85 SIP Request: ACK sip:0017174695631@213.133.102.85:5060
1470  17.894954 83.211.224.67 -> 213.133.102.85 SIP/SDP Request: INVITE sip:0017174695631@213.133.102.85:5060, with session description
1471  17.896137 213.133.102.85 -> 83.211.224.67 SIP Status: 100 Trying
1472  17.896431 213.133.102.85 -> 83.211.224.67 SIP/SDP Status: 200 OK, with session description
1515  18.241112 213.133.102.85 -> 83.211.224.67 SIP/SDP Status: 200 OK, with session description
1540  18.499563 83.211.224.67 -> 213.133.102.85 SIP Request: ACK sip:0017174695631@213.133.102.85:5060
1545  18.533226 83.211.224.67 -> 213.133.102.85 SIP Request: ACK sip:0017174695631@213.133.102.85:5060
2075  24.030585 213.133.102.85 -> 83.211.224.67 SIP Request: BYE sip:103-DEVEL@83.211.224.67:46056;ob
2079  24.097717 83.211.224.67 -> 213.133.102.85 SIP Status: 200 OK
```

Second Leg SIP decoded Full PCAP dump

```
   1  0.000000 213.133.102.85 -> 216.115.69.144 SIP/SDP Request: INVITE sip:0017174695631@sip.flowroute.com, with session description
   2  0.161370 216.115.69.144 -> 213.133.102.85 SIP Status: 100 Trying
   3  0.164033 216.115.69.144 -> 213.133.102.85 SIP Status: 407 Proxy Authentication Required
   4  0.164144 213.133.102.85 -> 216.115.69.144 SIP Request: ACK sip:0017174695631@sip.flowroute.com
   5  0.164366 213.133.102.85 -> 216.115.69.144 SIP/SDP Request: INVITE sip:0017174695631@sip.flowroute.com, with session description
   6  0.326127 216.115.69.144 -> 213.133.102.85 SIP Status: 100 Trying
   7  2.651255 216.115.69.144 -> 213.133.102.85 SIP/SDP Status: 180 Ringing, with session description
1436  17.327910 216.115.69.144 -> 213.133.102.85 SIP/SDP Status: 200 OK, with session description
1437  17.328671 213.133.102.85 -> 216.115.69.144 SIP Request: ACK sip:12.194.223.216:5060;transport=udp
2040  23.342422 216.115.69.144 -> 213.133.102.85 SIP Request: BYE sip:+4551808863@213.133.102.85:5060
2041  23.344435 213.133.102.85 -> 216.115.69.144 SIP Status: 200 OK
```

Using the floppy disk link is possible to download the complete pcap of the call or just check a detailed pcap dump.

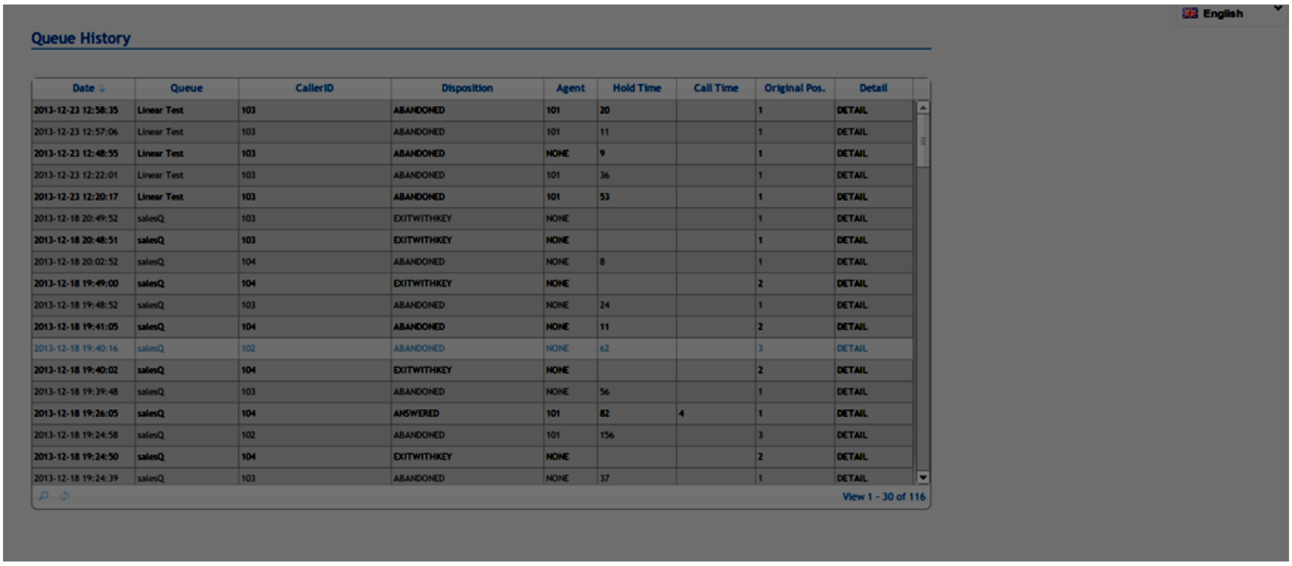
English

Full PCAP dump

```
 1  0.000000 83.211.224.67 -> 213.133.102.85 SIP/SDP Request: INVITE sip:0017174695631@demo.mirtapbx.com, with session description
 2  0.000986 213.133.102.85 -> 83.211.224.67 SIP Status: 401 Unauthorized
 3  0.090018 83.211.224.67 -> 213.133.102.85 SIP Request: ACK sip:0017174695631@demo.mirtapbx.com
 4  0.110133 83.211.224.67 -> 213.133.102.85 SIP/SDP Request: INVITE sip:0017174695631@demo.mirtapbx.com, with session description
 5  0.113996 213.133.102.85 -> 83.211.224.67 SIP Status: 100 Trying
 6  2.939414 213.133.102.85 -> 83.211.224.67 SIP Status: 180 Ringing
 7  2.939876 213.133.102.85 -> 83.211.224.67 SIP/SDP Status: 183 Session Progress, with session description
 8  3.122450 83.211.224.67 -> 213.133.102.85 RTCP Receiver Report   Source description
 9  3.124615 83.211.224.67 -> 213.133.102.85 RTCP Receiver Report   Source description
10  3.130347 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22185, Time=160, Mark
11  3.132517 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22186, Time=320
12  3.151119 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22187, Time=480
13  3.224677 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22188, Time=640
14  3.228285 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22189, Time=800
15  3.231805 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22190, Time=960
16  3.232795 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62784, Time=270268184
17  3.235493 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22191, Time=1120
18  3.252467 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62785, Time=270268344
19  3.272431 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62786, Time=270268504
20  3.277723 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22192, Time=1280
21  3.281245 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22193, Time=1440
22  3.292385 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62787, Time=270268664
23  3.309601 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22194, Time=1600
24  3.312370 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62788, Time=270268824
25  3.332610 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62789, Time=270268984
26  3.344405 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22195, Time=1760
27  3.348009 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22196, Time=1920
28  3.352583 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62790, Time=270269144
29  3.372660 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62791, Time=270269304
30  3.380899 83.211.224.67 -> 213.133.102.85 RTP PT=ITU-T G.711 PCMA, SSRC=0x308B840F, Seq=22197, Time=2080
31  3.392612 213.133.102.85 -> 83.211.224.67 RTP PT=ITU-T G.711 PCMA, SSRC=0x535CC723, Seq=62792, Time=270269464
```

Queue History

The info reported in the Call History are not enough when the call is processed using a Queue. More info for each call using the queue system can be found in the Queue History and the Caller ID, a detailed disposition of the call is shown with the agent who gets the call, the hold time of the caller and the call time. The original position of the caller, when the queue is accessed is also reported.
More detail of the call can be seen following the DETAIL link.



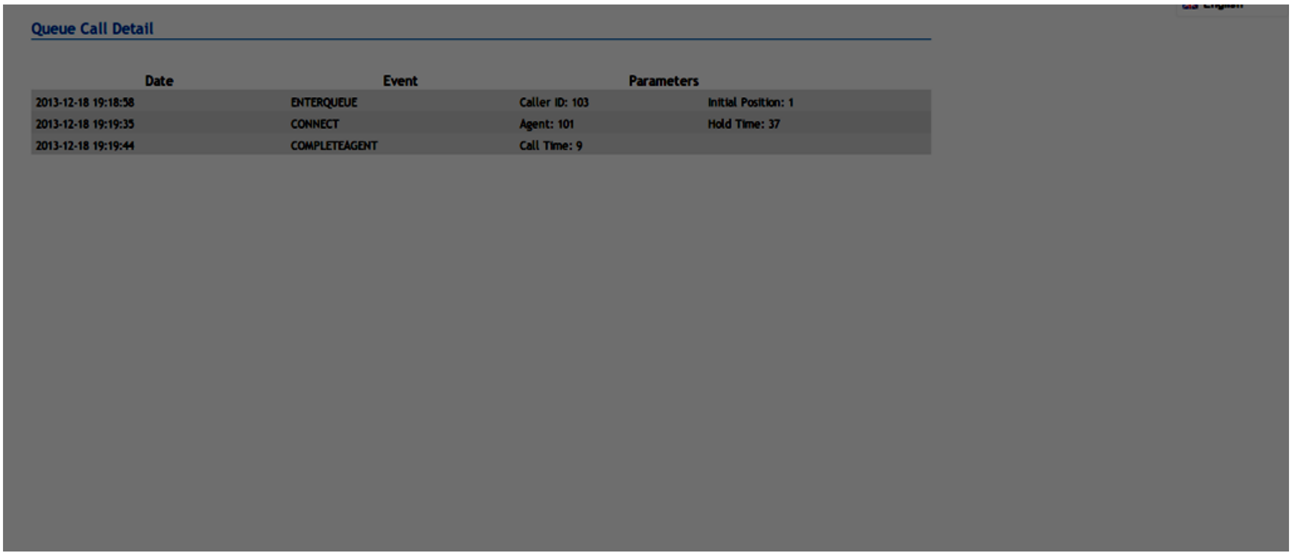
Queue History

Date	Queue	CallerID	Disposition	Agent	Hold Time	Call Time	Original Pos.	Detail
2013-12-23 12:58:35	Linear Test	103	ABANDONED	101	20		1	DETAIL
2013-12-23 12:57:06	Linear Test	103	ABANDONED	101	11		1	DETAIL
2013-12-23 12:48:55	Linear Test	103	ABANDONED	NONE	9		1	DETAIL
2013-12-23 12:22:01	Linear Test	103	ABANDONED	101	36		1	DETAIL
2013-12-23 12:20:17	Linear Test	103	ABANDONED	101	53		1	DETAIL
2013-12-18 20:49:52	salesQ	103	EXITWITHKEY	NONE			1	DETAIL
2013-12-18 20:48:51	salesQ	103	EXITWITHKEY	NONE			1	DETAIL
2013-12-18 20:02:52	salesQ	104	ABANDONED	NONE	8		1	DETAIL
2013-12-18 19:49:00	salesQ	104	EXITWITHKEY	NONE			2	DETAIL
2013-12-18 19:48:52	salesQ	103	ABANDONED	NONE	24		1	DETAIL
2013-12-18 19:41:05	salesQ	104	ABANDONED	NONE	11		2	DETAIL
2013-12-18 19:40:16	salesQ	102	ABANDONED	NONE	62		3	DETAIL
2013-12-18 19:40:02	salesQ	104	EXITWITHKEY	NONE			2	DETAIL
2013-12-18 19:39:48	salesQ	103	ABANDONED	NONE	56		1	DETAIL
2013-12-18 19:26:05	salesQ	104	ANSWERED	101	82	4	1	DETAIL
2013-12-18 19:24:58	salesQ	102	ABANDONED	101	156		3	DETAIL
2013-12-18 19:24:50	salesQ	104	EXITWITHKEY	NONE			2	DETAIL
2013-12-18 19:24:39	salesQ	103	ABANDONED	NONE	37		1	DETAIL

View 1 - 30 of 116

Peers

Info on all the connected peers are shown along with all the info gathered during the registration process, like the brand and firmware version of the phone and even the internal IP address.

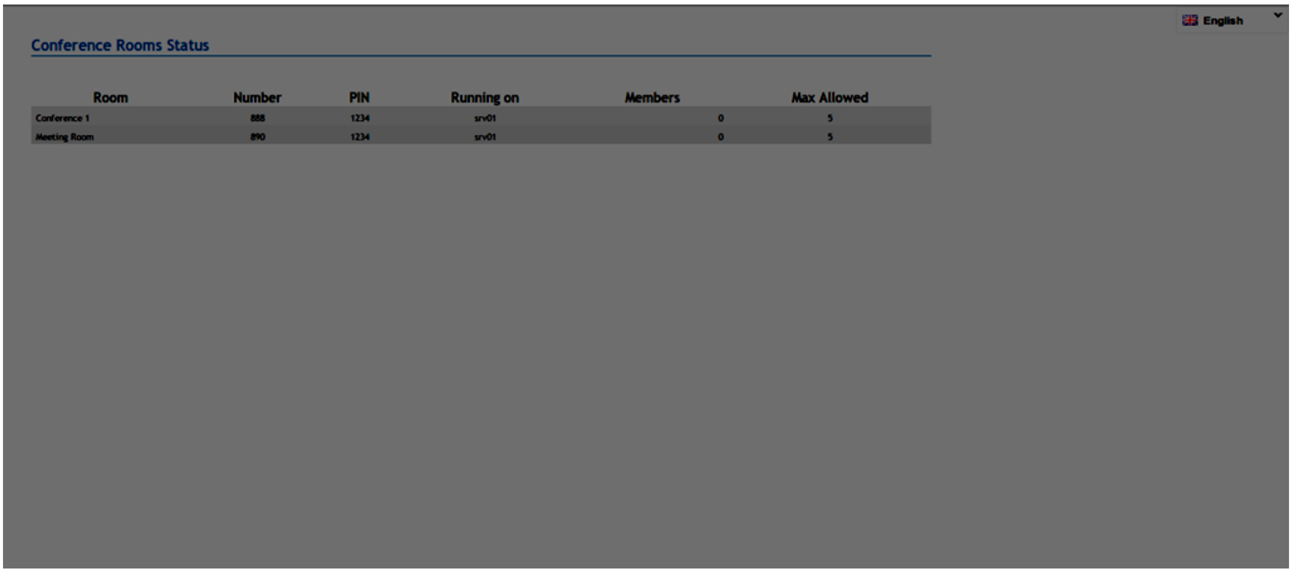


Queue Call Detail

Date	Event	Parameters	
2013-12-18 19:18:58	ENTERQUEUE	Caller ID: 103	Initial Position: 1
2013-12-18 19:19:35	CONNECT	Agent: 101	Hold Time: 37
2013-12-18 19:19:44	COMPLETEAGENT	Call Time: 9	

Using the big red cross is possible to unregister a phone. Unregistering a phone is needed every time there is a configuration for the phone; like its password or the codec are changed. Some phone models, like Tiptel and Yealink, allow remote reboots the phone using the circled red arrow.

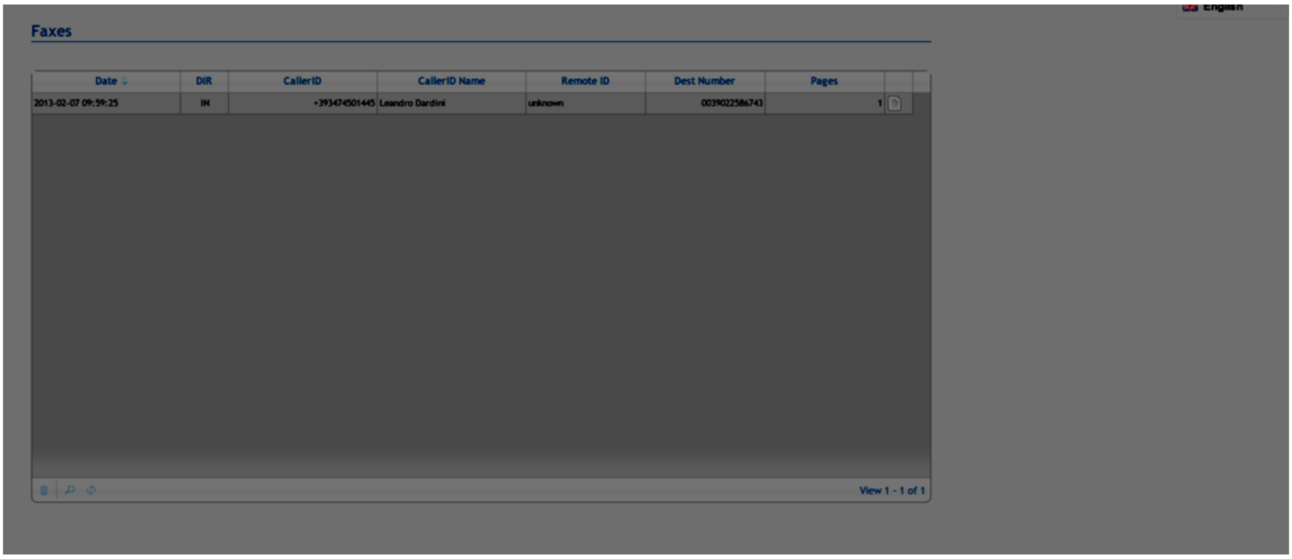
Conferences



Using this menu, the status of all the conference rooms taking place on the server, for the selected PBX can be shown.

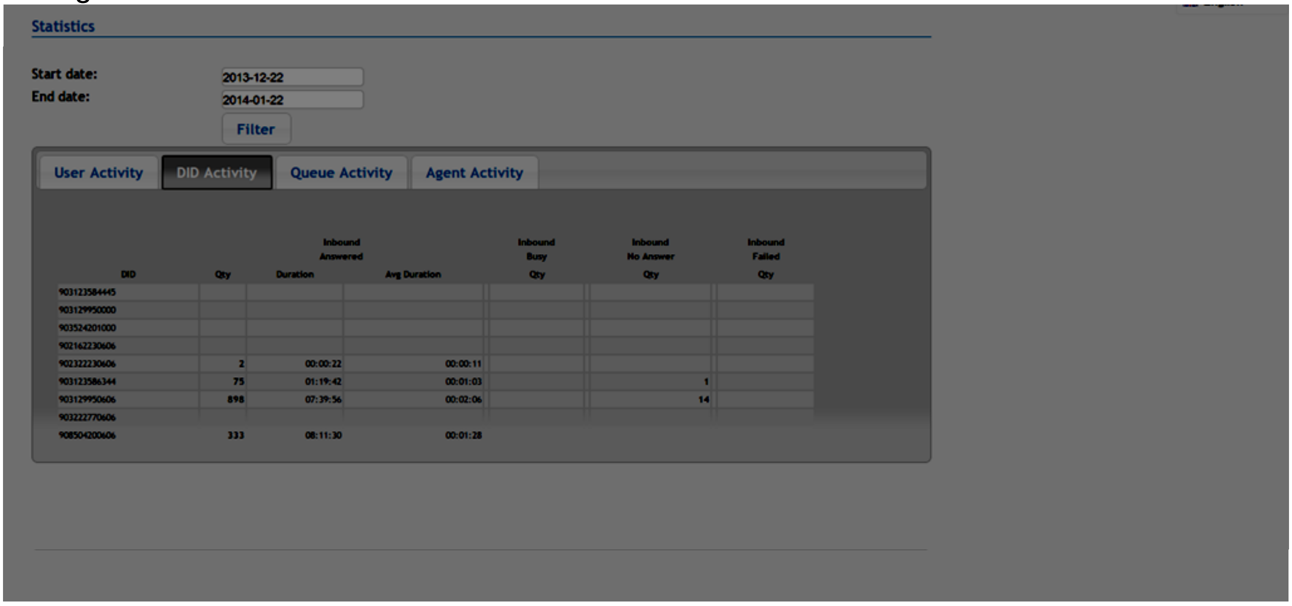
Faxes

All faxes received by the system for the selected PBX can be viewed here if the fax has been configured for storage



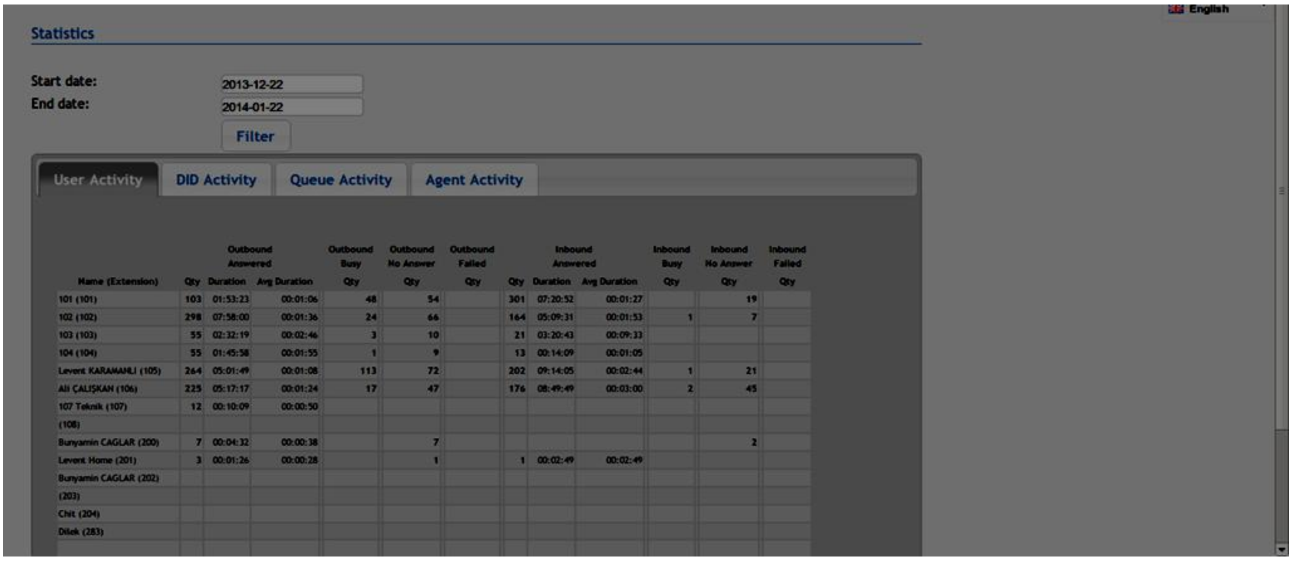
Voicemail Messages

Voicemail messages received by the PBX can be viewed here if they are configured for storage

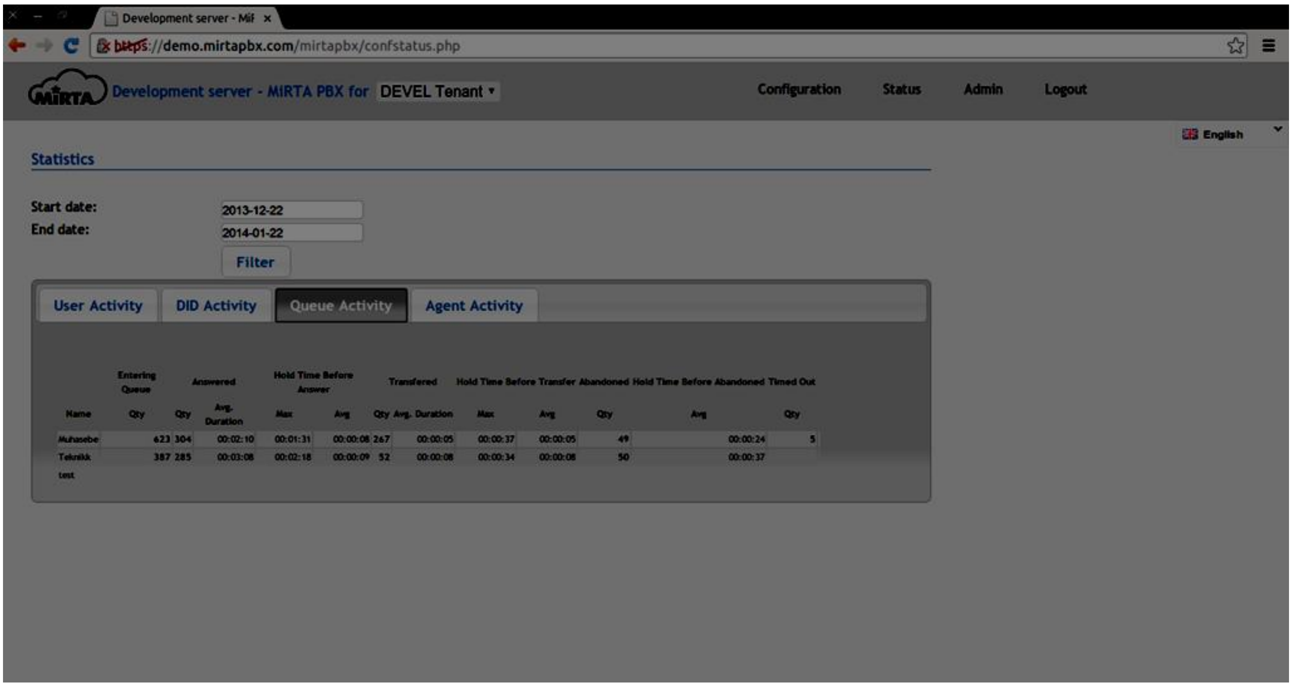


Stats

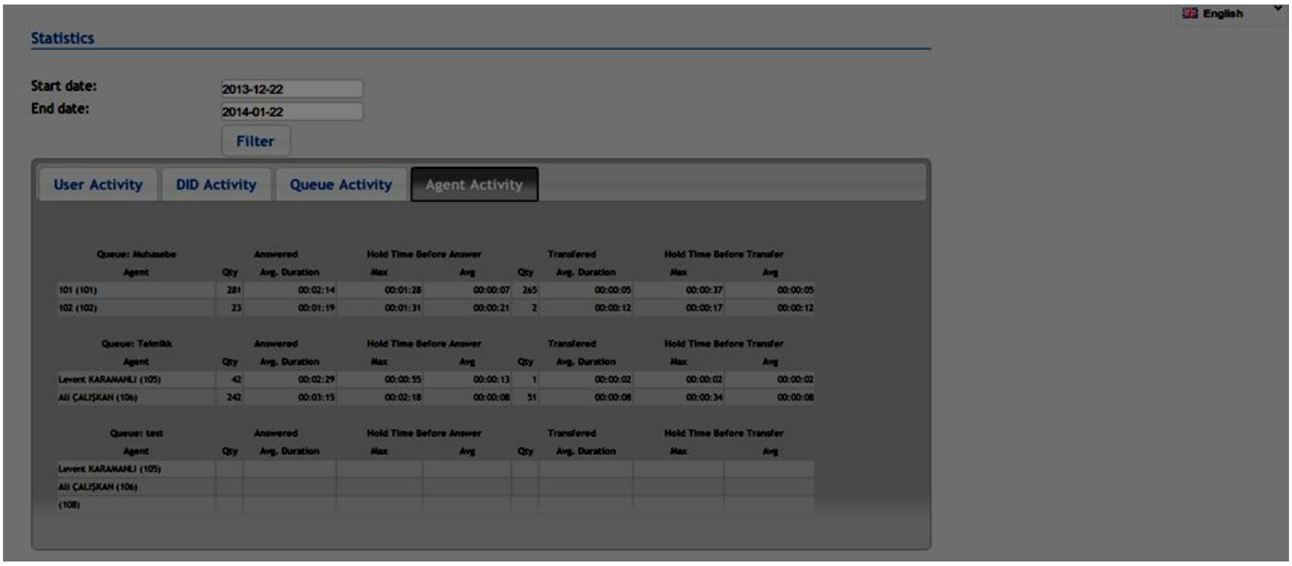
Several statistics are available in the system for inbound, outbound and queue/agents calls. Clicking on the values will show the calls counted.

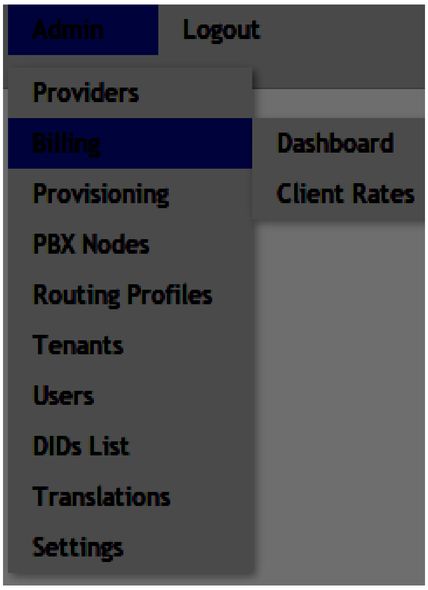


Stats are available for User Activity showing Outbound Calls Answered, Busy, No Answered and Failed, Inbound Answered, Busy, No Answered and Failed.
Stats are available for DID Activity showing Inbound Calls Answered, Busy, No Answered and Failed for each DID.
Stats are available also for Queue Activity and Agent Activity, showing the calls entered in the queue and how the call ended, if answered, transferred, abandoned or time out, along with the holding time.



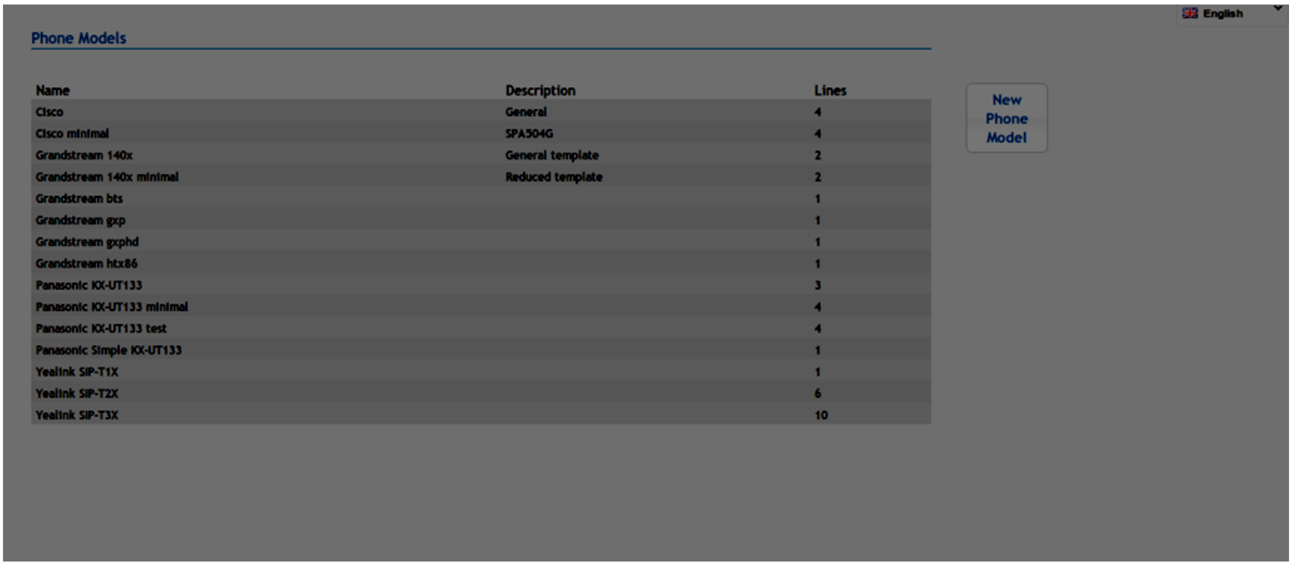
The same values can be read divided by each agent and each queue:





Provisioning

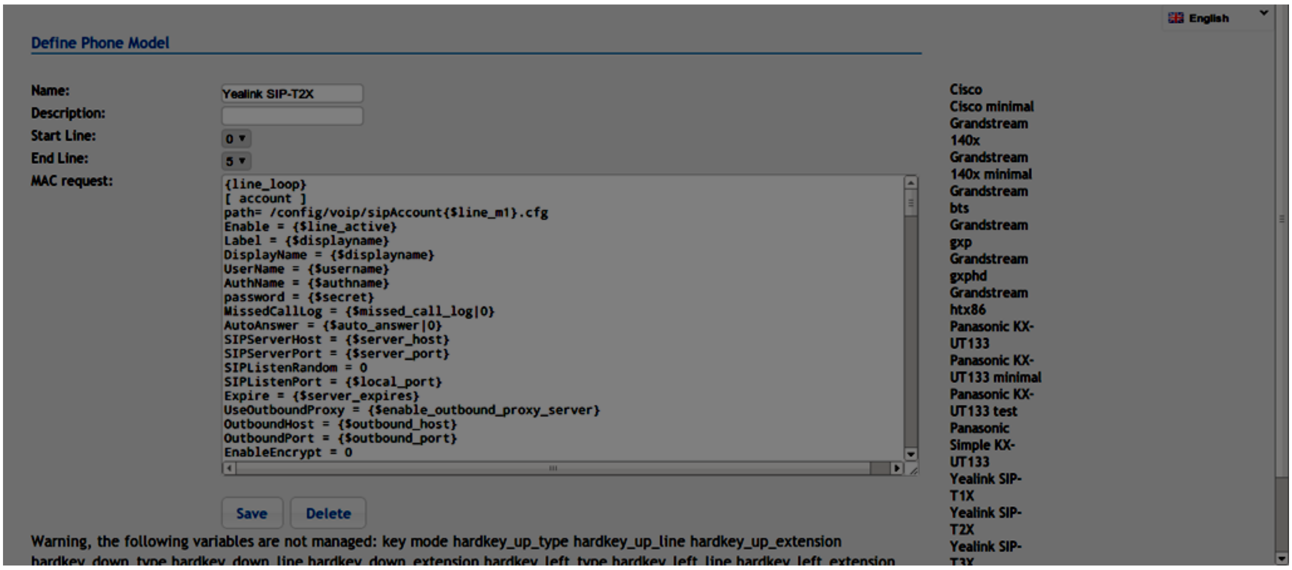
Provisioning of phones is phone independent by providing the framework to create a template and use some variable. Variables are predefined in the system, but more can be added easily.



Name	Description	Lines
Cisco	General	4
Cisco minimal	SPA504G	4
Grandstream 140x	General template	2
Grandstream 140x minimal	Reduced template	2
Grandstream bts		1
Grandstream gxp		1
Grandstream gxphd		1
Grandstream htx86		1
Panasonic KX-UT133		3
Panasonic KX-UT133 minimal		4
Panasonic KX-UT133 test		4
Panasonic Simple KX-UT133		1
Yealink SIP-T1X		1
Yealink SIP-T2X		6
Yealink SIP-T3X		10

Phone Models

Several phone models can be defined. For each phone model the number of lines needs to be defined, usually with the starting and ending line number.
The MAC request contains the text file that needs to be delivered to the phone for the provisioning. Please inquire for your phone model to receive the correct template.



SIP TAPI integration

SIP TAPI integration is made using the freely available SIPTAPI application: http://sourceforge.net/projects/siptapi/

*** IMPORTANT ***
The SIP TAPI integration is made by dialing your own extension and then connecting to the destination. It is important the “Recover VM messages dialing your own same number” is set to No.

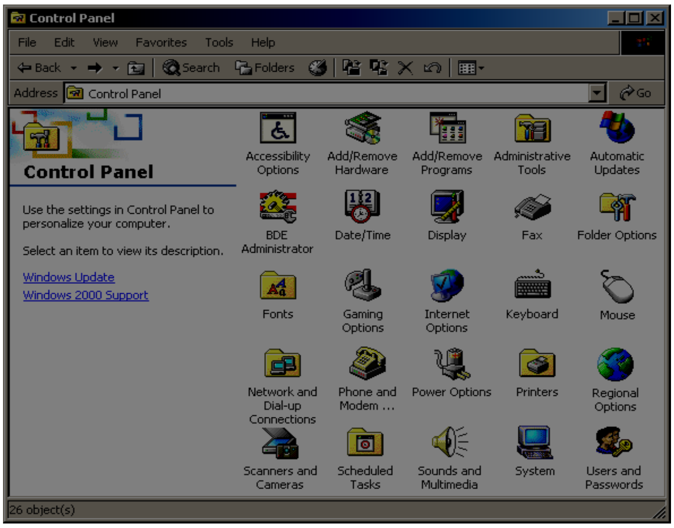
The following example has been made using version 0.2.13.

Download and uncompress the zip package

On 32bit Windows
================
1. Copy siptapi.tsp from 32bit folder into your Windows system32 directory (usually C:\Windows\System32)
2. Install and configure SIPTAPI using the telephony options from control panel

On 64bit Windows
================
1. Copy siptapi.tsp from 64bit folder into your Windows system32 directory (usually C:\Windows\System32). This gives you full TAPI support with 64 and 32bit TAPI applications

2. If you want to configure SIPTAPI not only from the telephony control panel and from 64bit applications, but also from 32bit applications, copy the 32bit siptapi.tsp from 32bit



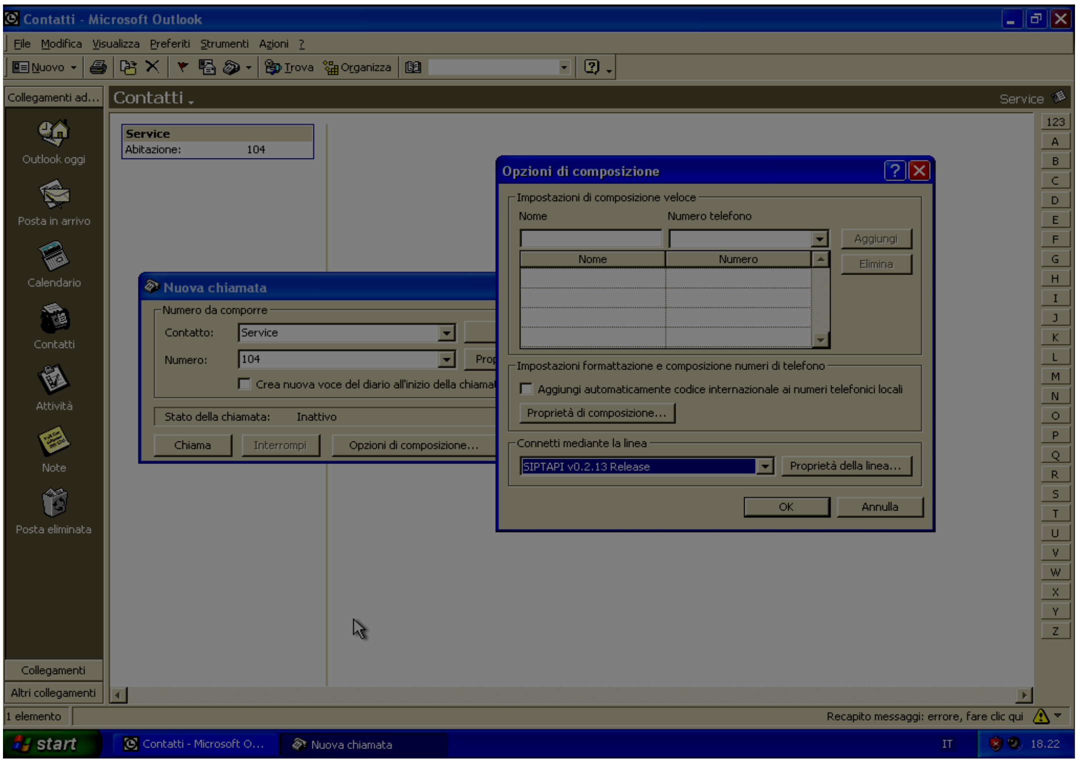
folder into your WoW64 directory (usually C:\Windows\SysWow64)
3. Install and configure SIPTAPI using the telephony options from control panel

Open Control Panel and click on the Phone and Modem Option

In the Advanced tab add a new Provider

Select “SIP TAPI Service Provider...” and press Add.
Configure it by entering the PBX host address, the extension and the password

Open you TAPI compatible application, like Outlook and dial any contact. You need to configure the application to dial using the SIPTAPI.

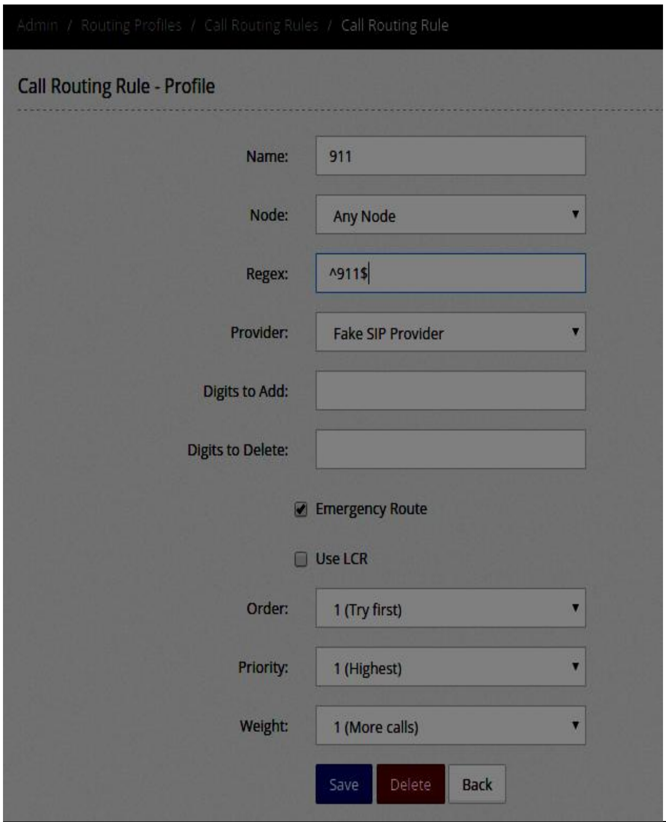


Setup Guides

How can I setup and use the Emergency Caller ID?

To be able to use Emergency Caller ID, you need to follow the following steps:

Create a routing rule using Admin/Routing Profiles for all the “emergency numbers” you like to use with the Emergency Caller ID. So for example you need to create a specific rule for 112, 113, 911, 999

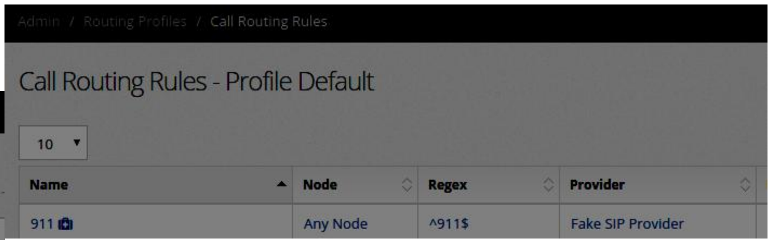


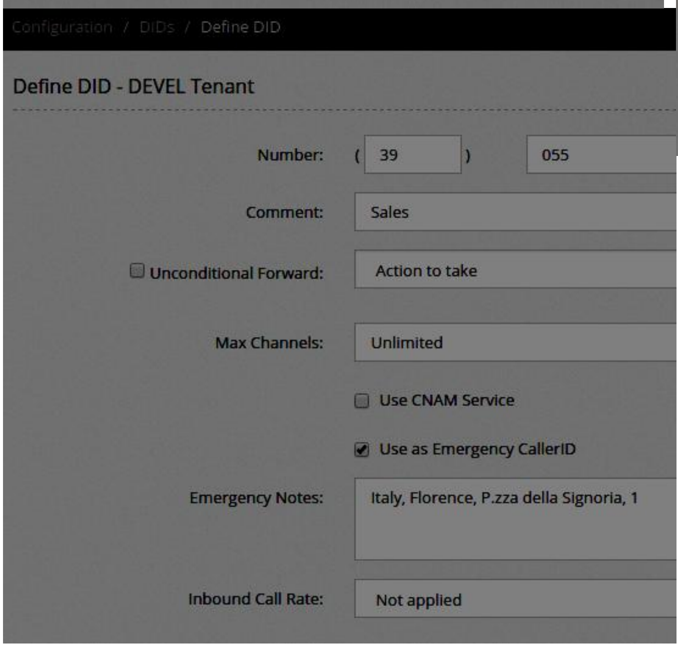
or whichever number is used in your country. It can be good to add also a normal number to check if the Emergency Caller ID usage has been configured correctly.

Please don't forget if multiple routing rules match your dialed digits, they are evaluated by Order and Priority. It can be a good idea to assign common rules to order 2 and emergency routing rules to order 1.

To match only and exactly 911 for example, you need to use the ^911$ regex.

Assign the “Emergency Route” flag to the extension. It will be shown as a little medical kit icon on the routing rules page.

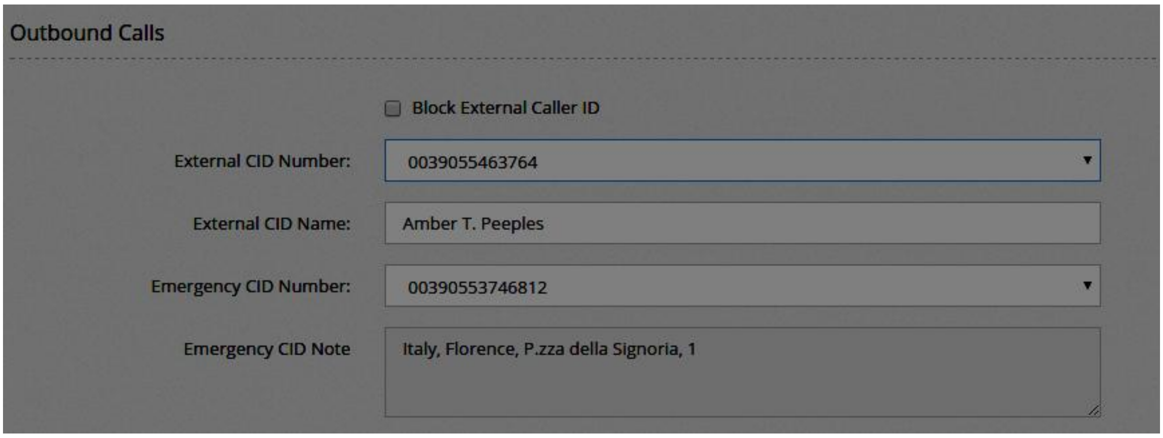




Select the DIDs you want to use as Emergency Caller ID using the Configuration/DIDs page. Before to be able to use a DID as “Emergency Caller ID”, the flag “Use as Emergency CallerID”. A box will be automatically shown to let you insert a note about the location of this DID. The “Emergency Enabled

+39 055 3746812	Sales	Unlimited	no
+39 055 453131	Test	Unlimited	no
+39 055 453135	Manager	1	no

DID” will be marked in the DIDs list with a little medical kit icon.



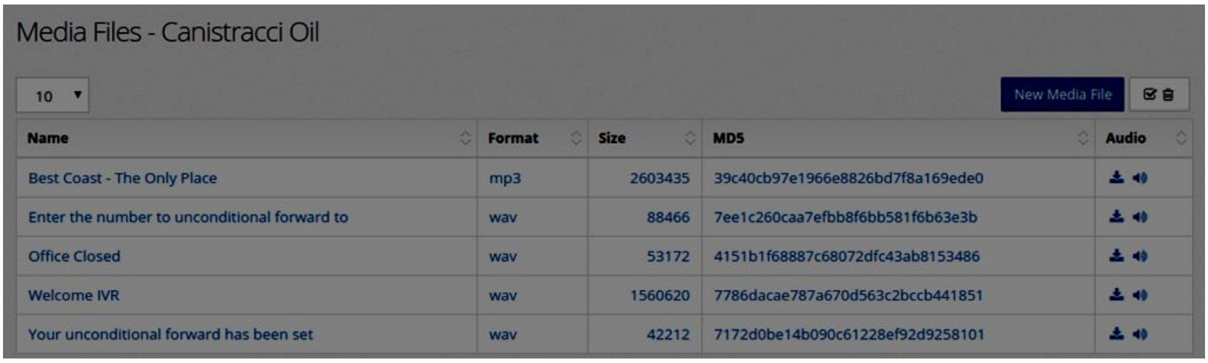
Assign the specified Emergency Caller ID to the extensions using the Configuration/Extensions page. Locate the ”Outbound Calls” section and assign the desired DID to be used as Emergency Caller ID when an emergency route is used.

How can I build a more user friendly feature code?

Some clients find the feature codes too rough and they prefer to have more user friendly feature even if it is slower to use. Normally, a feature code asks nothing, but accept all the parameters on the dialing itself, like for example *72[NUM] can be configured to set unconditional destination for a DID to the NUM dialed. However, feature codes and custom destination are the building blocks you can use to create something more complex and maybe user friendly. Let's see some examples:

Ask and set unconditional destination for a DID

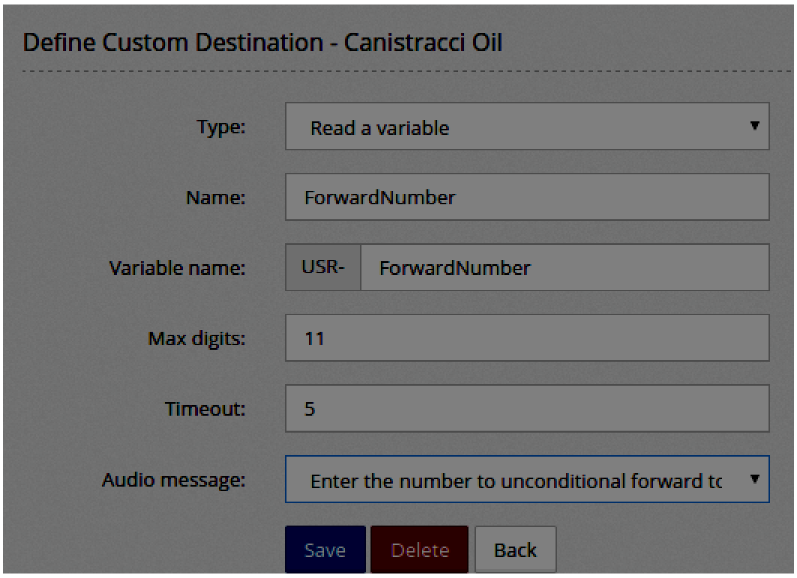
Our client likes to dial a feature code, like *72 and be prompted with a message asking to enter a number. That number will be set as unconditional destination for a DID. These are the steps:



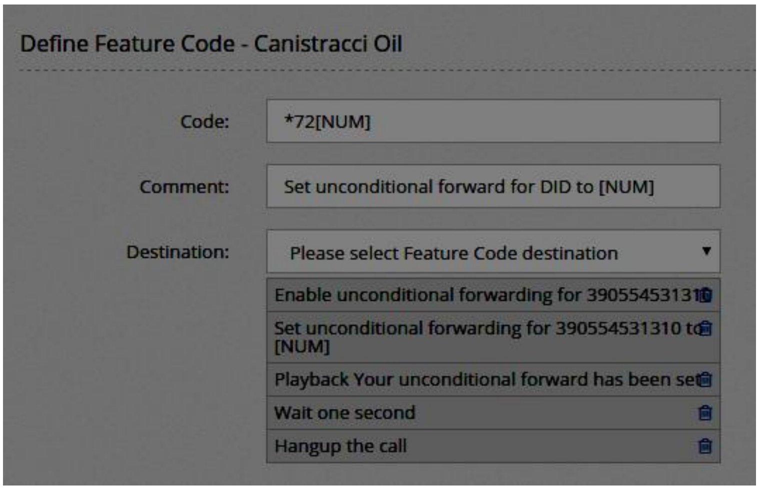
Create a recording asking for the number to use for forwarding your DID and load in the Media File sections.

You can create also a recording to confirm the configuration.

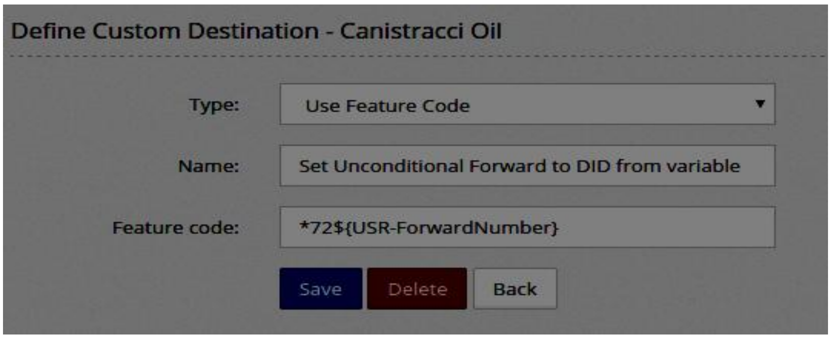
Create a custom destination to read a number entered by the user and use the above recording as prompt.



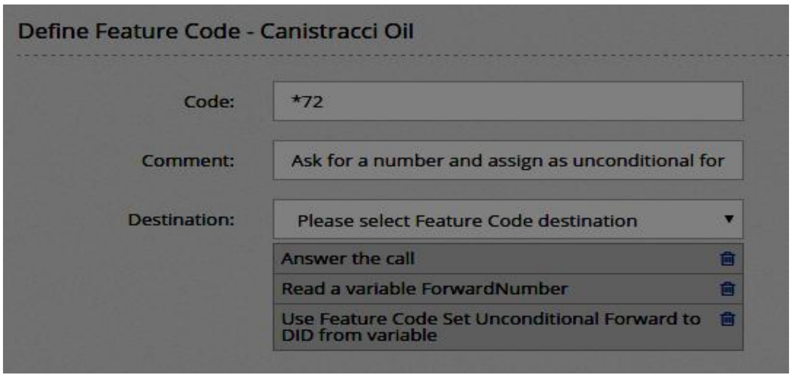
Create the “raw” feature code setting the unconditional forward for the number and playing the confirmation message.



Create a custom destination to use the above feature code with the variable entered.

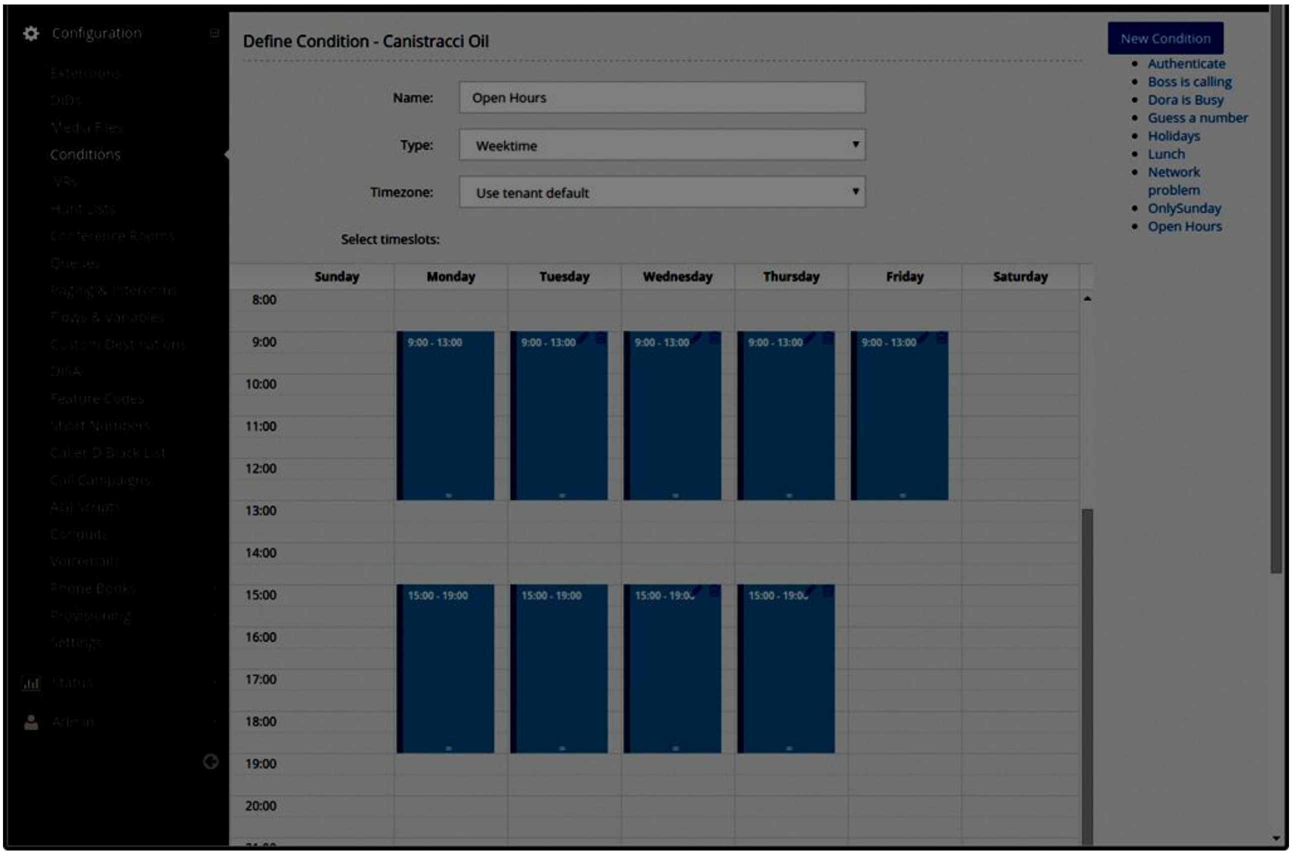


Create a feature code asking for the number and using that number to execute the above feature code.



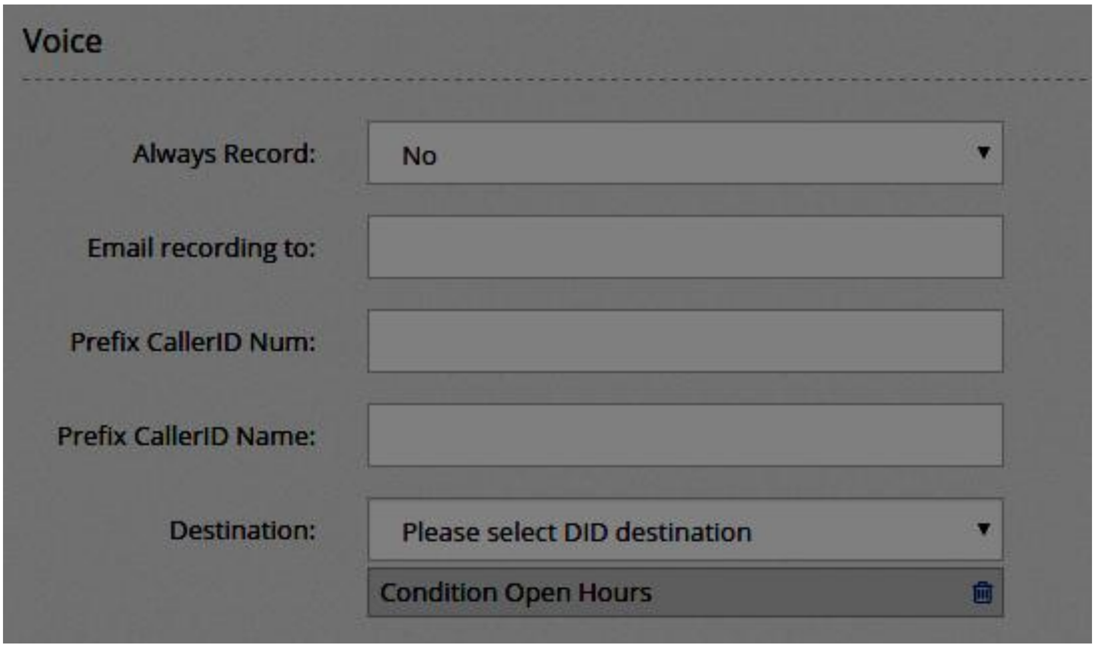
Condition override

Sometimes it can be useful to have a condition for Open/Close hours of the office, but a button on your BLF keyboard to override the condition, closing it during Open hours (maybe because you are going to have an extra coffee) or opening during Close hours.



Let's start by creating your time condition in the usual way:

Destination will be your normal destination for Open/Close hours:

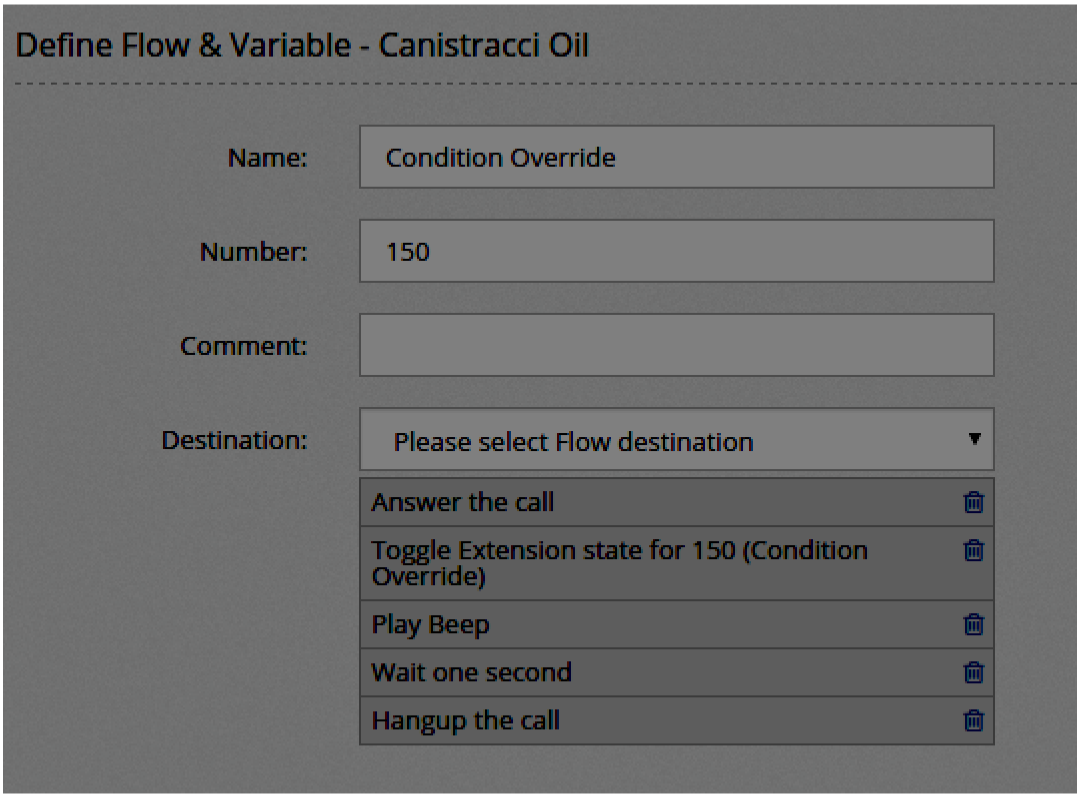


In your DID, you have assigned the Condition in the usual way:

Now we are going to create a BLF key to override this condition.

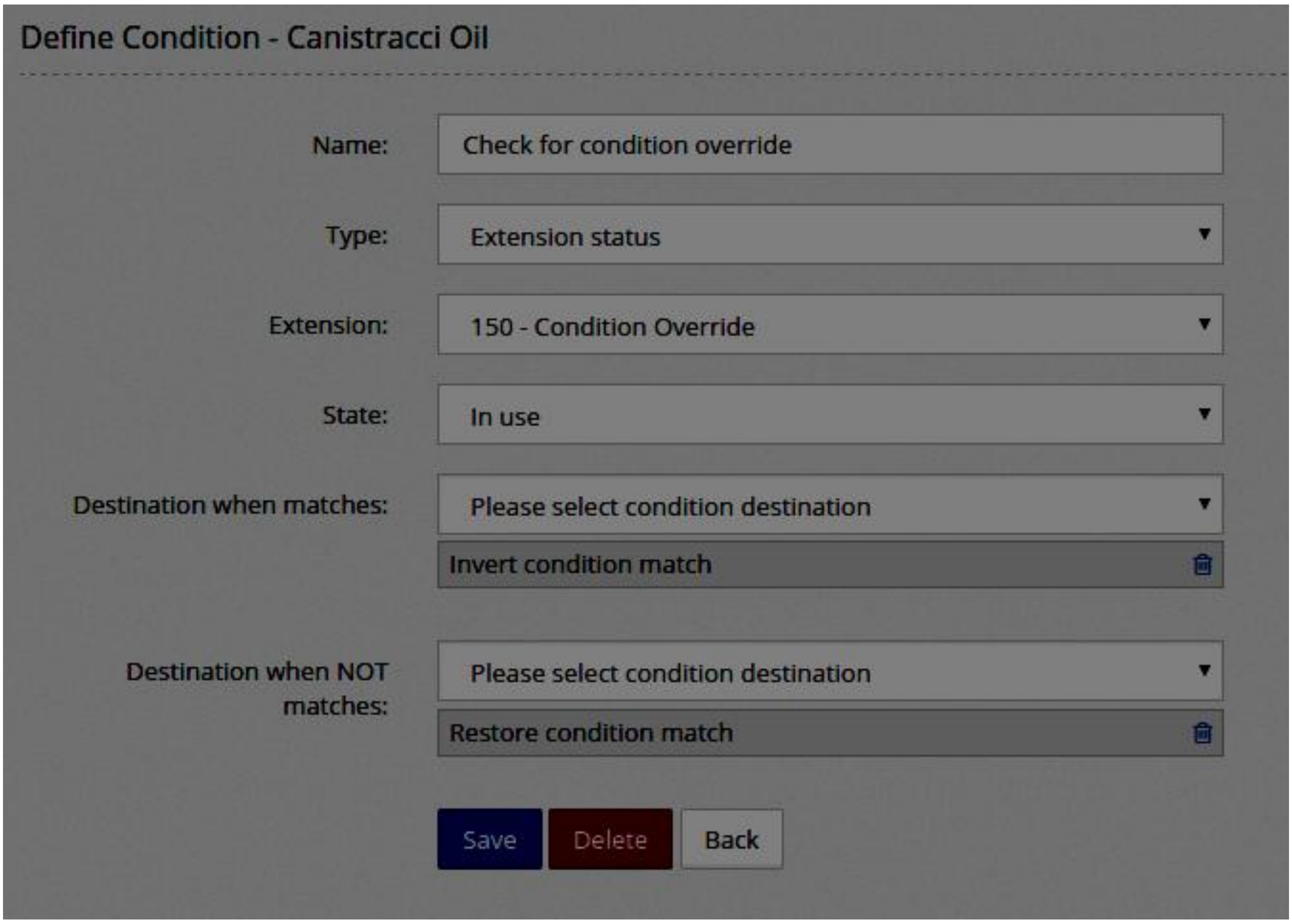
We need to create the BLF number, using a Flow. Just assign for example the number 150 to the key, so this client will monitor 150.

It can be useful to start creating the custom destination to toggle the BLF state:



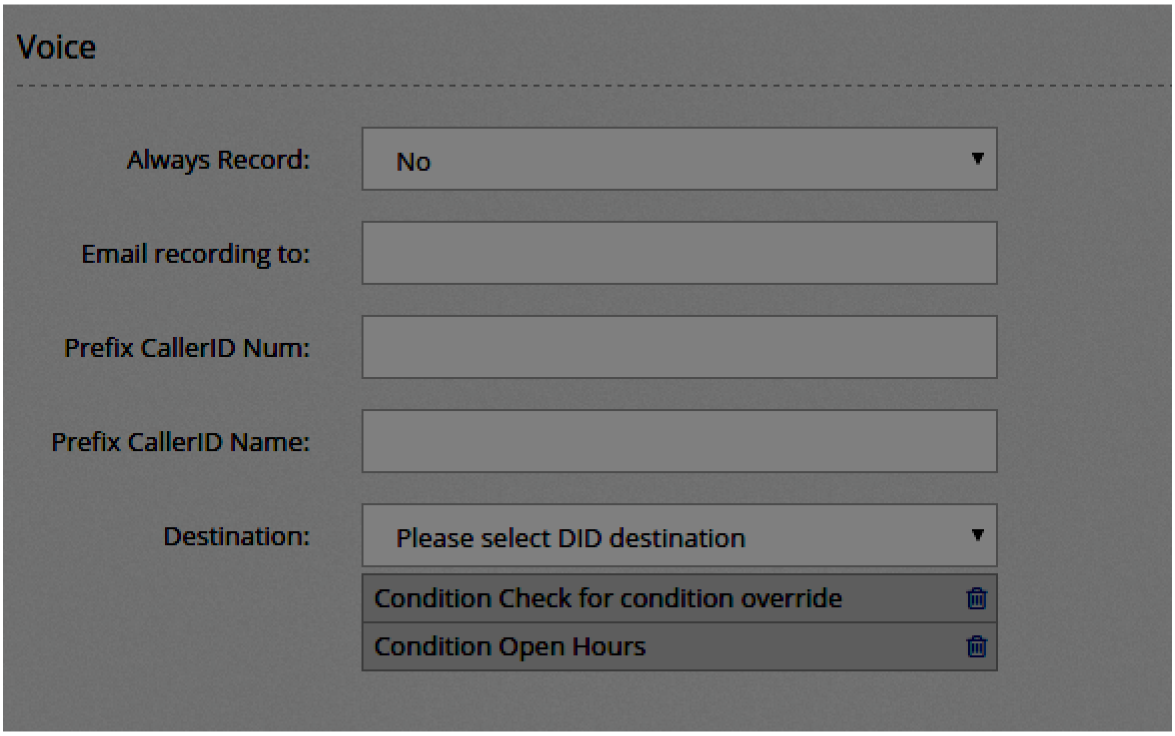
And use it in the Flow definition:

Remember to set the initial state, just assign NOT IN USE (green light) to normal condition and IN USE (red light) to condition override.



Now we need a condition to test for the BLF state and invert/override the condition.

Assembling all in the DID destination, we get:

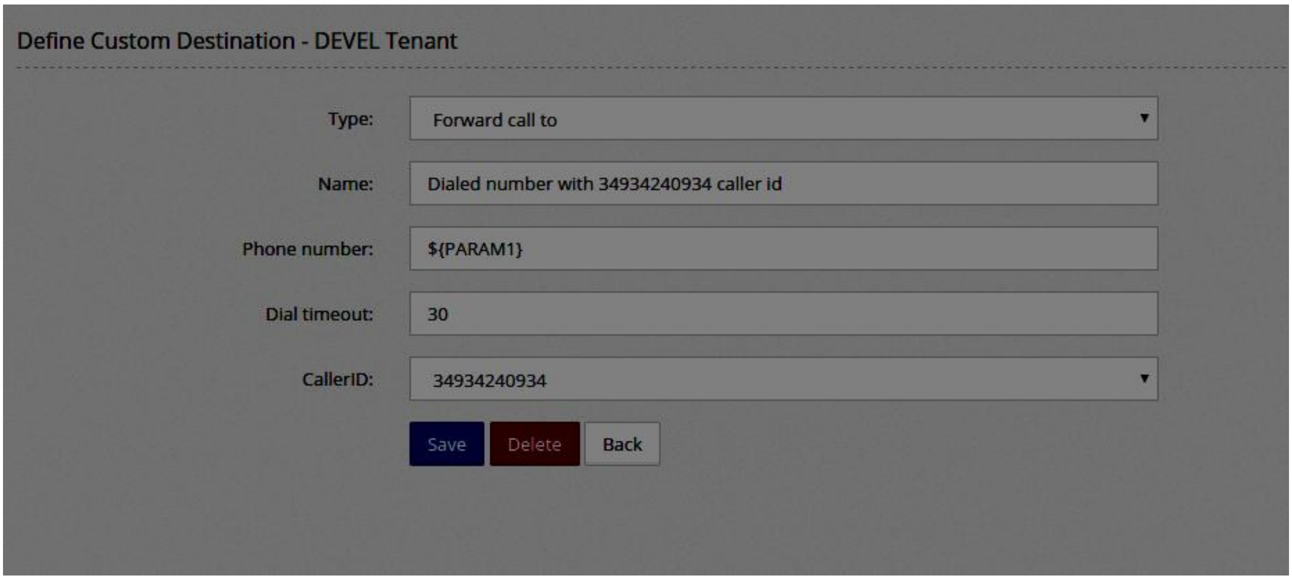


Configuration examples

Dial a number with different Caller ID

Sometimes an office can have an extension serving more companies, so when dialing out for company “Kartoon Cars” a different Caller ID needs to be used than when dialing out for company “Sleep Flex”. To accomplish this, we use two feature codes with two custom destinations. The idea is to dial *11 plus the number to dial out when you want to use the first Caller ID and *22 plus the number to dial out when you want to use the second Caller ID.

Let's start by configuring two custom destinations as following:



And in a similar way, but for the other Caller ID number.

Once configured, you need to create a feature code using each one of the custom destinations created.