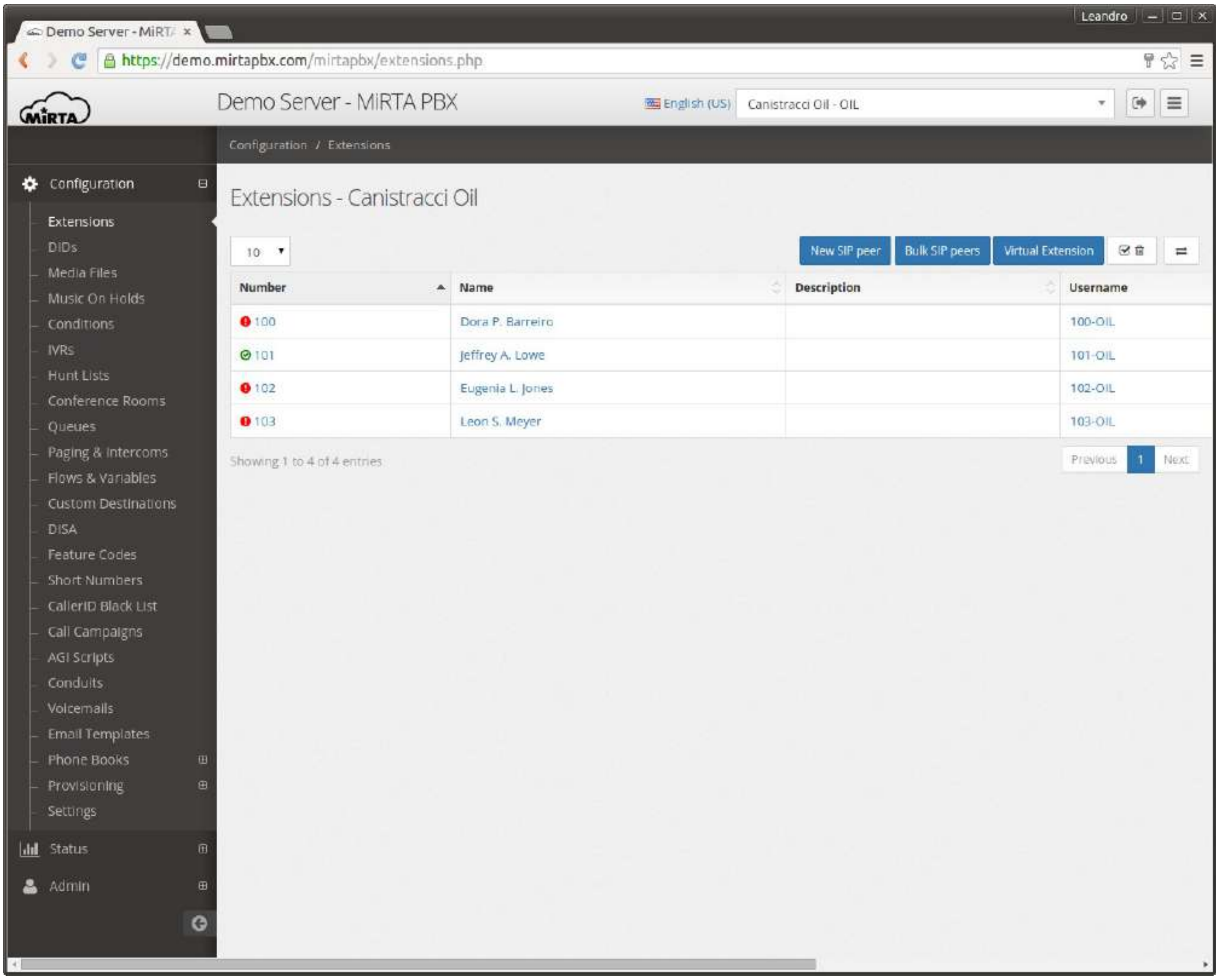
Configuration Section

The Configuration Section is used to configure every working aspect of the PBX. It can used by admin and not admin users. The options selected within this menu are applied only to the selected tenant using the “Tenant Selection Menu”.

Extensions

The list of extensions defined for the selected tenant are shown along with the callerid, 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.

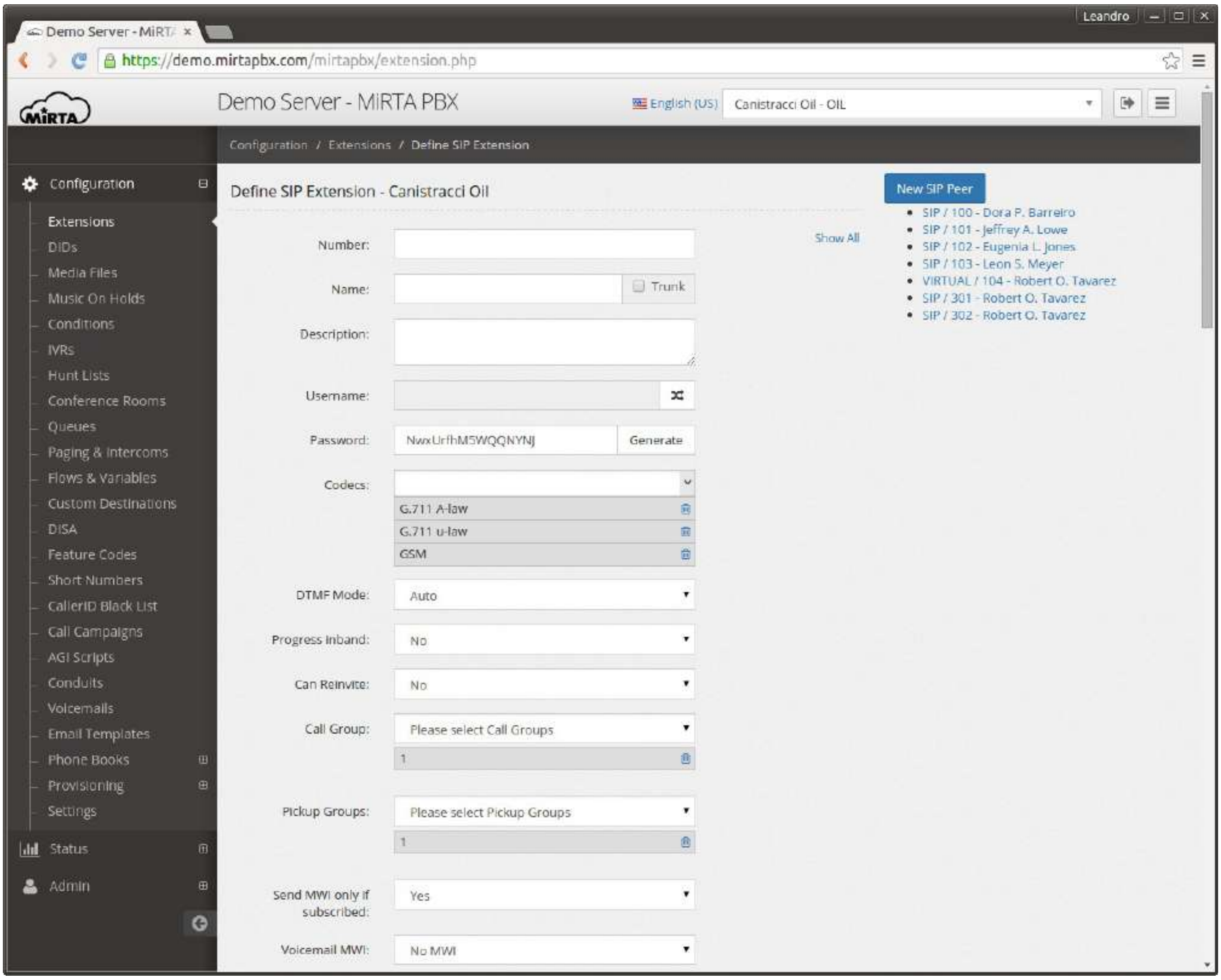


New SIP Extension/Define SIP Extension

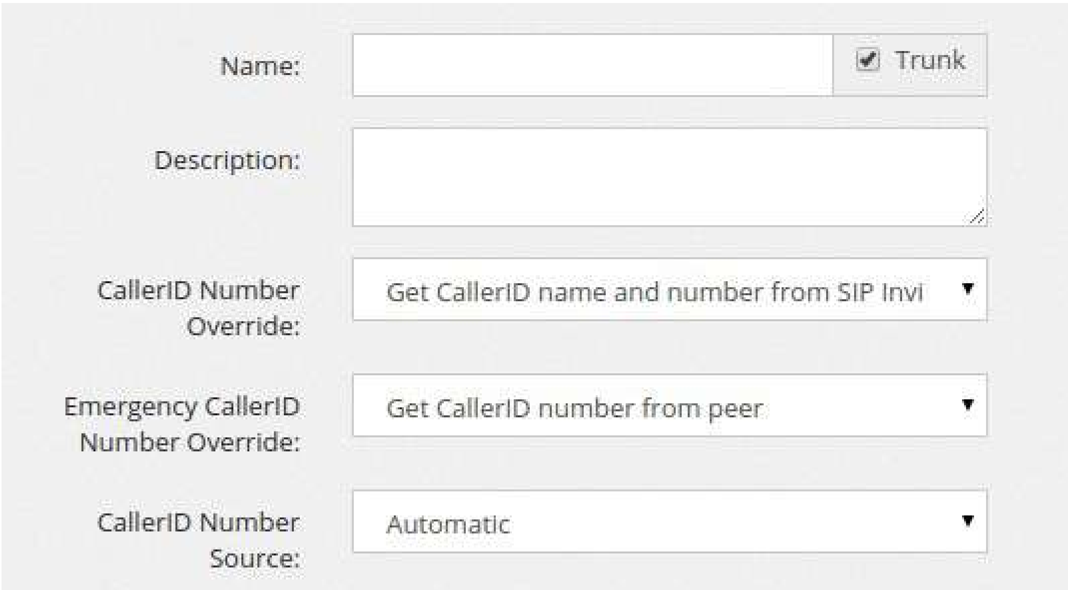
The definition of an extension is comprised of multiple sections. The most important one is the general one, where you can define the internal number for the extension and the password.

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

The number assigned to an extension for a tenant can be the same number assigned to another extension for another tenant. MiRTA PBX is completely multi tenant, so each tenant configuration is completely independent from others. This is a general rule and apply on every aspect of the configuration.



The name provided will be used as CallerID for internal calls. This means the CallerID on the phone will be overwritten with the one specify here. If you don't want to have the CallerID forced to the one configured, but rather you want to use the CallerID 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 effect also incoming call to the phone (or PBX). If the “Trunk” checkbox is set, the SIP INVITE sent to the account will include the number dialed.



When you select the Trunk checkbox, you have access to few more options to control how the CallerID Number for the calls generated by that extension will be altered:

You can get both the CallerID number and Name from the SIP Invite, just the number, just the name or neither of them. You can also specify from which portion of the SIP packet to get the CallerID number and name, Automatic (asterisk default) or from FROM, PAI or RPID section.

Username is automatically generated adding the tenant code to the number provided. The format used by default is using the “-”, but some phones has 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 “_” is discouraged and needs to be used only when really needed.

Password can be autogenerated 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 the G.729/723.1 codecs, even if listed in the system, you might 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.

Can reinvite 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 of the party is behind NAT, you may experience one way audio. Usually set to No.

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

Call groups/Pickup groups defines 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 allow usually 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 eventually set.

Inbound Dial Timeout sets the time in seconds one extension have to ring before going to the "No Answer" additional destination. You can avoid setting a Dial Timeout value and the default value will be used.

Recording

Always Record sets the recording preference for the extensions. If 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 to allows to set an email address to send the recordings once the call completes.

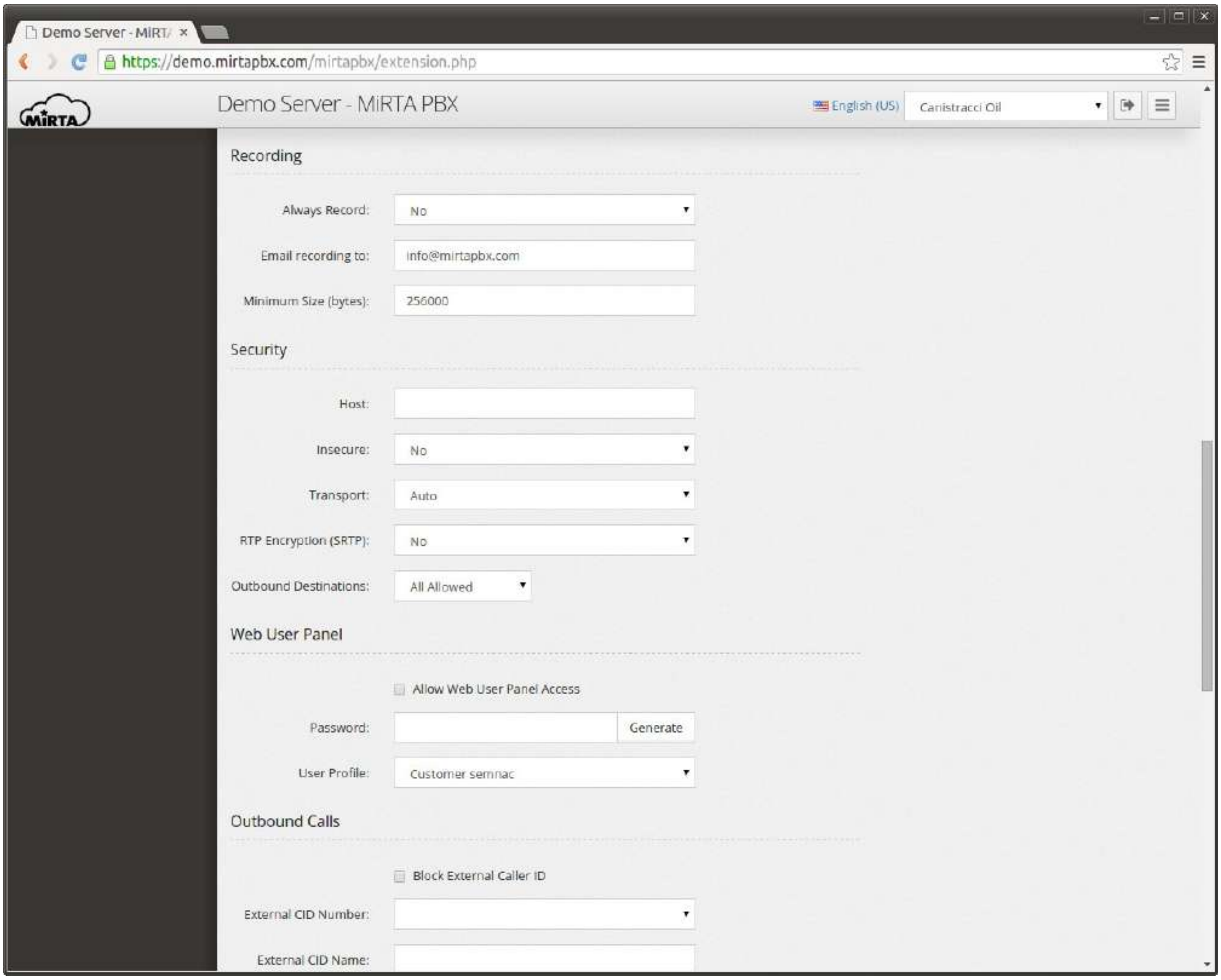
Minimum Size (Bytes) allows to receive those recordings bigger than the size set, in Bytes. Recording takes place only on bridged channels, so IVR prompts or Music On Hold will be not recorded.

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 to restrict the numbers the extension can dial. In other words, the destination allowed can be restricted. 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

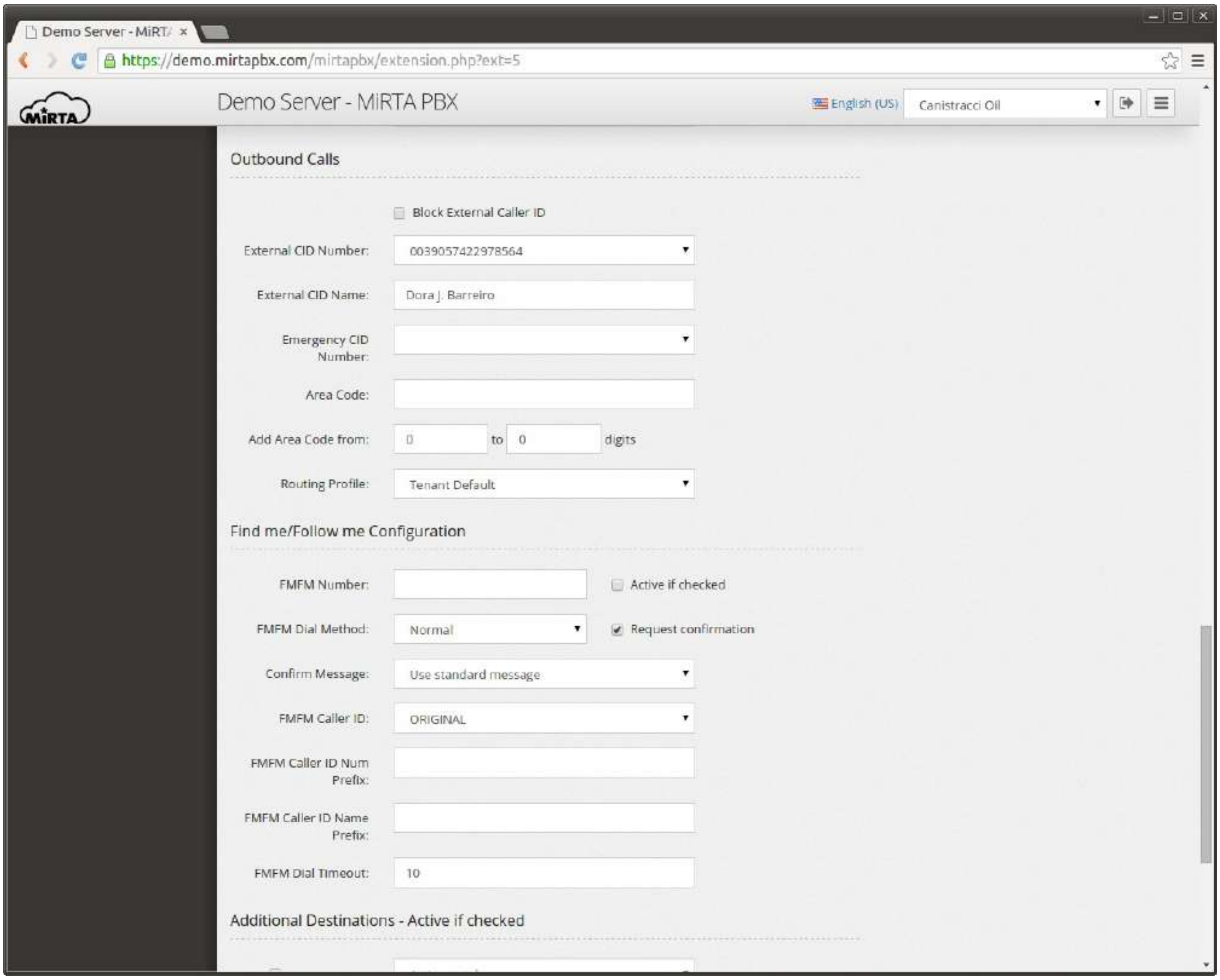
Allowed if matches: The call is allowed if the number dialed match the Regex associated

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

Web User Panel

Enabling the web user panel permits to 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 to assign to the user connecting to the web user panel.



Outbound Calls

This section allows to configure how the call is managed when dialing out the local virtual 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. The various options can be enabled or disabled using the Admin/Settings menu. Under normal confition, the External CID number can be chosen only among the DIDs assigned to the tenant. However if the user has the privilege “Can Edit CID Numbers”, a small “Edit” box will appear beside the number, allowing the user to customize the External Caller ID number.

External CID name allows to define 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 to specify a prefix to add to numbers when the number of digits entered is between the number of digits specified next, inclusive. For example, if you area code is 055 and your local area numbers are from 4 to 7 digits, you can enter the following data and your number will be automatically completed with the area code. So, if you enter 453131, automatically the number dialed will be 055453131.

Routing Profile permits to assign to the extension a different routing profile than the one assigned to the tenant.

Find me/Follow me Configuration

It allows to define 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 permits to choose between two dialing method, “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 to ask 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 to choose which Caller ID to display to the called number. Two special options are available:

Use Original will use the caller 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 permit to specify the destination of the call when the extension is not answered, Busy or Offline. A special destination "Unconditional" allows to redirect 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.

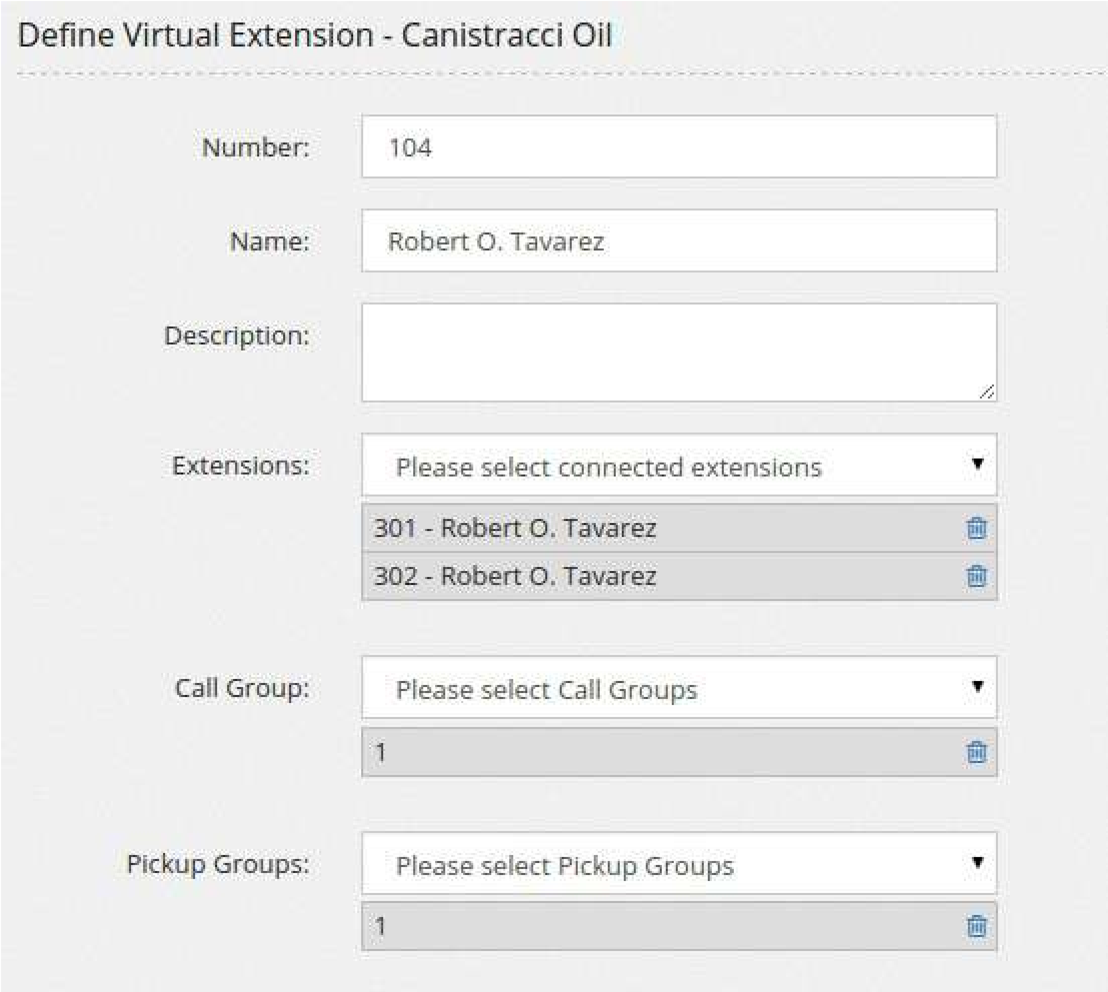
Virtual Extensions

A virtual extension is an extension that is not connected automatically to a SIP device, so you can connect multiple devices to the same extension number. When one of them will be busy, the virtual extension will be shown as busy. When the virtual extension number is dialed, all SIP devices connected will ring.

104	Robert O. Tavarez		301 302	
301	Robert O. Tavarez	Home	301-OIL	bP8em7e9UZUHmwss
302	Robert O. Tavarez	Office	302-OIL	6WKn4bEcYVXFTdTj

All usual options regarding an extension are available in the virtual extension. When an extension is part of a virtual extension, the additional options beside the main one are no more valid because the one from the virtual extension will take place.

A virtual extension can be used for Hot Desking, where a single physical phone is used by multiple workers, in this case you need to create a virtual extension for each of your workers and they will “assign” the phone they found on their desk to their own number using feature code.



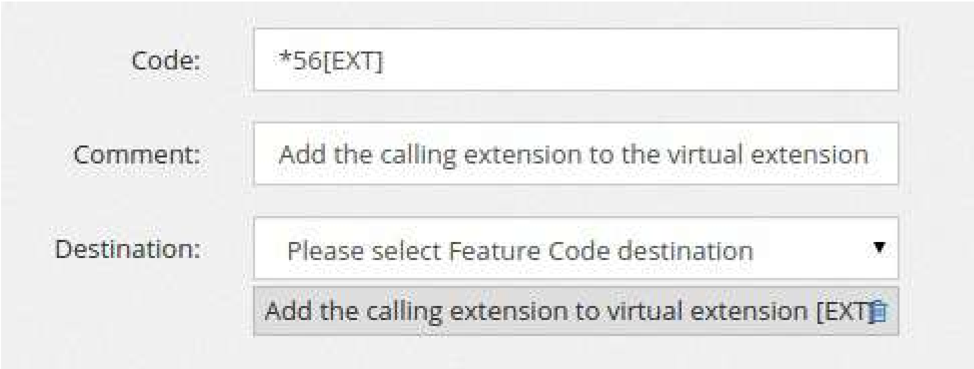
Let's make an example: you have three employees and just two desks, but they will be never work all together, so this will be a perfect case for Hot Desking. Each employee has a virtual extension, so for example:

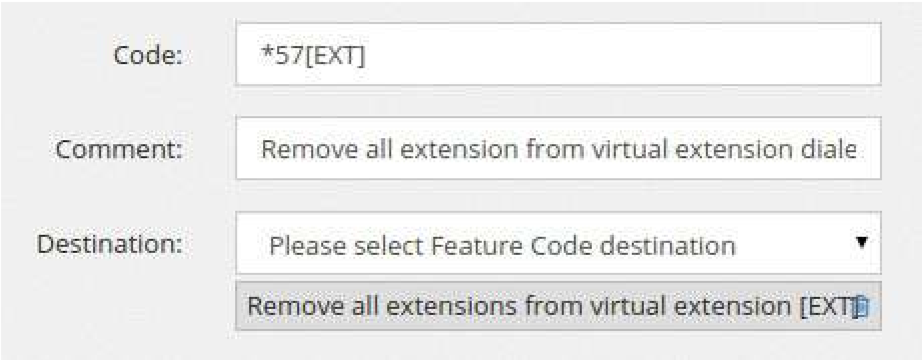
Bob, virtual extension 401

John, virtual extension 402

Mary, virtual extension 403

On your desks you'll configure the phones, so you'll have device 250-OIL on the desk near the window and the device 251-OIL on the desk near the door. You have defined two feature code, *56[EXT] for assigning the device to the EXT virtual extension and *57[EXT] to remove.





Bob is coming in the office and he sits on the desk near the window. He “assign” the device 250-OIL to his virtual extension, so he lift the receiver and dials *56401. From now on, the virtual extension 401 will have the device 250-OIL. Mary is coming and do the same, but from the 251-OIL extension. Now, dialing 401 will ring the phone on the desk beside the window and dialing 403 will ring the one near the door.

It is lunch time and Bob leaves the office for a break. He removes his phone number from the extension dialing *57401. Mary has instead finished his day and goes home, she removes his number from the phone too, dialing *57403.

John arrives in the office and both desks are empty, so he sits in the one near the window and assign the phone to his virtual extension, by dialing *56402.

Bob is back and the only desk available is the one near the door, so he assigns his virtual extension to that phone, by dialing *56401.

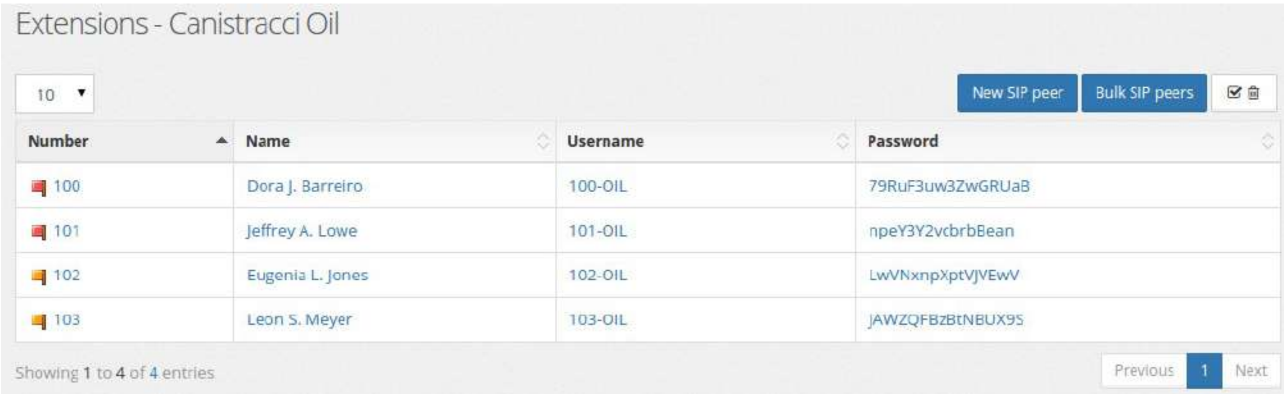
Now, dialing 401 will ring the phone on the desk near the phone and dialing 402 will ring the one near the window. Dialing 403 will go to the “On Offline” Additional Destination for Mary number.

Delete of Extension

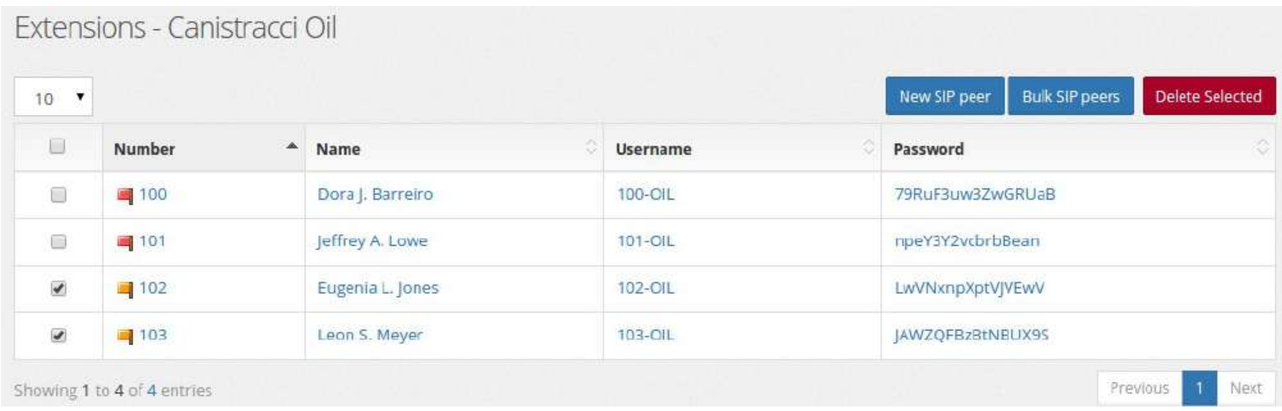
To delete an extension, it is enough to just press on the delete button at the end of the extension definition. A message will request confirmation. Deleting the extension will unregister and clean it from the asterisk peer cache, denying any other operation for the deleted extension.

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.

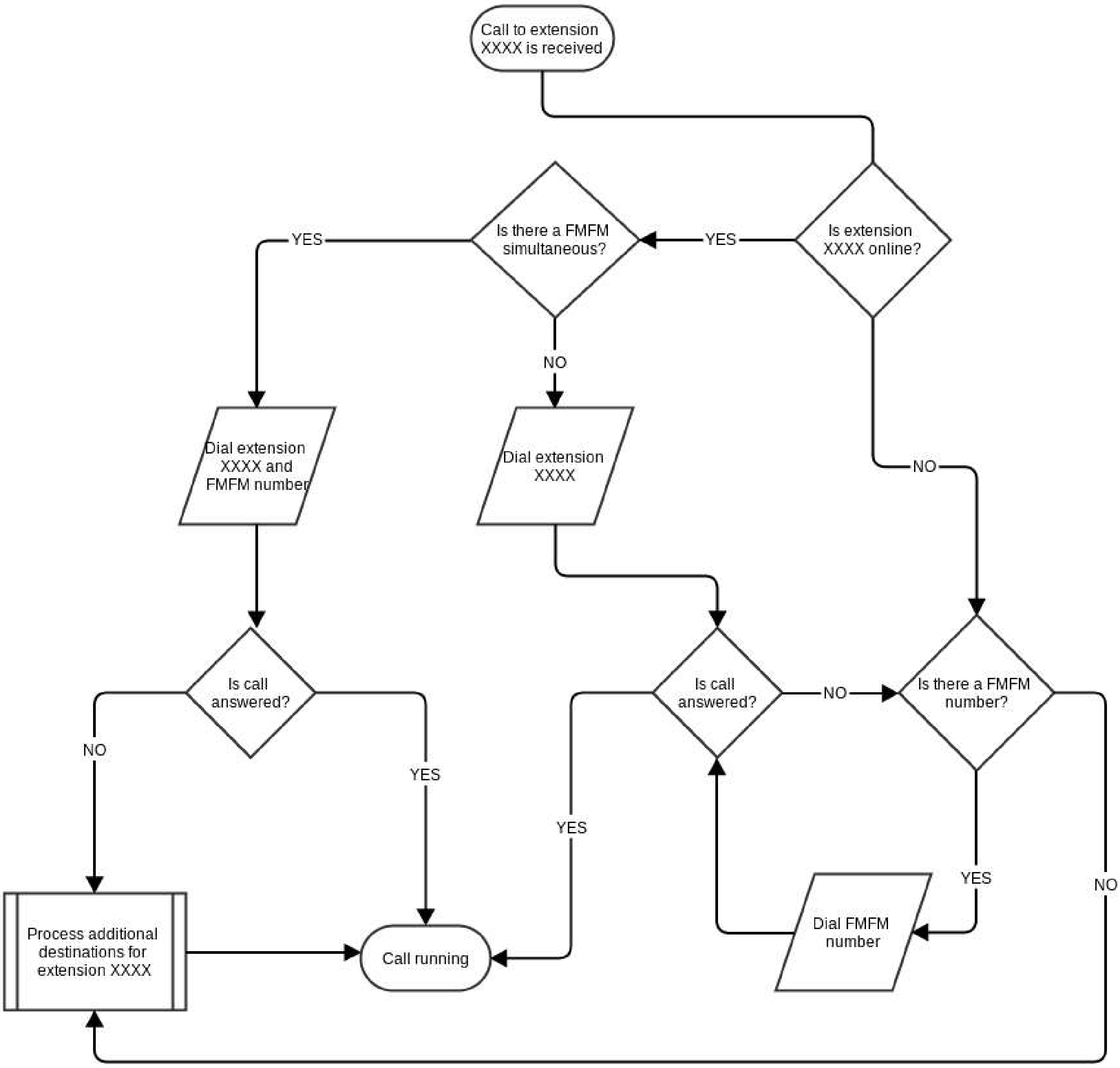


When pressed, a new column will appear on the left, allowing to select the extensions to delete by pressing the newly appeared button “Delete Selected”.



How dialing works

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



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

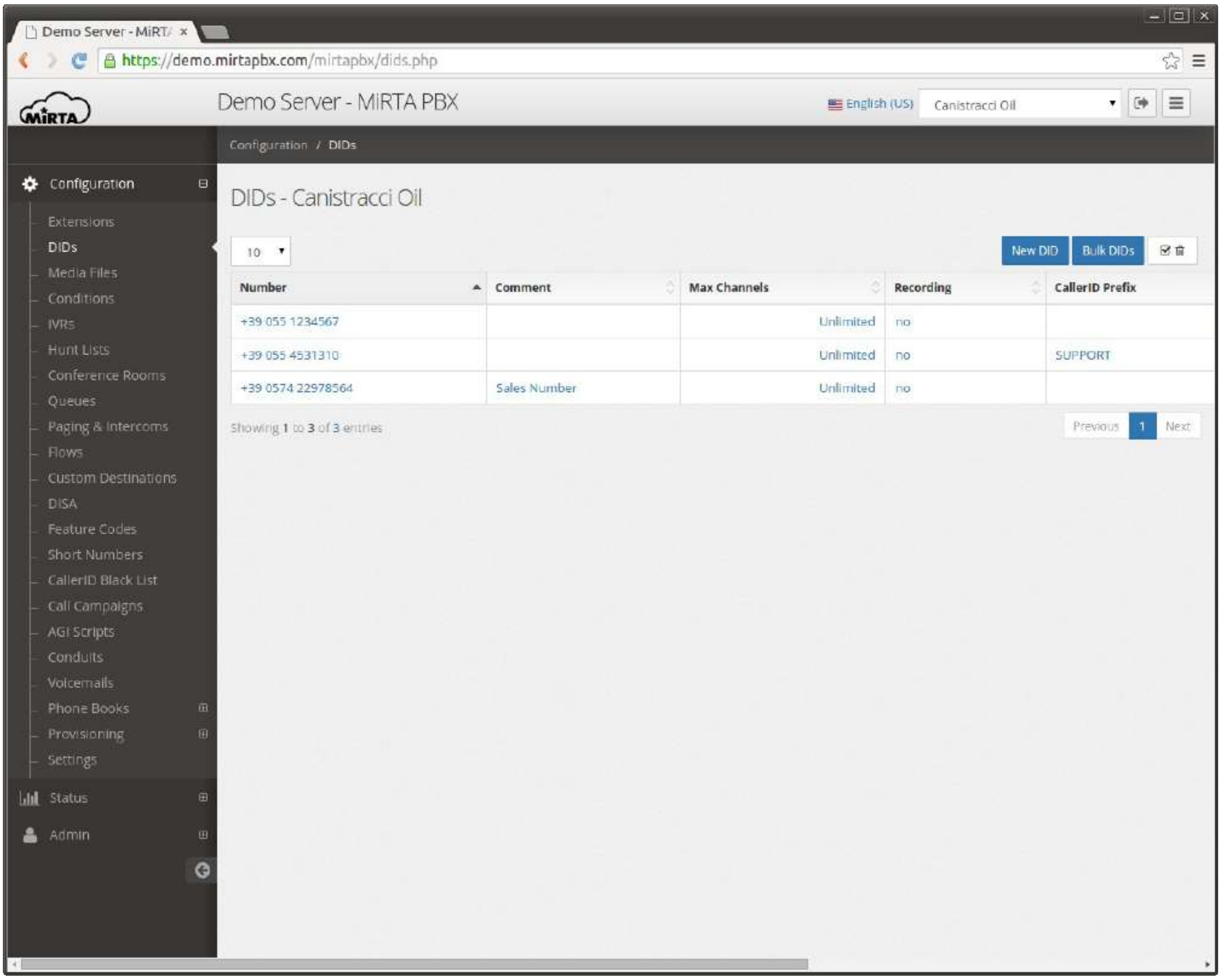
Online Extension with FMFM number defined and with an additional destination to Voicemail for the No Answer

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

DIDs

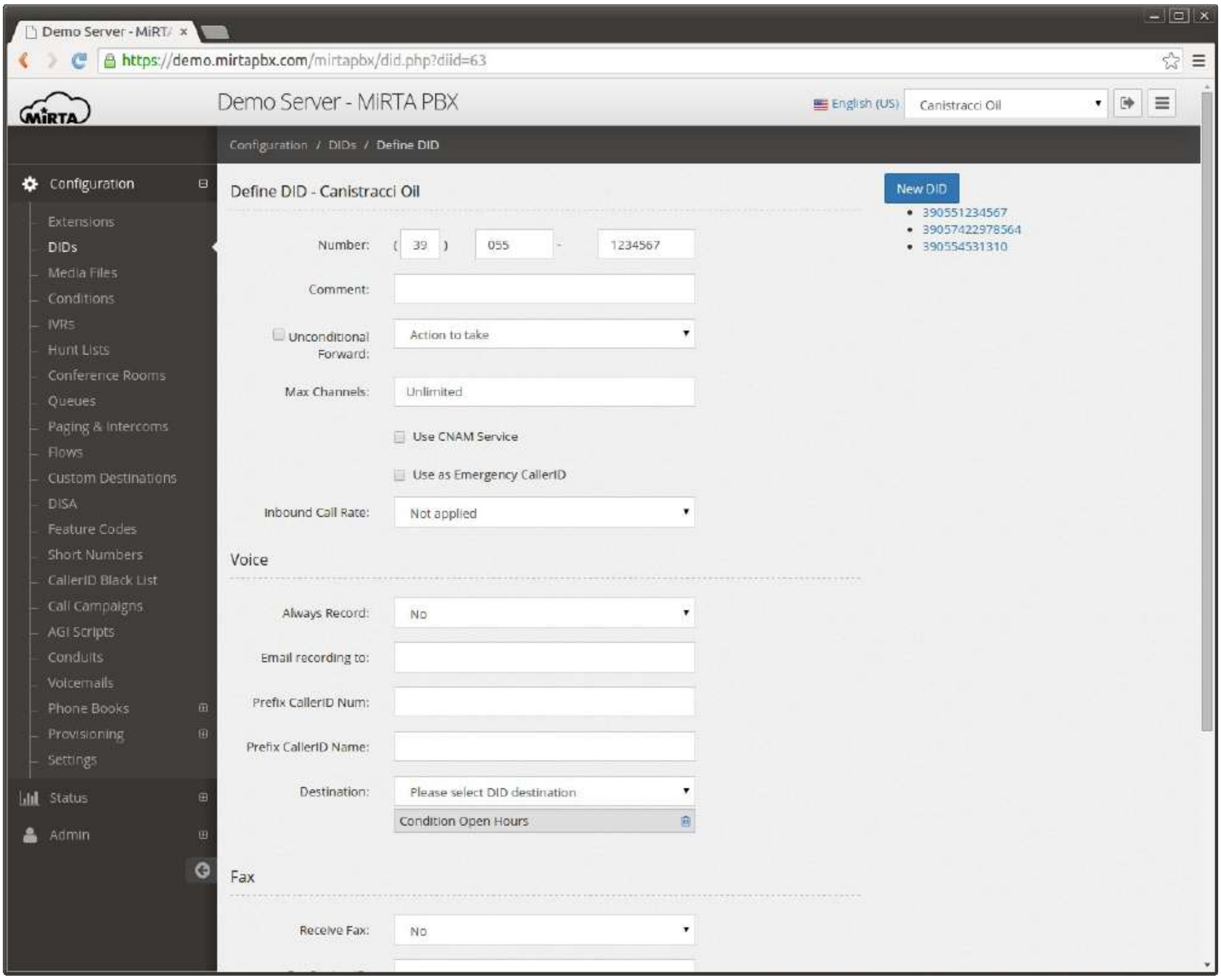
To every tenant can be assigned a group of DIDs, numbers to be dialed from the outside. Each DID can be configured to reach a certain extension or Queue or Hunt Group or IVR or any combination of the elements defined for the tenant. The same DID cannot be assigned to more than one tenant and there is no correlation between the dialing profile assigned to a tenant, the trunk provider used and the incoming DID. For this reason is highly important the incoming INVITE from the trunk provider contains all the info to identify correctly the DID it is addressing.

Due to the fact there is a lack of strong standardization among trunk providers on how to transmit the DID information, a “best guess” is used to identify the correct DID, using any of the format currently in use around the world.



New/Define DID

A DID can be configured to accept voice, fax or try to guess the calling party (if voice of fax). Autodetection relies on signalling 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.



The number must be entered in three parts, starting with the international prefix, the area code and the number. Even if your telephone standard doesn't allow the usage of any “short” form for dialing the number, requesting for example, to dial always the area code, the number is requested to be entered this way. Using the Admin/Settings menu is possible to enable the use of a “single box DID” to just enter the number in a single box.

It is possible to use regular expression inside the number. In case multiple regular expressions match the number dialed, the ordering is based alphabetically on the Comment.

Comment is just a comment and it is not used in any way.

Unconditional Forward is a destination that can be set on the DID to send the call to a particular destination. It is enable/disabled by the checkbox and 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 CallerID permits you to mark this DID as one of the available to be used when an emergency call is dialed. Due to the fact the emergency numbers are not standard among 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 to define how to route the call when a voice call is received.

Always record permits to force the recording of the call, whichever is the future of the call. Recordings will be available through the Call History menu.

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

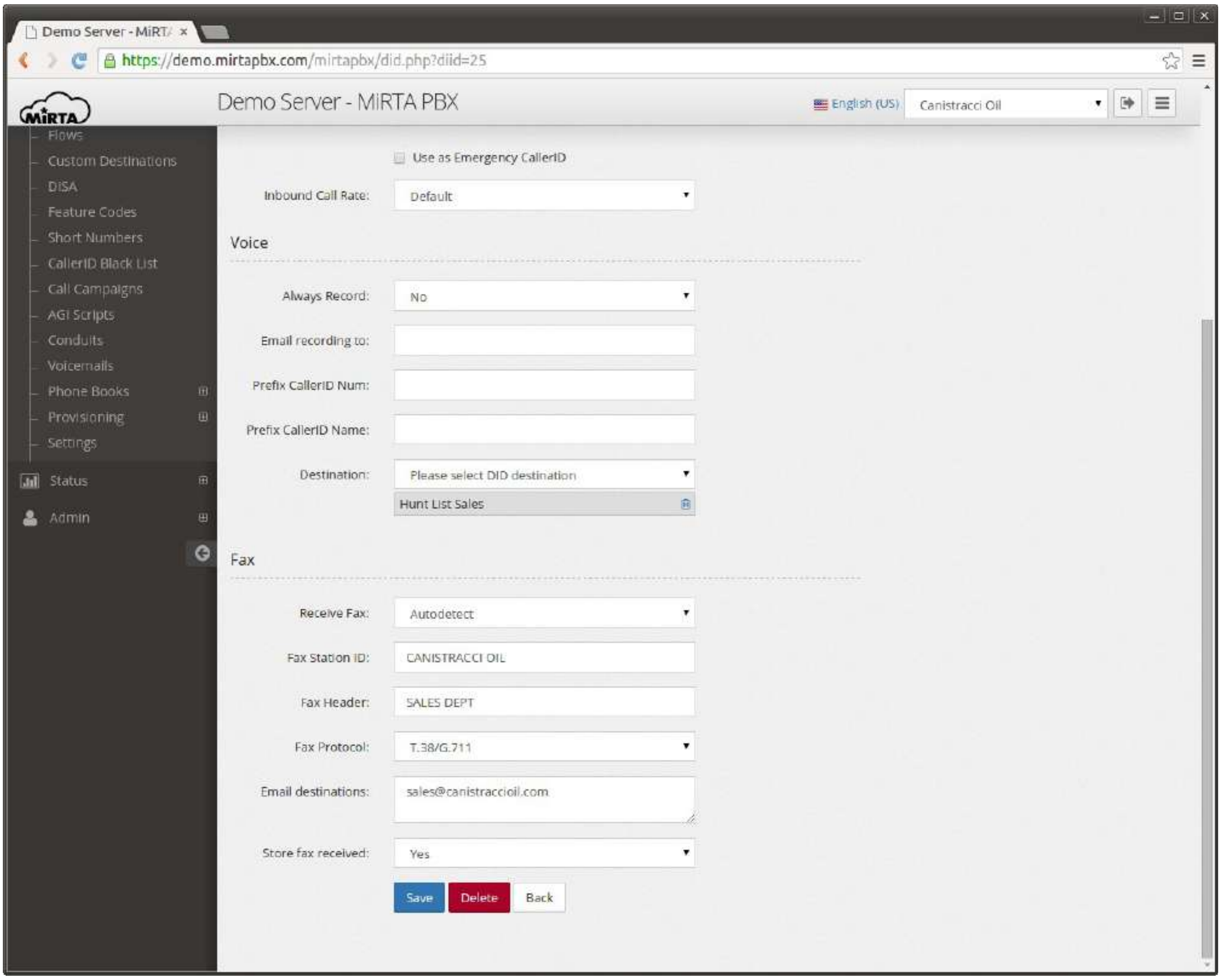
Minimum Size (Bytes) allows to receive those recordings bigger than the size set, in Bytes. Recording takes place only on bridged channels, so IVR prompts or Music On Hold will be not recorded.

Using the **Prefix CallerID Num** is possible to define a string to be added to any CallerID 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 allow to define the list and the order of the objects receiving the call.

Fax section allows to define 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.



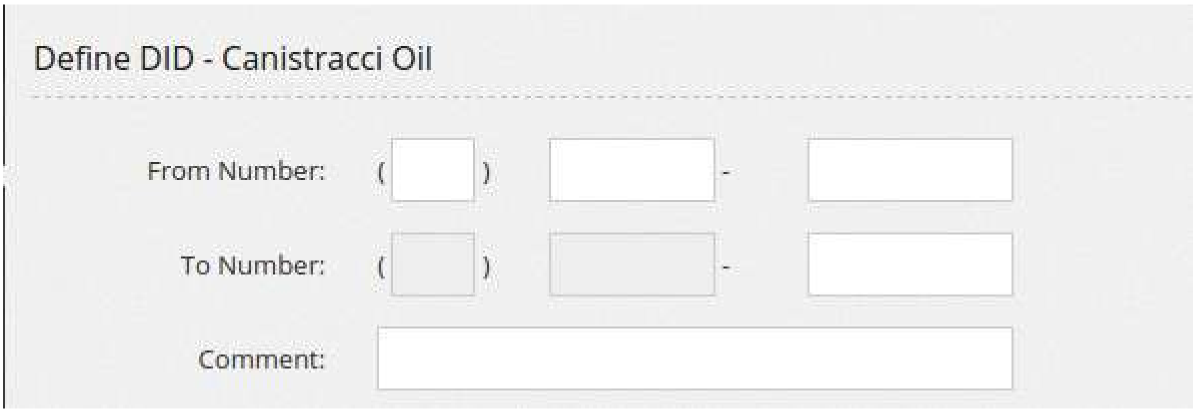
Fax Station ID and **Fax Header** allows you to customize your virtual fax.

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.

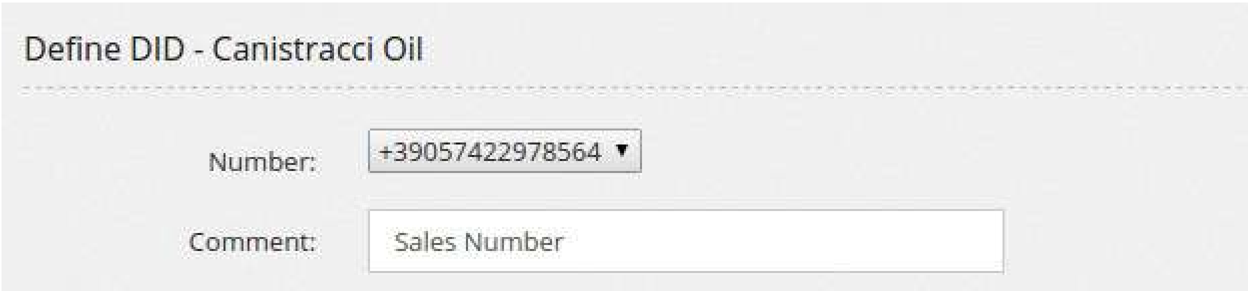
Email destinations box permits to enter one or multiple comma delimited email address to forward the received fax. Received fax are sent in PDF format. Partially received faxes are sent in the same way.

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

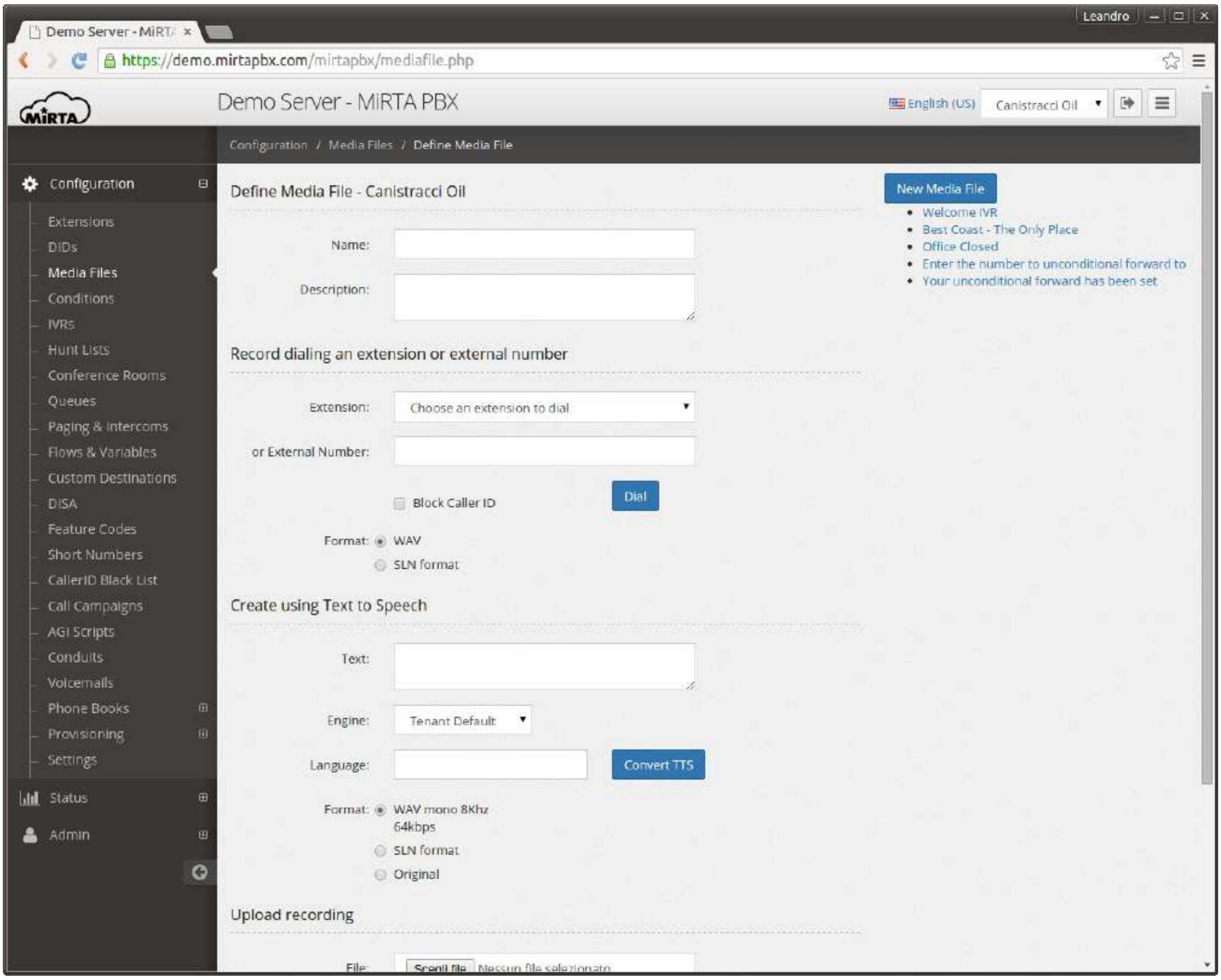
Bulk DIDs creation



Use DIDs storage

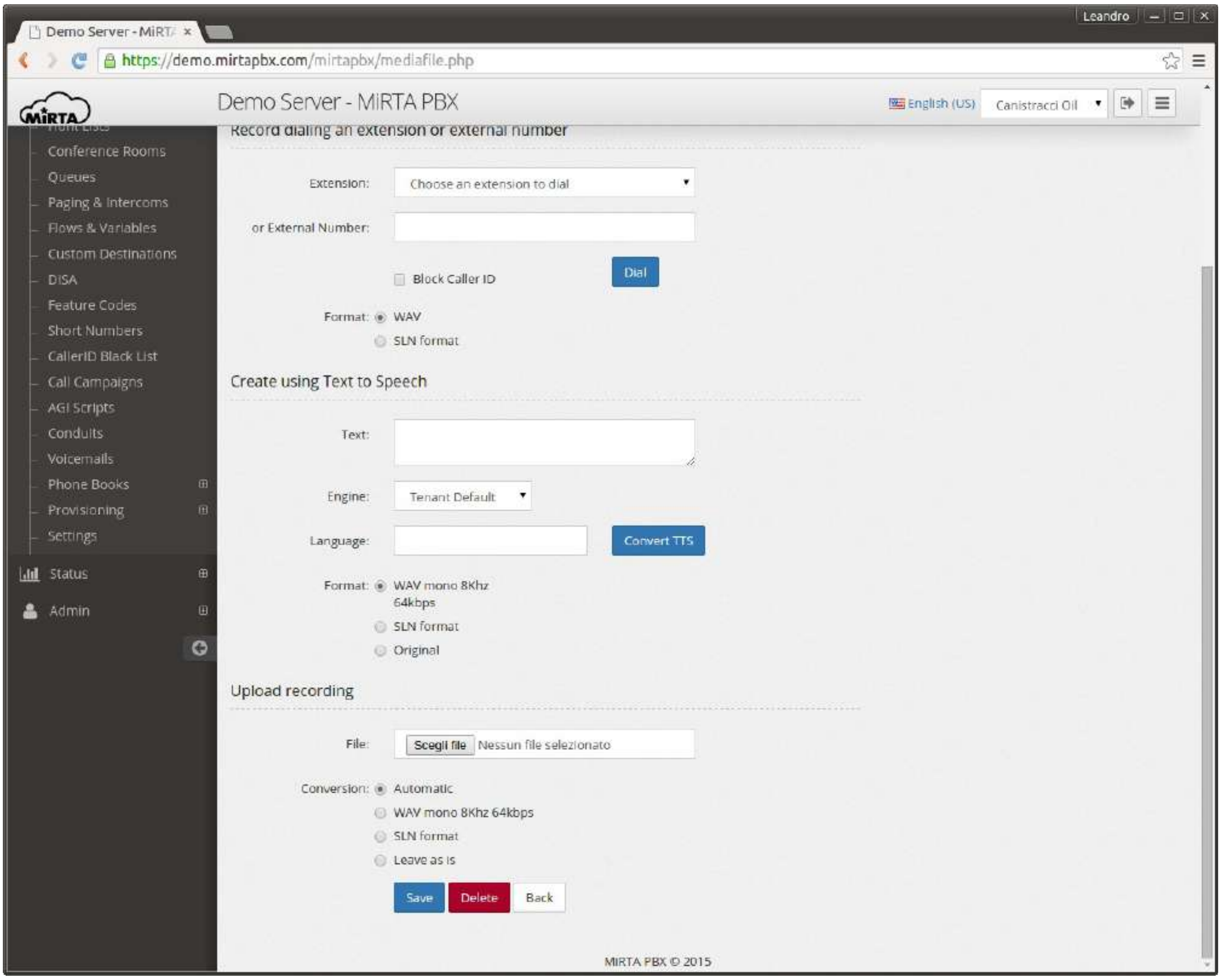


Media Files



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). Unfortunately it seems not available any player for MS Windows able to play the slin format, so if you plan to convert your media files to slin, you'll be not able to listen to them from the web interface.

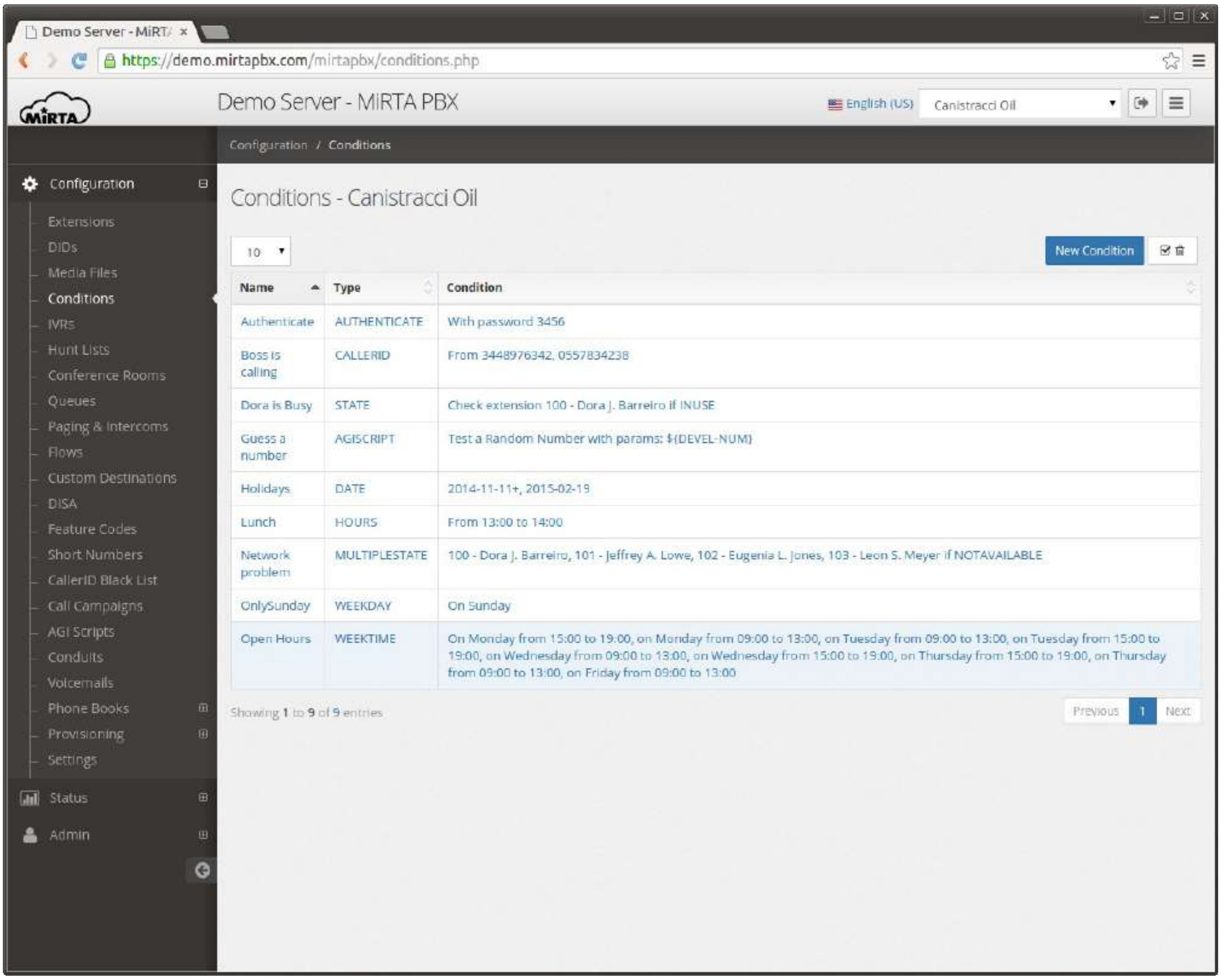
It is possible to dial a number, internal or external and have a message recorded. The call will come with the number to dial as caller ID, so if your phone can have some problem receiving a call from its same number, check the “Block Caller ID” to hide it.



It is possible to upload a file and convert it to the desired format or leave it as is. Please note not any kind of wav file can be played, so if unsure, leave it on “Automatic”.

Conditions

Conditions allow to manage the call flow, playing for example a different messages or routing the call based on hours, or days, or calling party. Conditions can also apply to extension state or user input.



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 reach directly the support staff without waiting in the Queue.

Weekday allows to route calls based on the day of the week

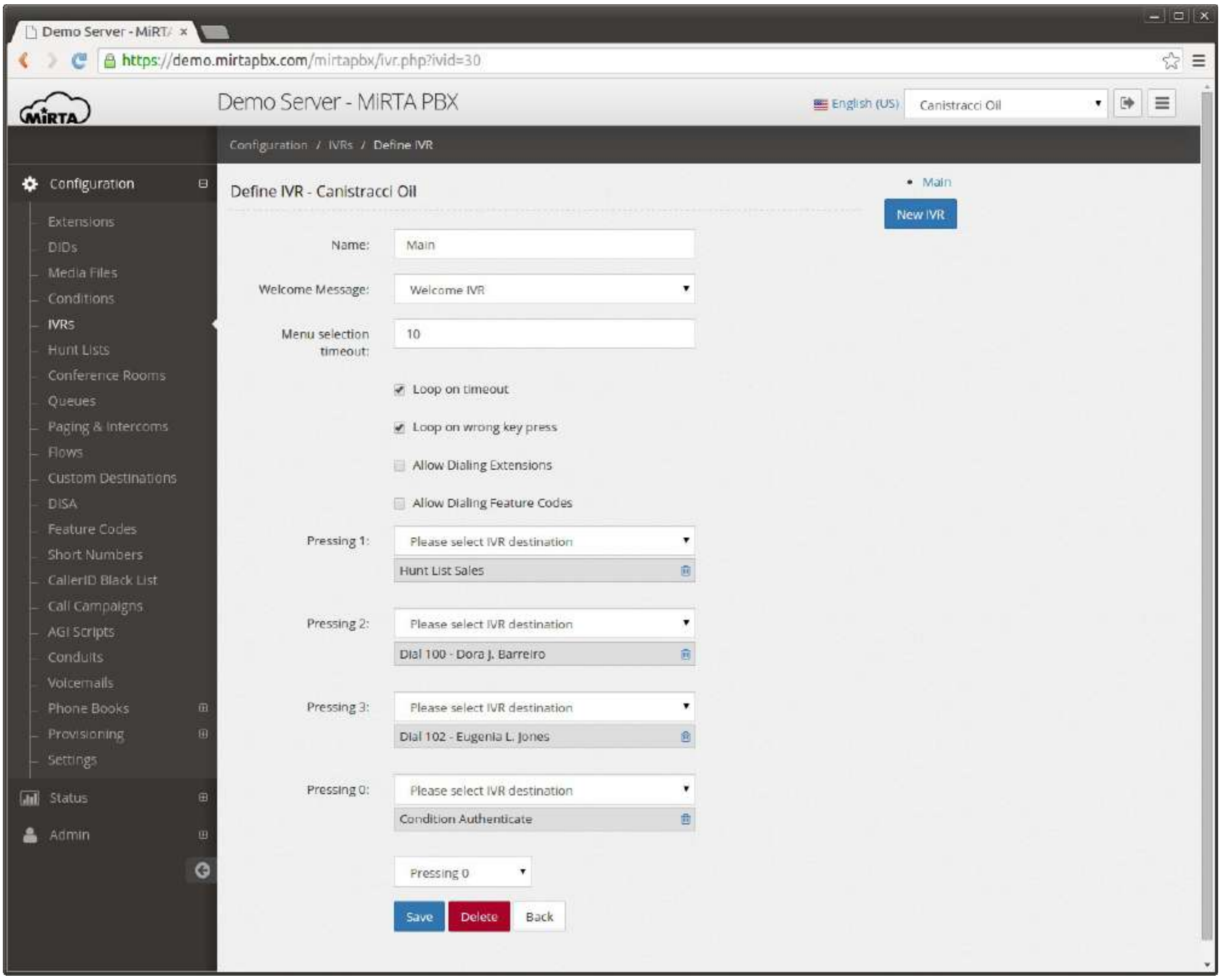
Date permits to route the calls based on specified dates. A date, like Christmas, can be made “recurring”, so it will trigger every day, regardless the year.

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 allows to define 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 allows to determine the amount of seconds to wait before considering the number entered as "complete"

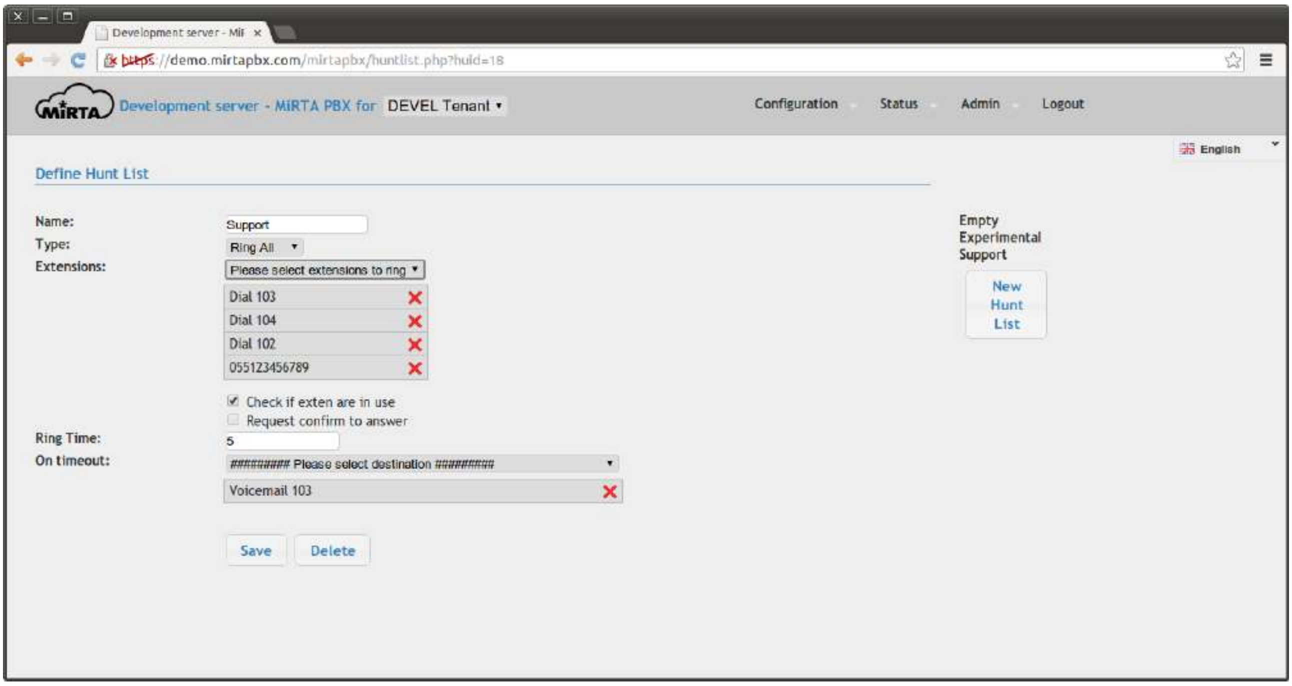
Loop on timeout permits to continue to play the welcome message and to wait for the selection every time the Menu selection timeout expires.

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

Allow Dialing Extensions permits to the calling user to dial directly an extension instead of

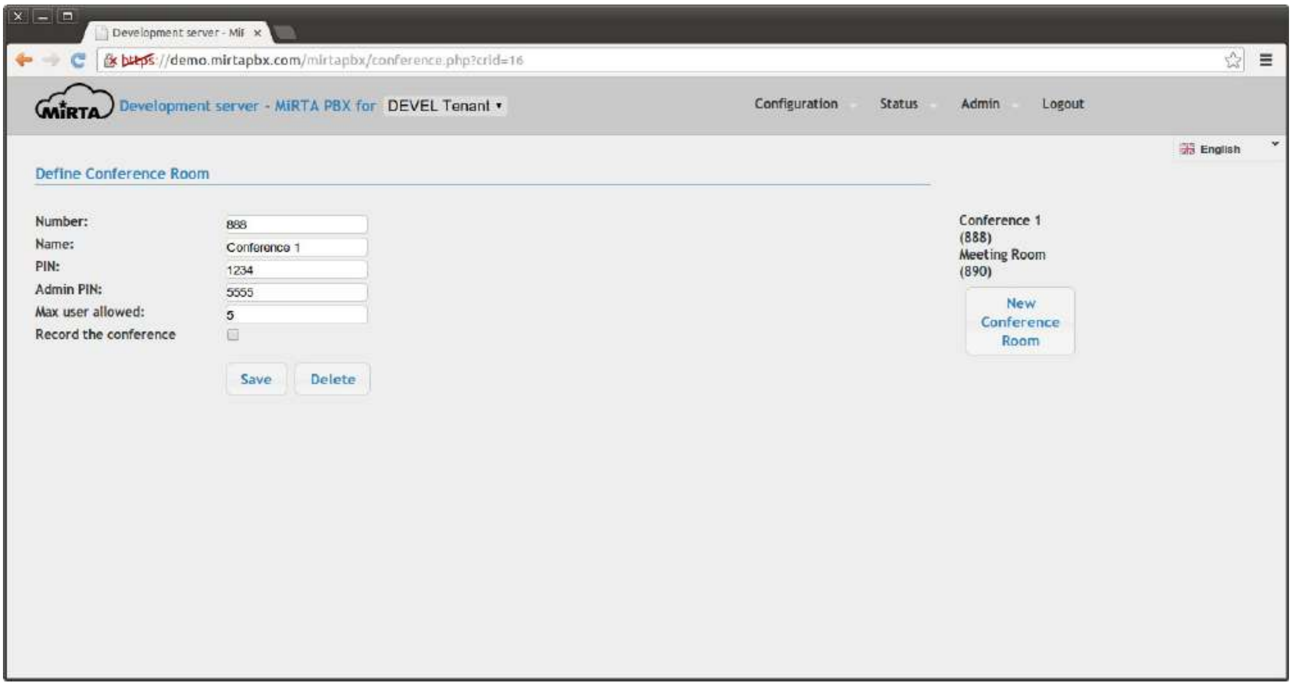
picking one of the digits.
Allow Dialing Feature Codes, like above, but for feature codes.

*Hunt List**



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 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 to specify the time each extension or external number has to be dialed before skipping to the next item.

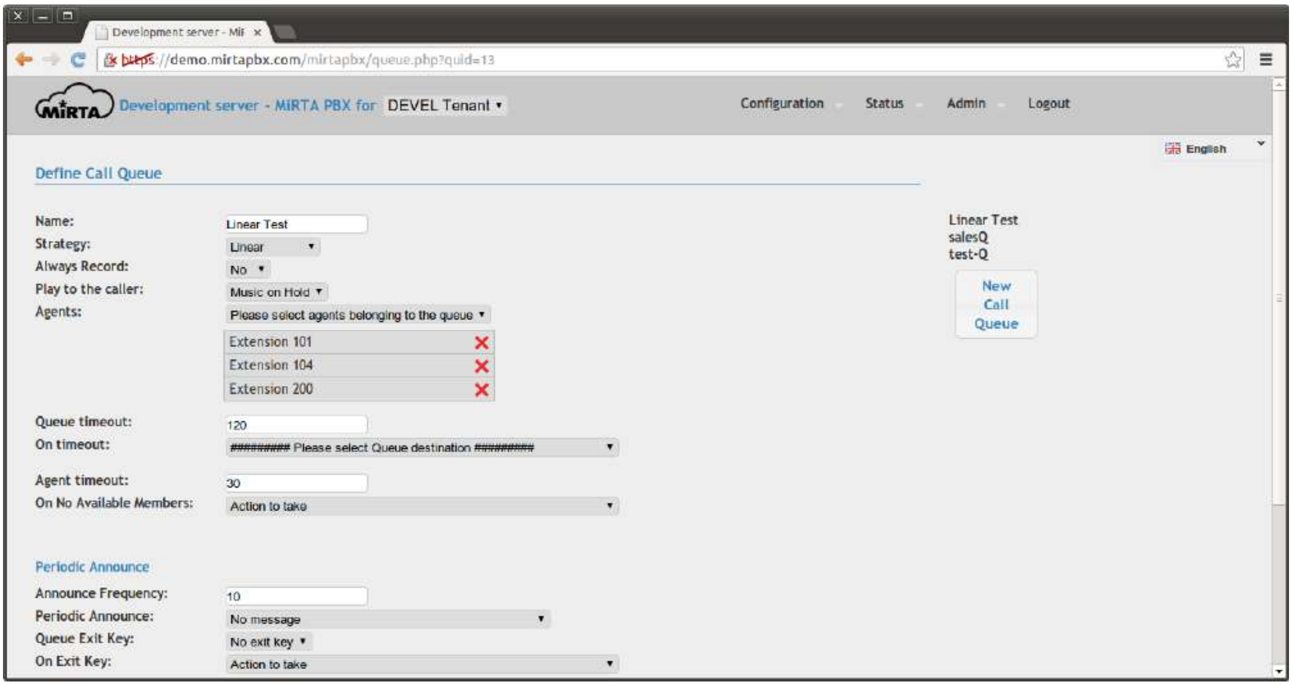
*Conference Rooms**



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

*Call Queue**

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

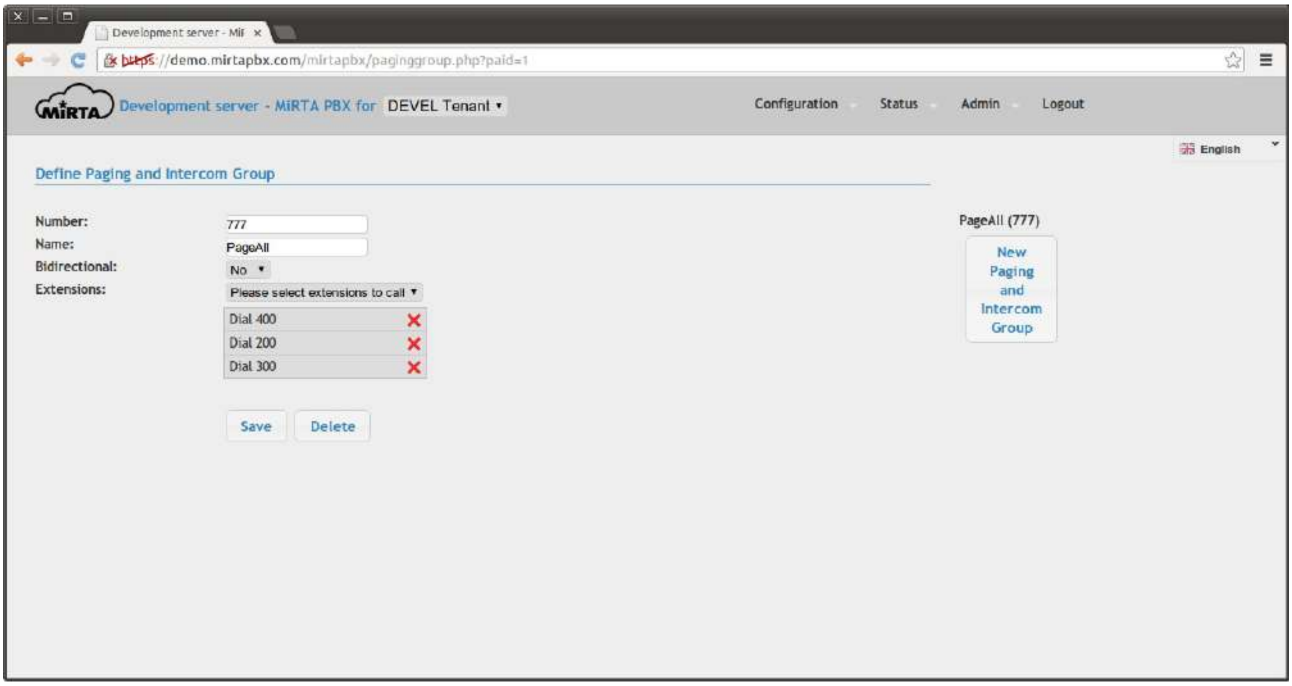


Ring All – All agents are rang at the same time
Round Robin – agents are rang, one at the time, in a round robin way
Random – agents are rang in a random order
Least Recent – The least recent agent is rang
Fewest Calls – The agent with the fewest calls is rang
Linear – agents are rang in the order specified. Due to some asterisk limitation, it is not possible to change the Strategy of a already defined Queue to "Linear". The queue needs to be destroyed and recreated.
Always Record allows to always record the call. The call record is available trough the Call History.
Play to the caller permits to define if playing the defined Music on Hold for the tenant or a generic Ringing
Agents lists the agents in the queue. For each extensions, two kind of agent are available, the first is normal and the "following to A.D." permits to forward 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 ring 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 announce 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 hangup the call and being 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 redefines the messages usually played to the user choosing them between the media file uploaded

*Paging and Intercom**

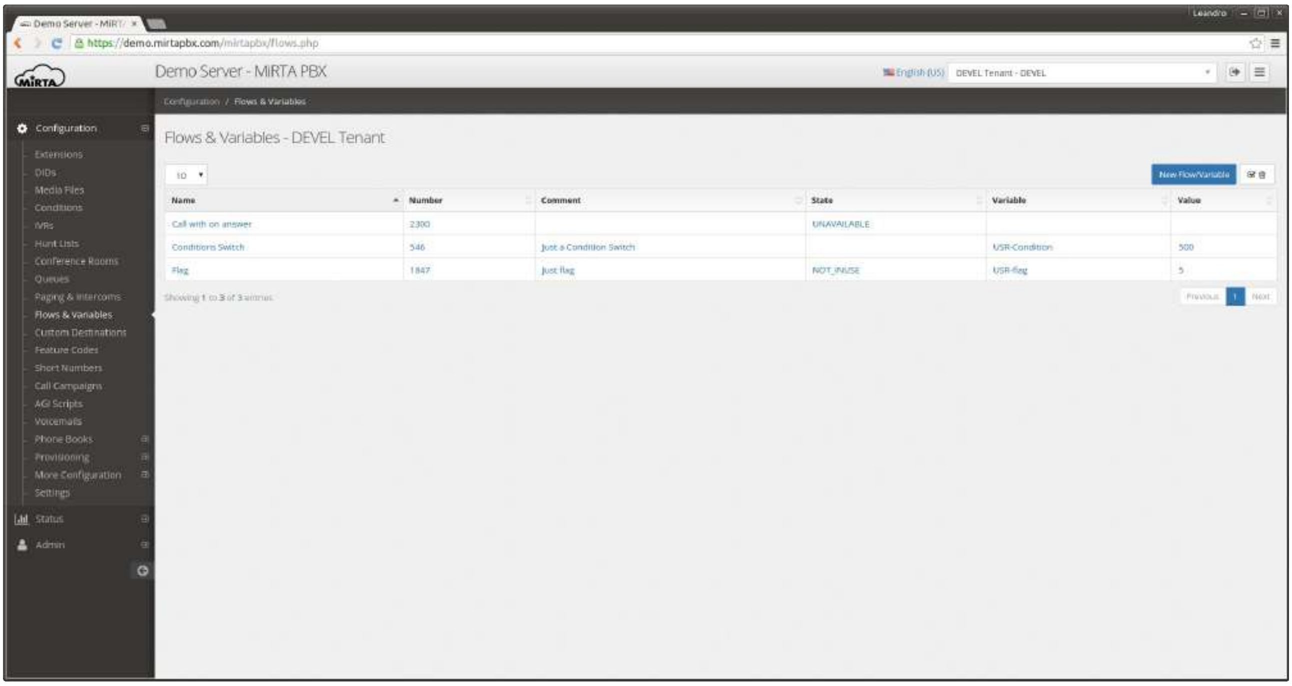
Almost all SIP phones allows to page them and use as intercom: the ability to establish a mono directional or bidirectional communication without making them ring.

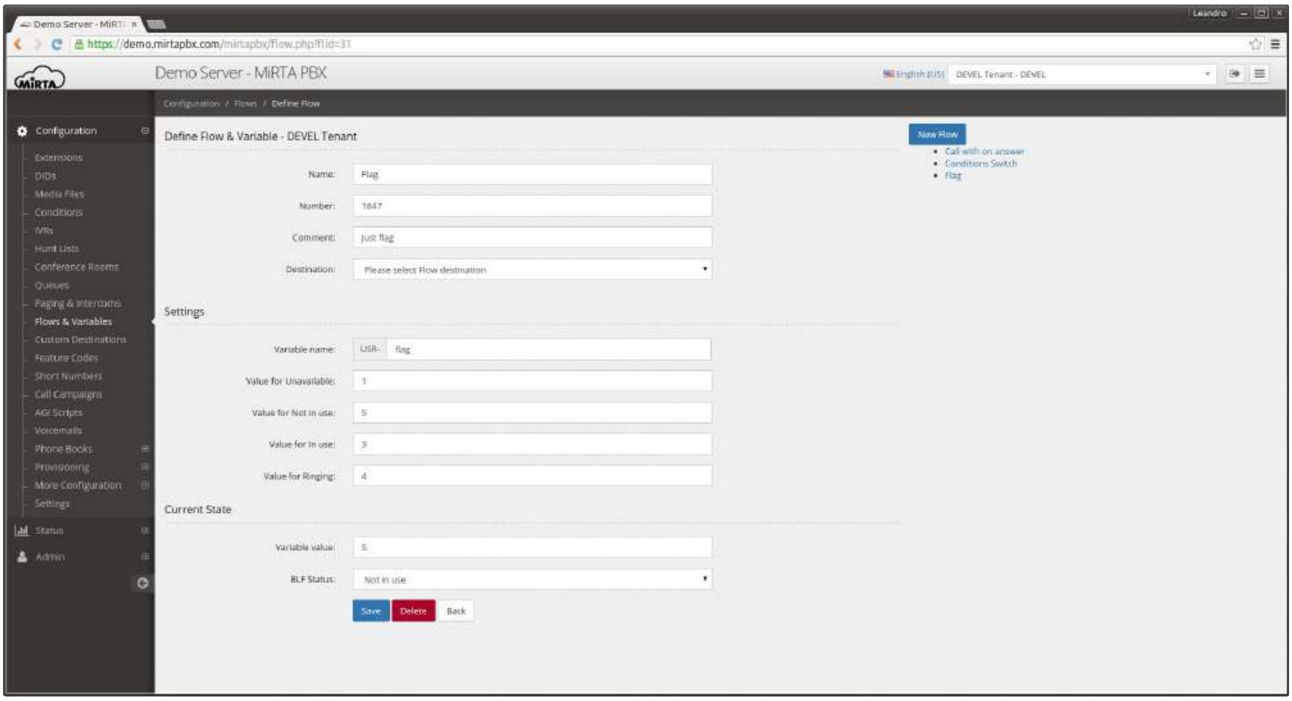


Some phones requires adjustment to the configuration to allow them to auto answer. For example, the Polycom VVX 300 requires the following setup:

Flow & Variables

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



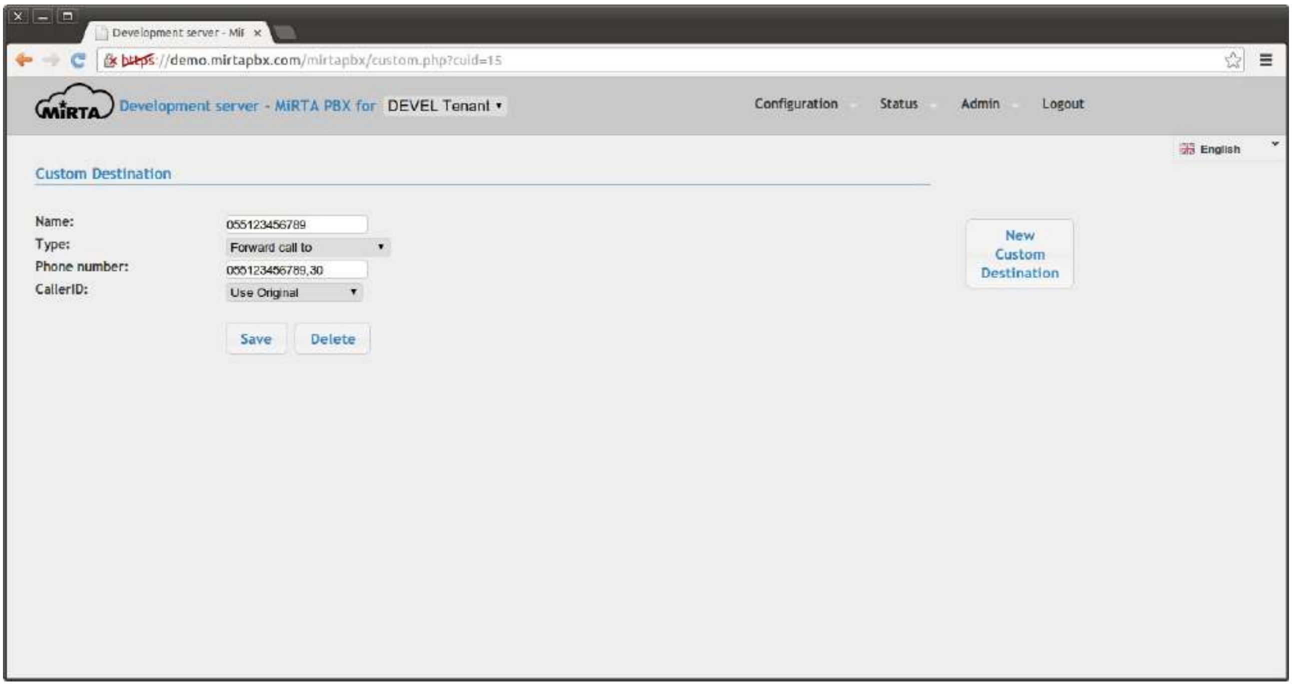


Assigned to each flow there is a variable that can be used to track a PBX status, like day/night or even something more fancy. The variable name can be referenced in the destinations, when allowed and several values can be assigned to each of the states, so the change in value will change also the status of the BLF for the flow number.

Check the “Condition Override” in the Setup Guides for an example

*Custom Destinations**

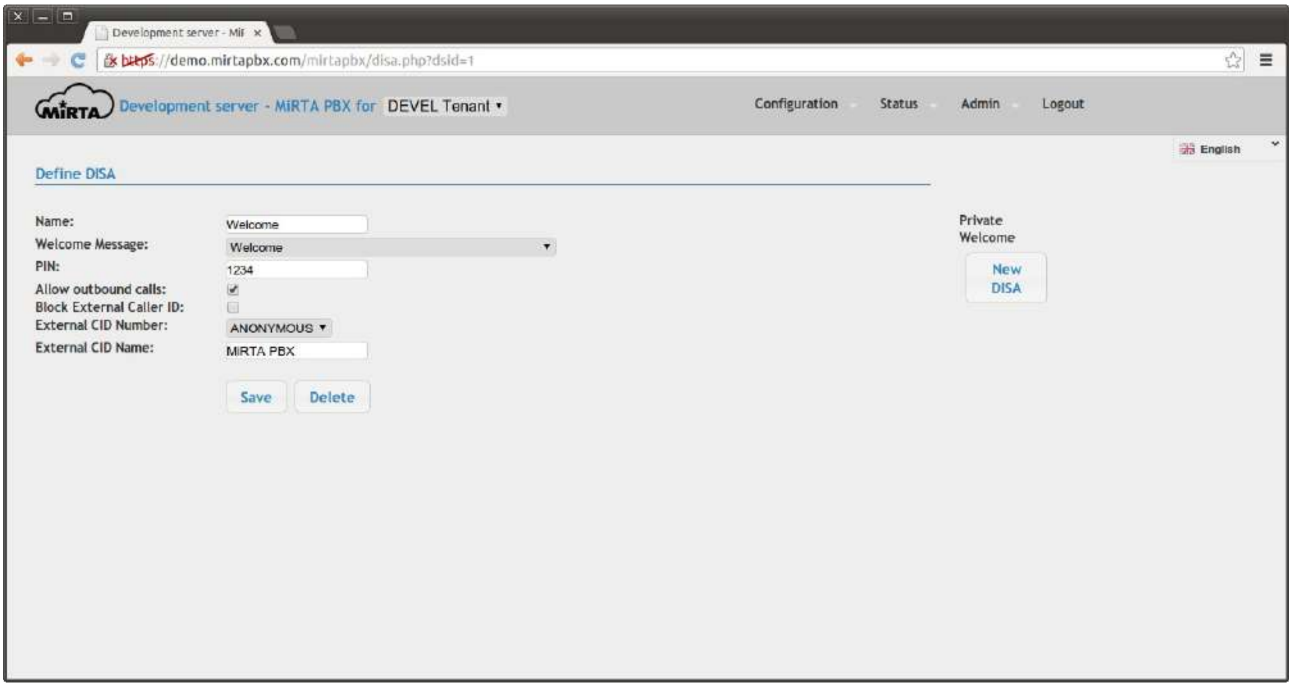
Custom Destinations allow the definition of custom destinations to be used in other Configuration settings.



Forward call to – will permit to 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 permit to 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 permit to 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 to 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 allows also to dial outbound numbers, usually protected by a PIN code.



*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 callerid on calling [NUM]” then if the number *625558764 is dialed, 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]

Feature Code	Description
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
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

Feature Code	Description
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 extension	Retrieve the voicemail of the calling 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

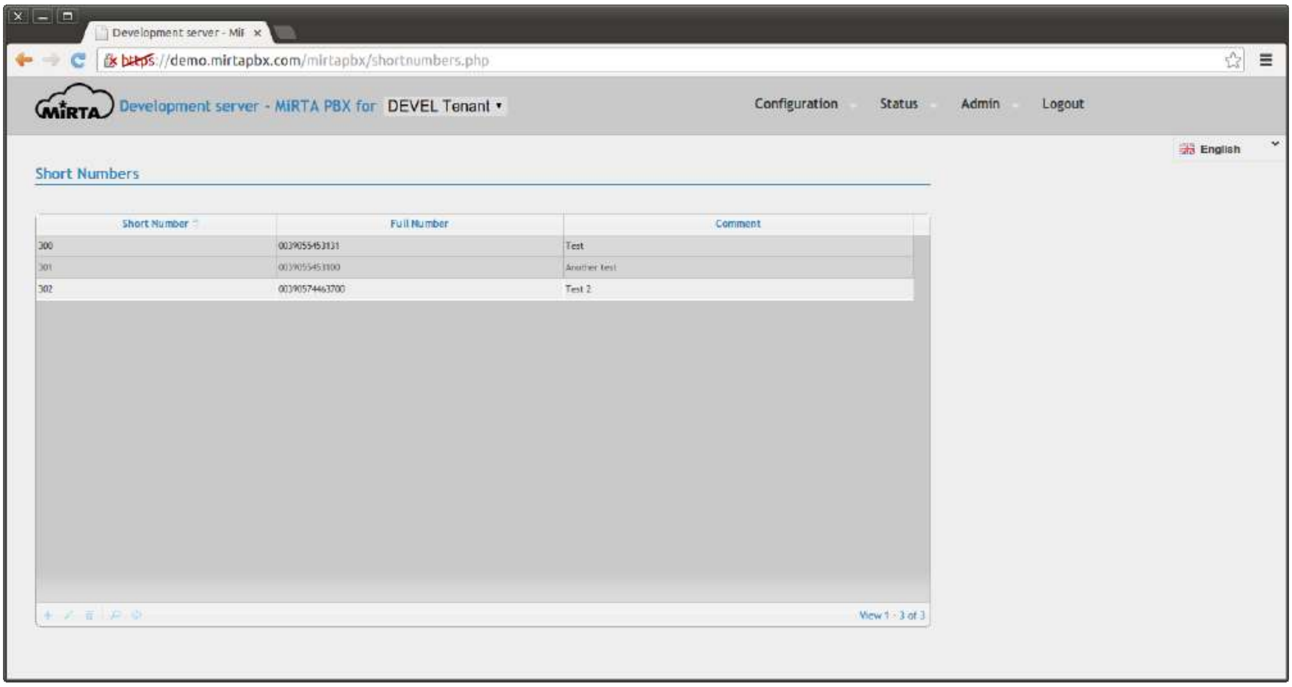
Feature Code	Description
	as Unconditional Forwarding.
Spy extension [EXT]	Spy on extension [EXT]
Toggle state of extension [EXT]	Change the state of the Extension [EXT] from IN USE to NOT IN USE and vice versa
Unmask the callerID on calling [NUM]	Remove the Privacy Mode while calling the [NUM]
Voicemail for Extension [EXT]	Retrieve the voicemail for extension [EXT]
Wait five seconds	Wait five seconds
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 object is 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]

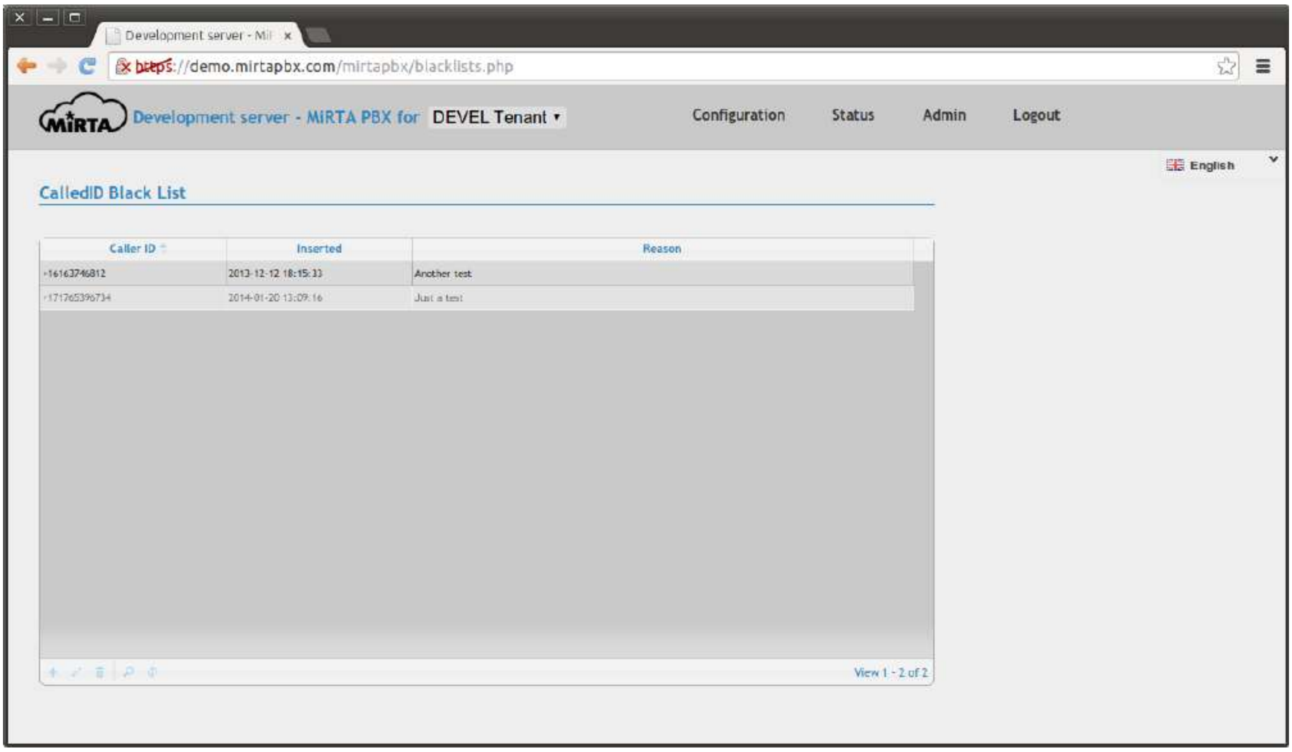
*Short Numbers**

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

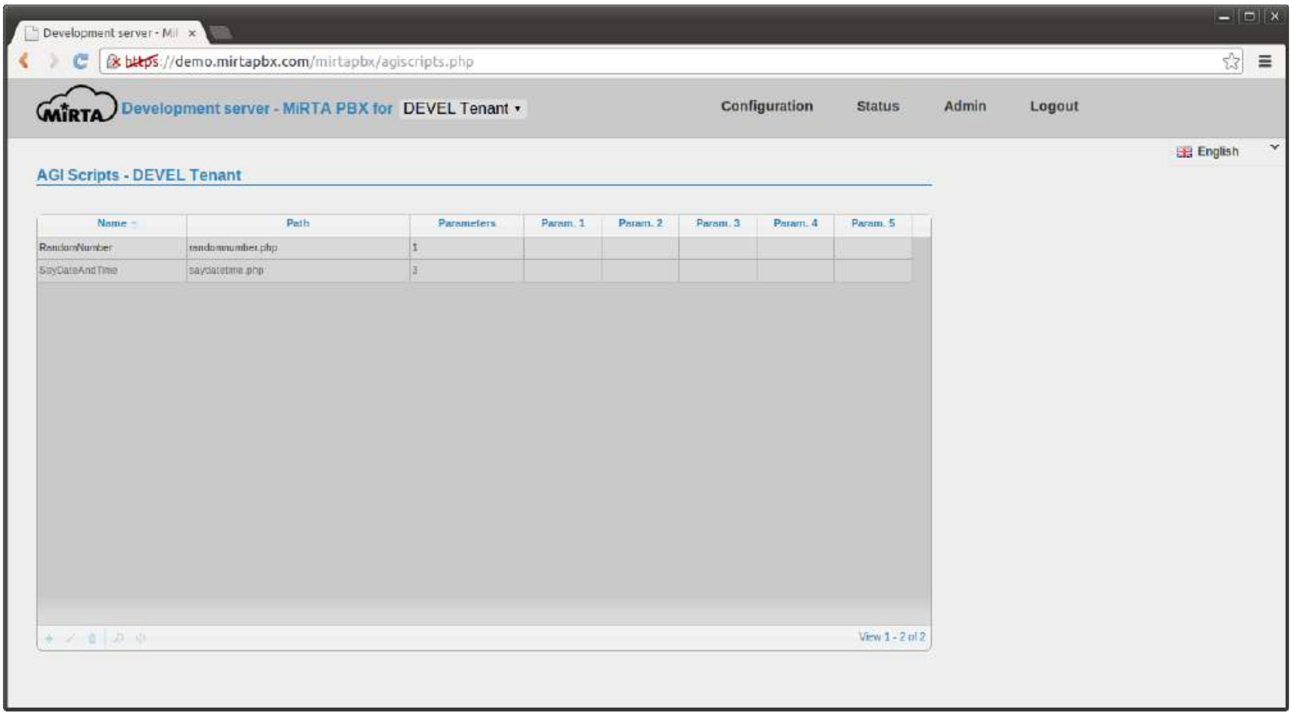


*CallerID Black List**

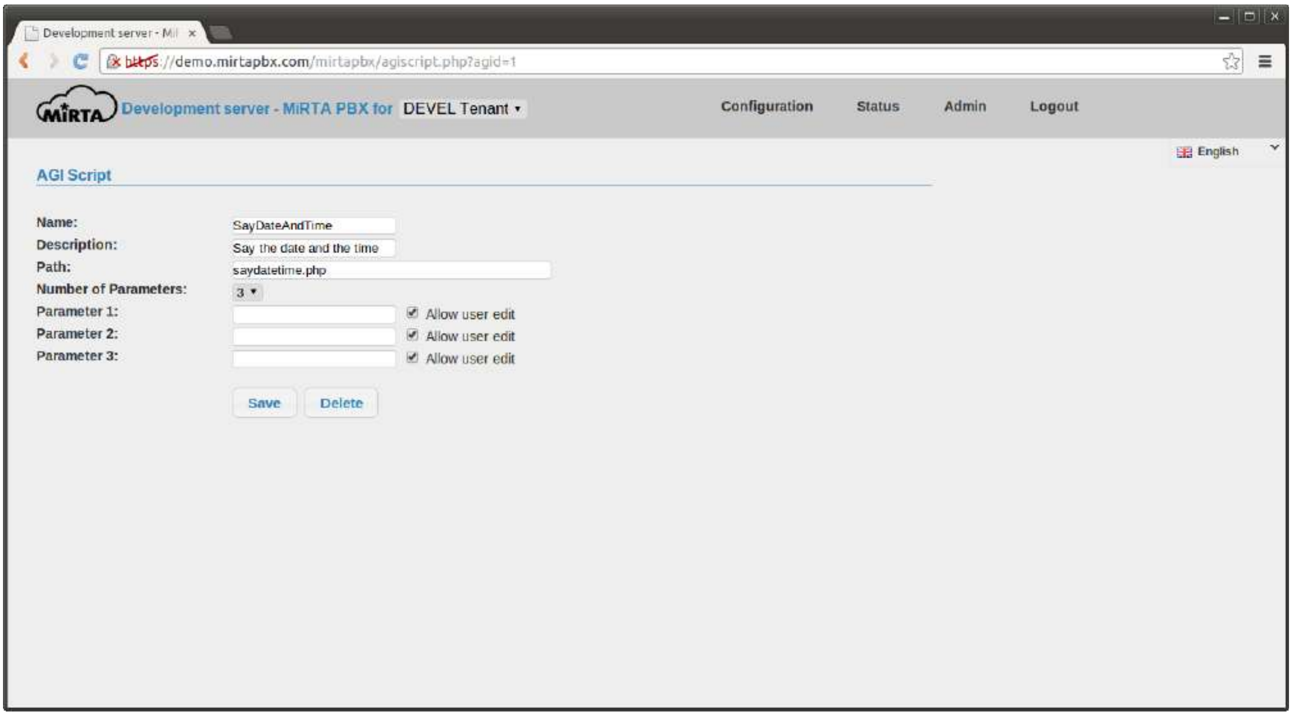
It is possible to avoid to receive calls from a list of caller ID by entering them in this list. Calls from those numbers will be hangup directly.



It is possible to use AGI Scripts as normal call flow elements or as conditions.



Each AGI Script can have at least 5 parameters and some can be assigned by the admin and others can be assigned by the user.



A simple AGI Script to say the date and time. Remember to set the execution bit.

```
#!/usr/bin/php

<?php

require_once('phpagi/phpagi.php');

$agi = new AGI();

$agi->exec('sayUnixTime',$argv[1].",".$argv[2].",".$argv[3]);

?>
```

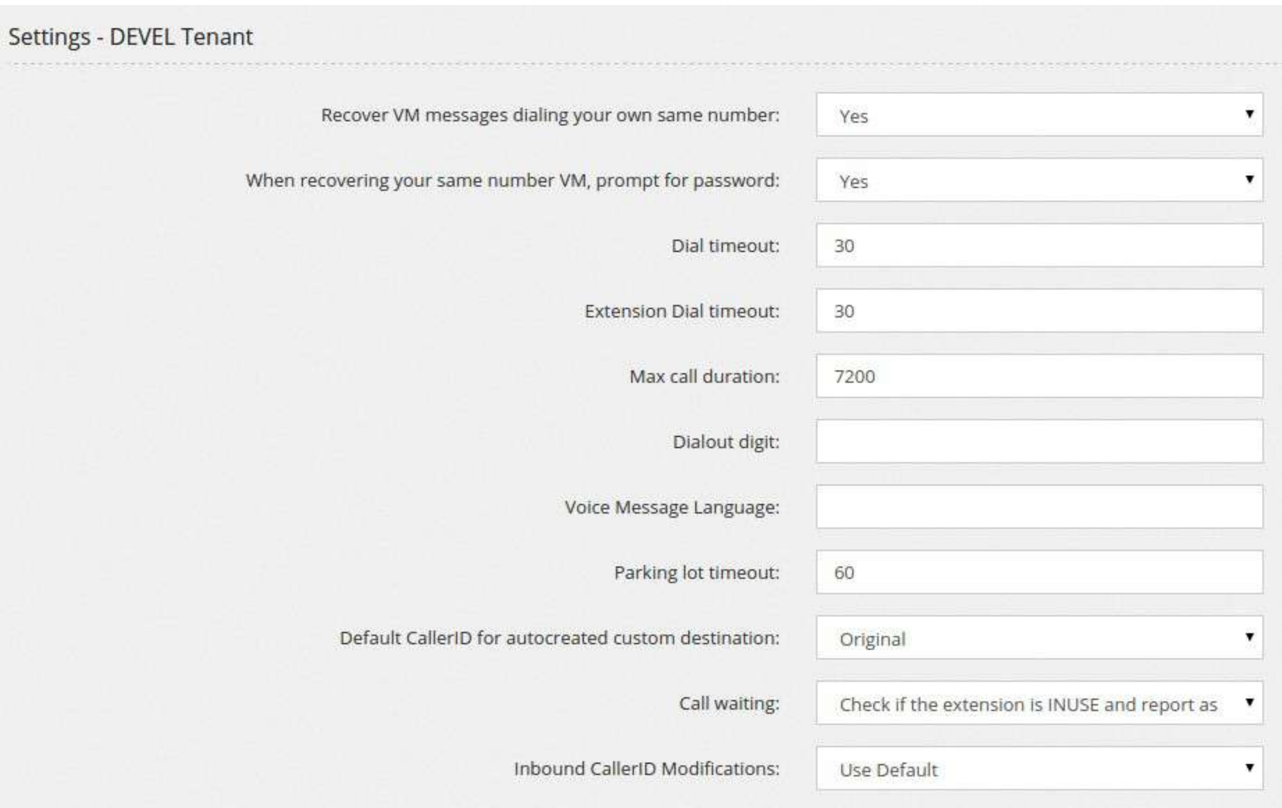
Another simple AGI Script randmizing a number and comparing with the one provided as parameter.

```
#!/usr/bin/php

<?php
```

```
$number=rand(1,10);

$agi = new AGI();

$agi->Verbose("Your guess is with the number ".$argv[1]);

$agi->Verbose("The random number $number as been selected");


if ($number==$argv[1]) {

  $agi->set_variable('AGIRESULT',"TRUE");

} else {

  $agi->set_variable('AGIRESULT',"FALSE");

}

?>
```

*Settings**

Every tenant can have its own settings. The settings page is divided in sections.



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

When recovering your same number VM, prompt for password permits to skip requesting the voicemail password.

Dial timeout identifies the standard time the dialing command will ring an external number or resource before reporting as “No Answer”.

Extension Dial timeout identifies the time the dialing command will ring an extension, so this is the time an extension will ring before going to the “No Answer” destination.

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

Voice Message Language lets you specify the default language to use in the Voicemail and all other voice messages.

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

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

Call waiting. Even if call waiting is usually set on the phone, rejecting or allowing the receiving of a call while online with another call, you can enforce also from the server, reporting busy an extension in use.

Inbound CallerID Modifications sets the CallerID modification to apply to all inbound calls.

Recording format. The recording format for monitoring calls can be choosen 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 to choose the files to use as Music On Hold for the given tenant. If no files are choosen 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, tenant code and secret key.

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

For example, getting the URL:

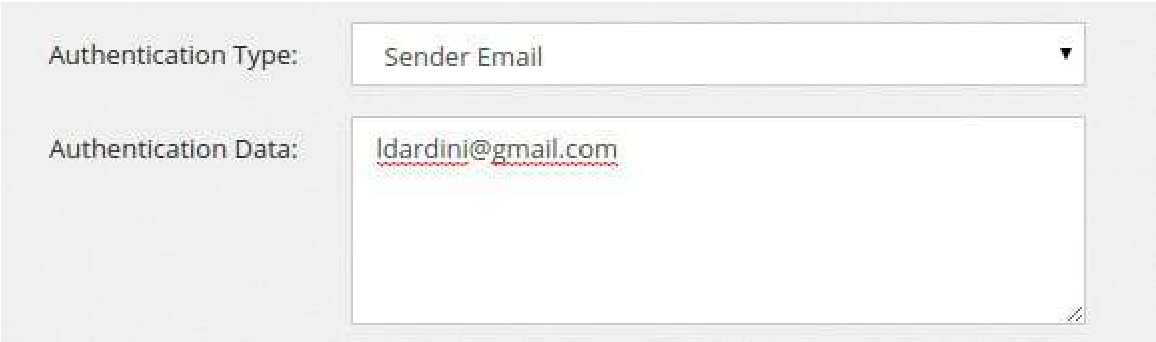
https://demo.mirtapbx.com/mirtapbx/webcall.php?source=104&dest=102&tenant=DEVEL&secret=H63JpSdPEWequMpr

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

Mail to Fax

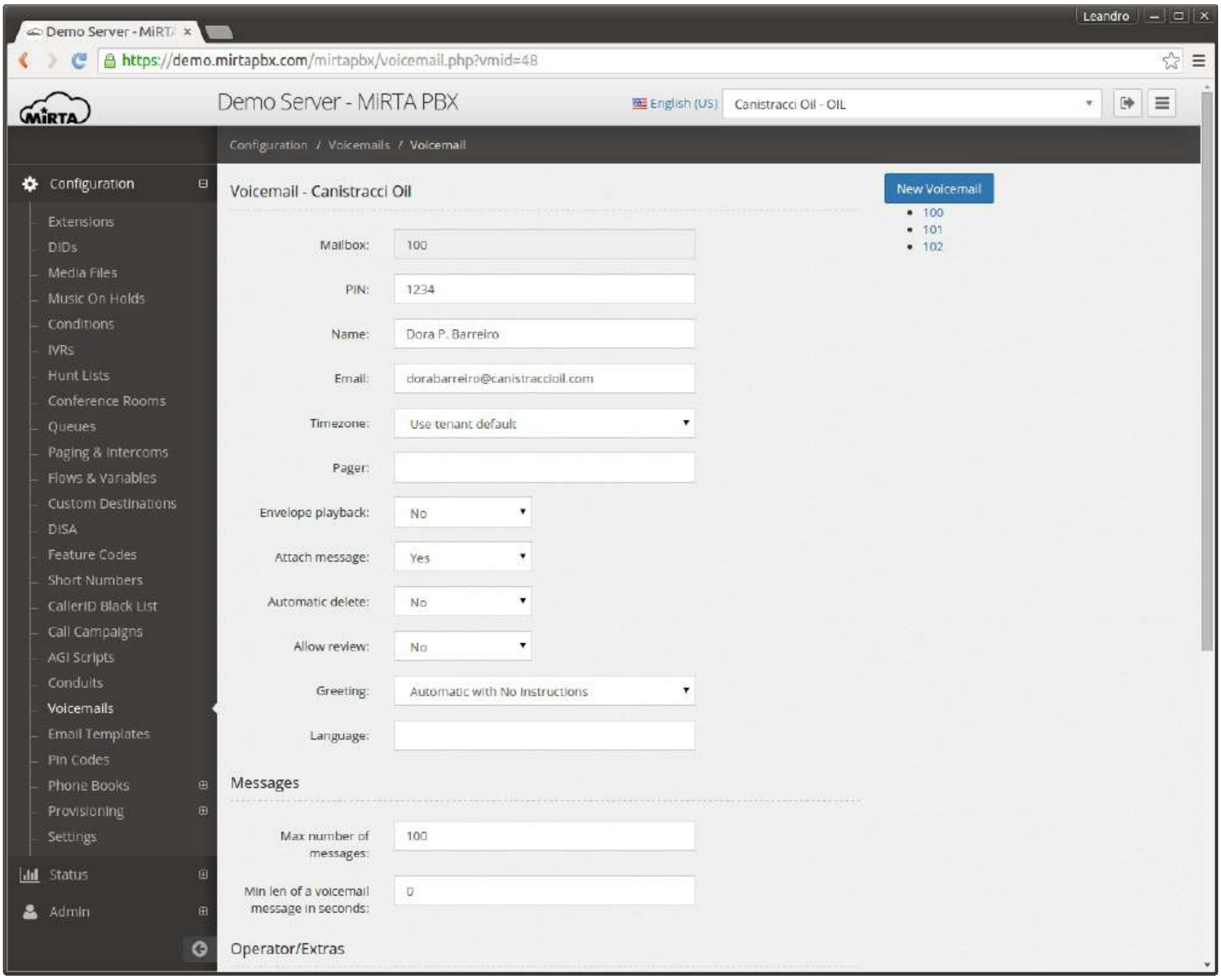
Mail to Fax permits to define one or more email accounts on your preferred hosting company, retrieve and authenticate the emails sent to them and use the attached PDF to deliver a fax to the number specified in the subject.

In the above case, an email account fax@mirtapbx.com was created on web-hosting company and it is automatically polled every minute to check for new email. If an email is found, the sender email is checked against the Authentication Data. The only authentication type currently allowed is “Sender Email”. Just put the email or domain separated by CR or ; or ,



Voicemails

Voicemails can be created directly when creating an extension or using this menu. A voicemail box or mailbox have a number associated and a PIN number. When creating the voicemail from extension page, the PIN is automatically generated. Voicemails can be listened from the phone, by creating a feature code to access them, dialing you own voicemail and accessing the operator panel or using the web interface, using the menu Status/Voicemails. Voicemail email can be customized using an email template.



Email is the email to send the voicemail message, with the attached voice message.

Timezone can be used to choose in which time zone the message will be sent.

Pager is another mailbox to be used for receiving small notification when new messages arrive.

Envelope playback controls if asterisk needs to play the date and time before playing any voicemail message.

Attach message allows to choose if to attach or not the recorded message to the email.

Automatic delete will delete automatically any received voicemail message once delivered by email.

Allow review permits to the caller to review the message left and maybe rerecord.

Greetings controls what message to play to the caller when hitting the voicemail box.

Language specifies the language to use when playing messages to the caller.

You can specify the Max number of messages, once the voicemail hits that number of messages, no more messages can be stored or received.

With Min len of voicemail message in seconds, if a message left was shorter than the limit set, the message will be discarded.

Enabling Operator/Extras permits to the caller to access the voicemail messages entering the PIN or performing special actions if the operator key is used. Press * to be prompted for the Pin or press 0 to access the operator destination defined.

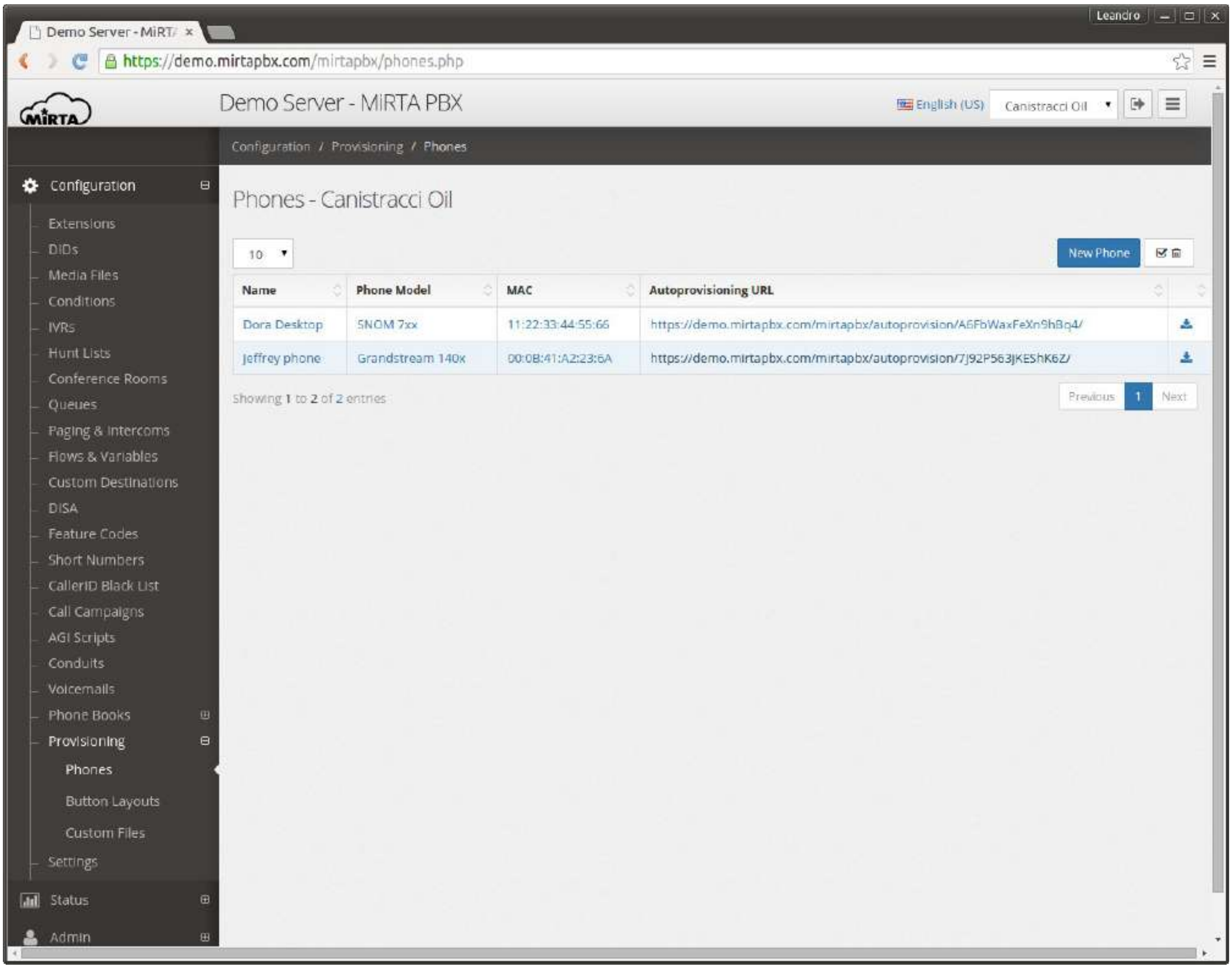
Greetings allow you to upload a custom message for voicemail internal messages.

Provisioning

Provisioning is the action of configuring a phone automatically, by providing only basic informations. MiRTA 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.

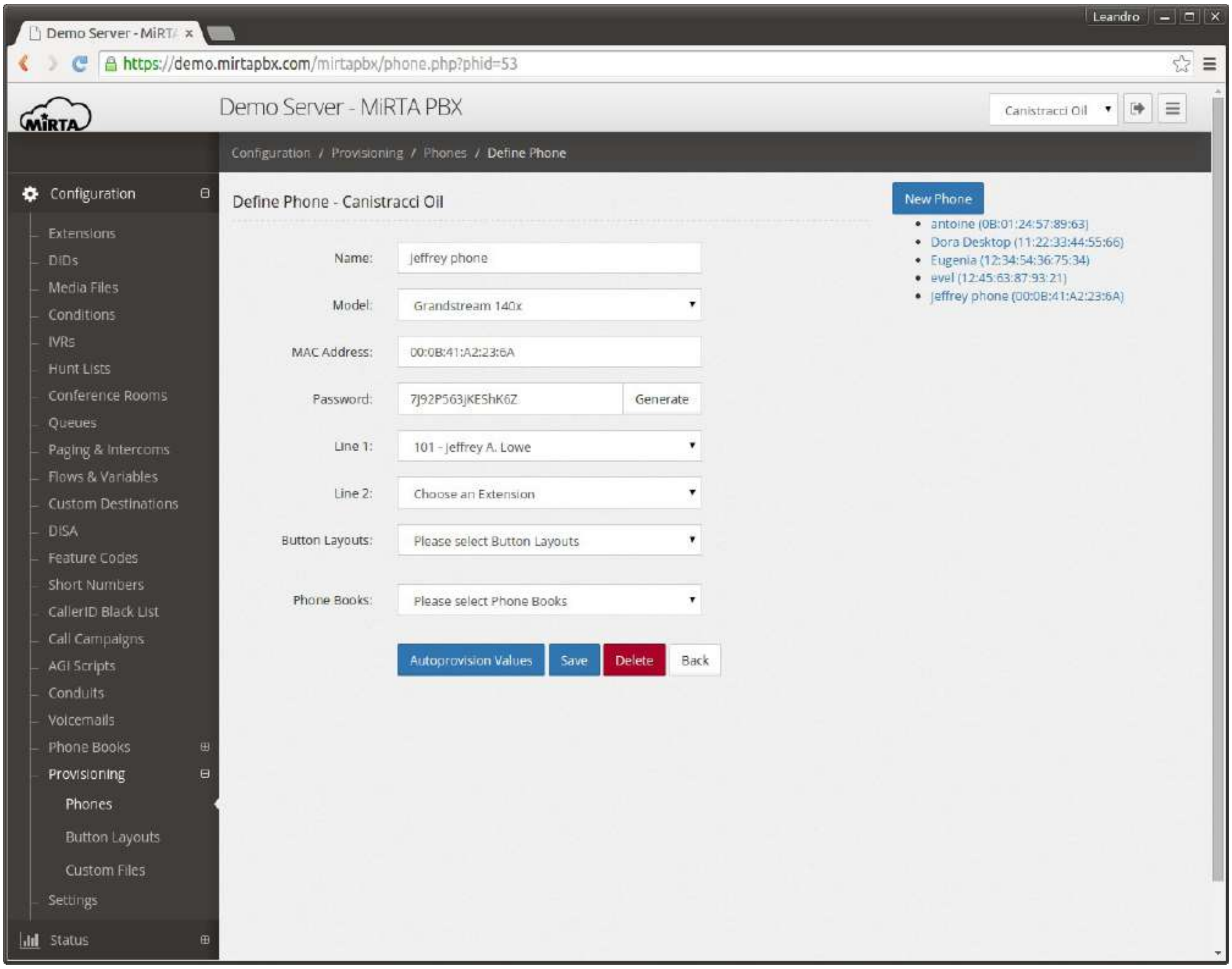
Phones

Files used for provisioning are usually text or XML file containing informations like the user and password and the ip address or hostname of the SIP server. Informations contained in the provisioning files need to keep confidential and the leakage of these informations can lead to unauthorized usage of voice traffic. To avoid any snooping on provisioning content, usage of HTTPS is recommended. Be aware some phones requires a valid SSL certificate to provision using HTTPS and some other (Cisco) require a certificate signed by the manufacturer. 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.



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

Password is a random key needed to prevent any brute forcing of the mac address. It is needed to be added to the provisioning URL.

Based on the definition of the phone model, one or multiple **lines** can be shown, allowing to select one or multiple accounts from the Extensions defined for the tenant.

One or multiple **button layouts** can be defined and assigned to the phone.

One or multiple **phone books** can be defined and assigned to the phone.