CloudPBX

Cloud PBX

call processing service

Management User Guide

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1. Organization of cloud PBX

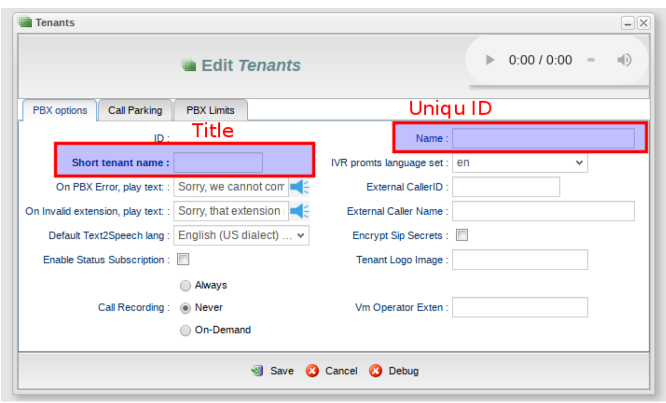
The cloud (or - Tenant, unit, department) is an independent PBX with independent settings, where a set of services can work exclusively for this group of subscribers / users of the cloud, without intersecting each other.

The cloud and global settings of the system (trunks, the pool of DID numbers) are created by the user with the rights 'superadmin', the administration of the cloud is created by the user with the rights 'Tenant Admin'.

DID numbers and trunks (SIP gateways) through which calls come and go are created globally. When creating a trunk for outgoing calls, you can specify one or more tenants to whom they will be available. When creating a DID number, you can specify only one tenant where the incoming call will be directed to this DID number.

All clouds on the same server operate on the same SIP port 5060 and are based inside the same SIP server, isolated from each other using the context and Dialplan structure.

The cloud has two main parameters:



- the name (title), which has a free text form and is displayed in reports or menus, can change;

- unique identifier (unique ID), may contain only letters and numbers, do not change after creation

It is important to understand that users of different tenants can have the same extension number 101, but at the same time the line will be sip registered using different SIP-alias and a password, which looks like sales-101, for example: p @ ssws and support-101: p @ sswd where sales and support are cloud identifiers.

Also, the cloud can have one common external Caller ID for outgoing calls. It will be displayed for all outgoing calls of the operators of this cloud calling through outgoing routes to external trunks. Thus, each department / unit with outgoing calls can have its own external number displayed (instead of the internal 101, 102 ...)

Authorization and access

Cloud PBX has two types of access - for system administration and for users (agents).

Administrative access, in turn, is divided into

- superadmin: the role of a global administrator - has access to the entire system, can switch between clouds and create / delete them, as well as a pool of DID numbers, trunks

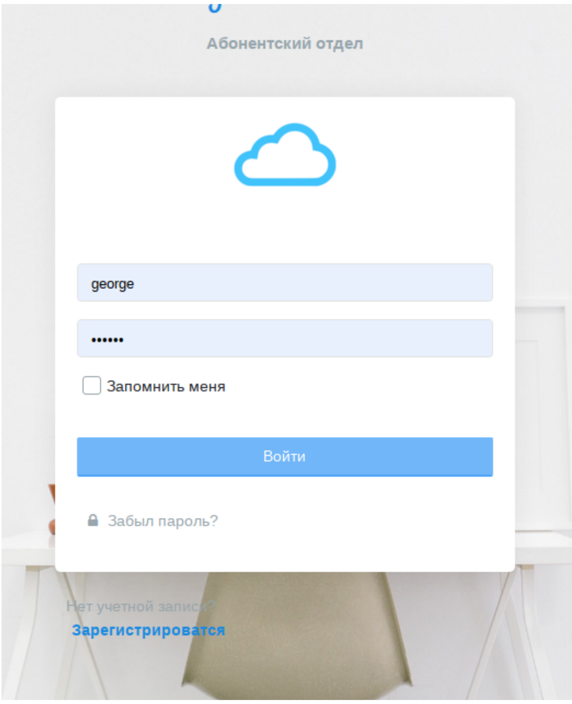
-PBX Admin: has access only to its designated cloud, does not see and does not have access to other system clouds. It can create internal numbers and services, configure routing for the cloud.

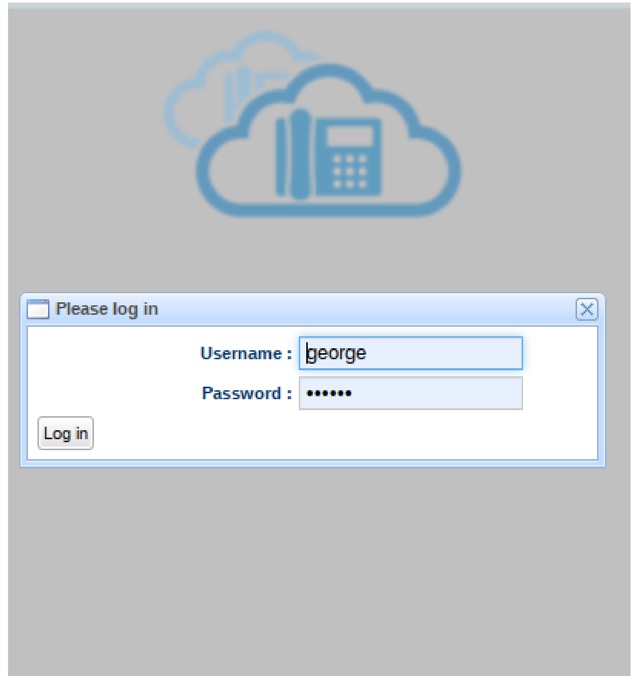
- **PBX Operator** : it has limited access to the administration panel (Allowed Sections), cannot add new users or lines - it can only change the settings of extension lines only, and if it has a designated line, it only sees it. The agent can use this access and configure its own line (SIP password, call forwarding, screening)

- **CRM Operator: has access only from the CRM panel, and sees the entire history of cloud calls, can receive and make calls ..**

- CRM Phone: it has the smallest access to CRM, sees only its calls and calls of those dialer groups where it consists, can receive and make calls.

The appearance of the CRM authorization window and the Administrative

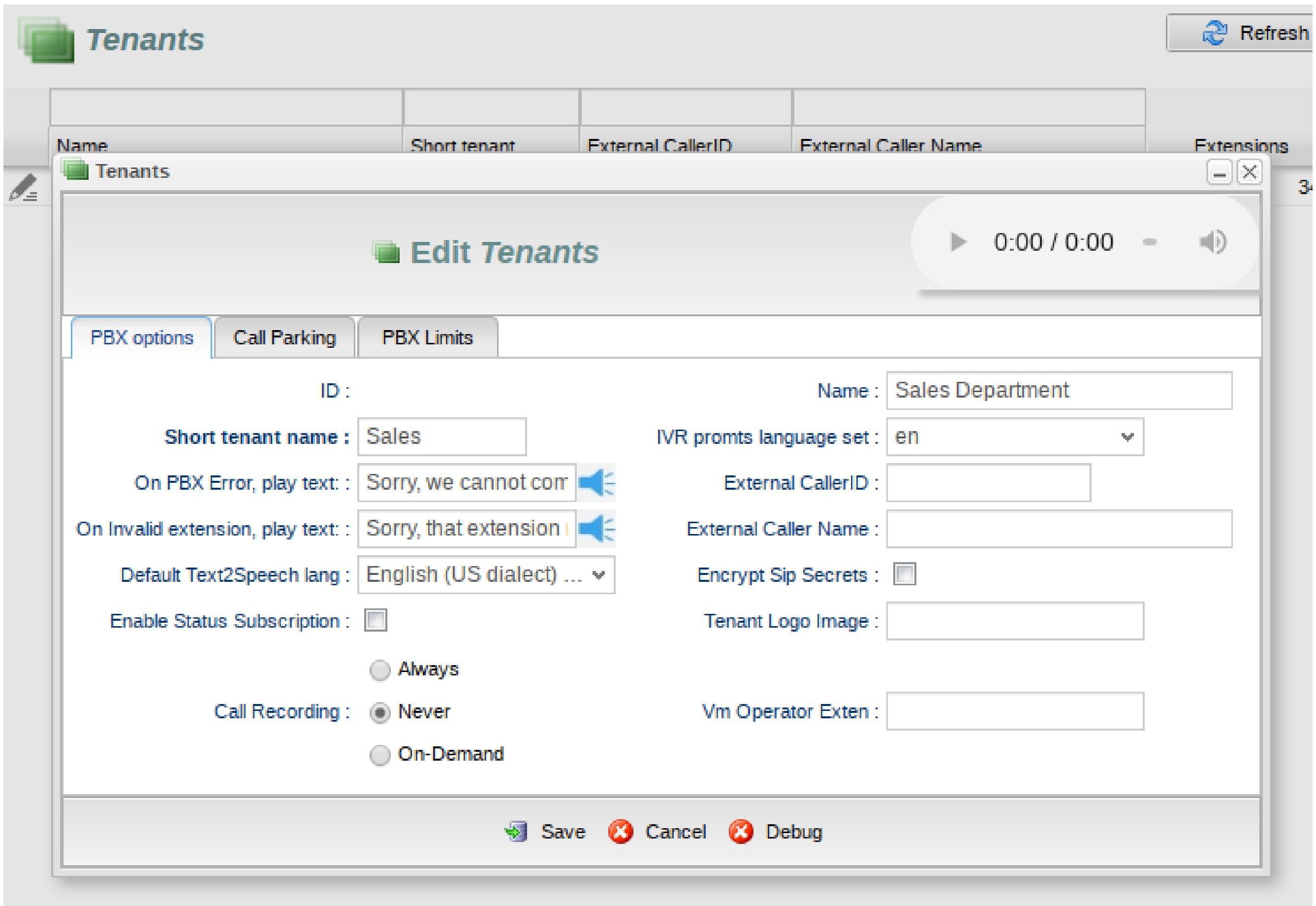




Panel:

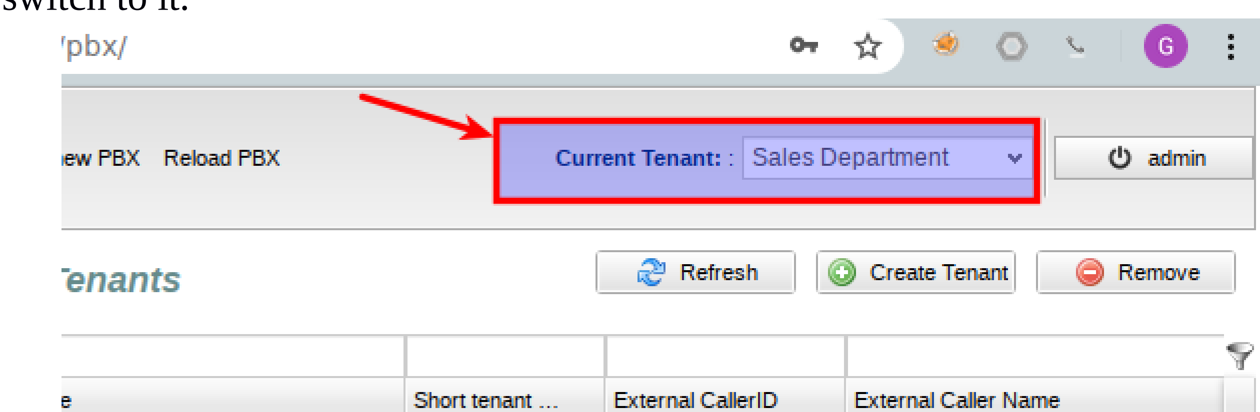
4. Creating a PBX cloud

Creation of a unit or cloud is performed by a user with rights **superadmin and it happens very simply:**
The minimum data for the cloud is its “Short Tenant Name” and “Name”



After clicking Save, a cloud will be created.

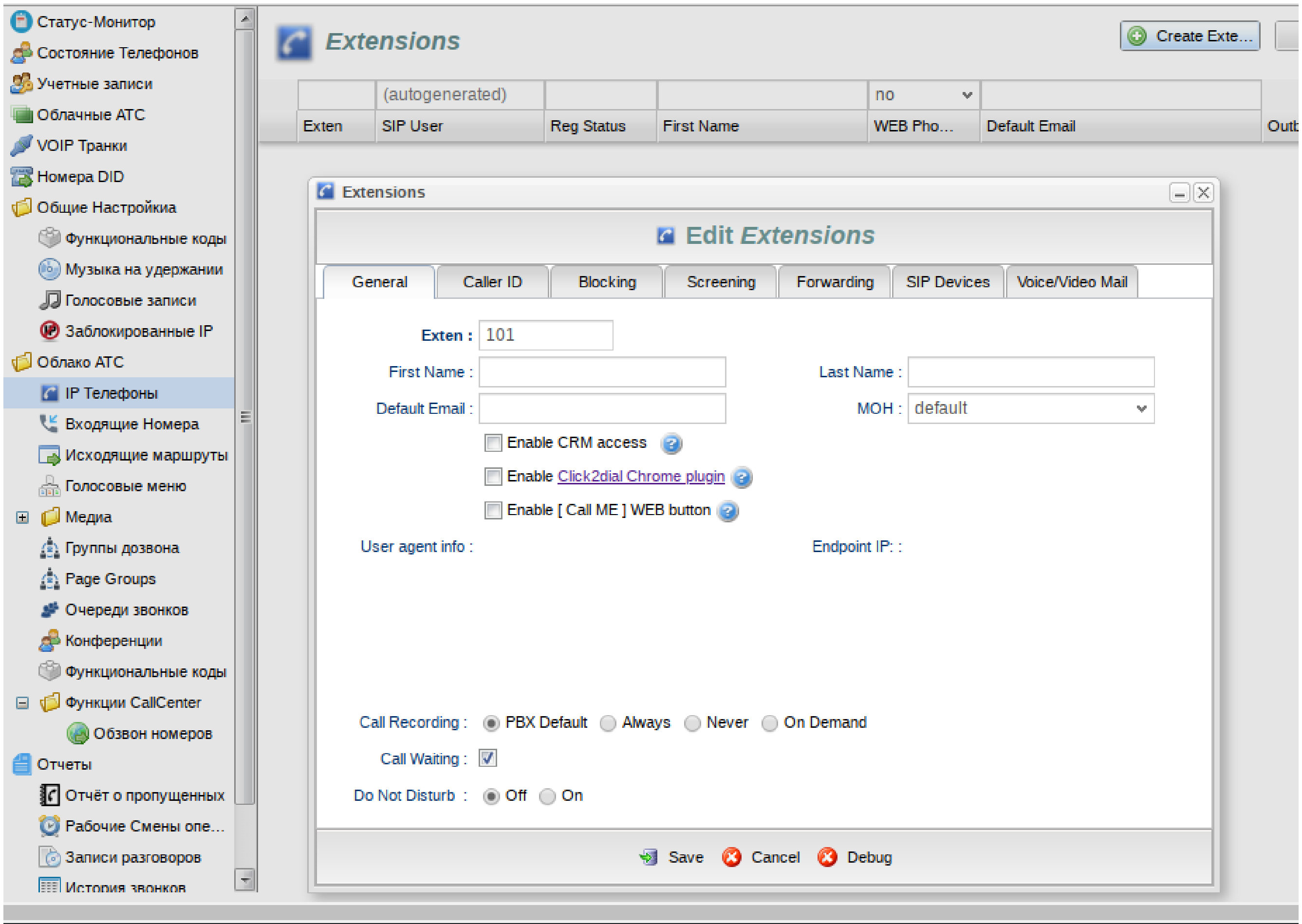
To switch the interface to a new cloud - find it in the list at the top right - and switch to it.



Key cloud features

Creating internal phones (SIP extensions)

Adding internal phones in the cloud with default settings takes place in a “one click” and the maximum is simplified.



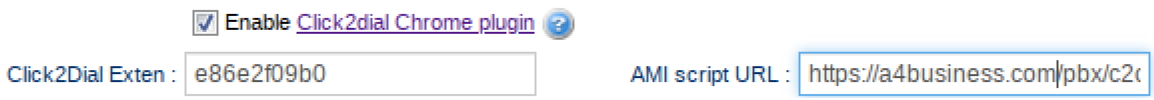
To create, go to the tree branch - General Settings: IP Phones, and click the “Create Extension” button.

In this case, all necessary fields are filled with default values, after clicking “Save” the line will be created. If necessary, later, it will be possible to set the name and surname, open the support for CRM (Enable CRM Access), etc. .. The next time you click “Create Extension”, the Exten field will be set to one more and you just have to click Save, this gives the ability to quickly create lines of the required quantity.

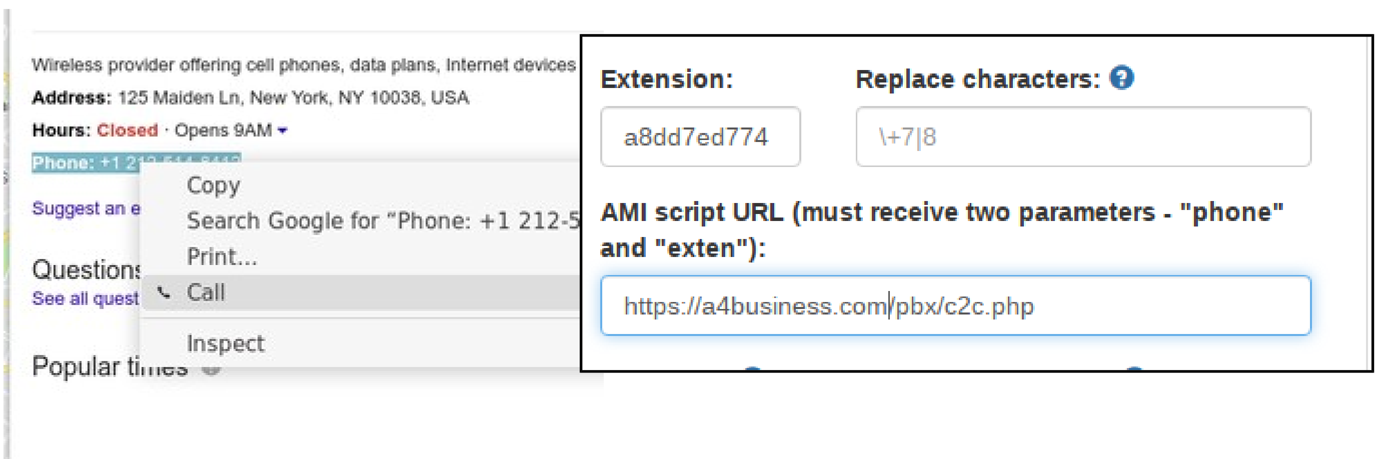
[General] Tab

Key tab options:

CRM Access: after the line is created, enabling this checkbox and setting the login password will create a CRM account through which the user can make and receive calls via the WEB phone. This option is not required - the user can use SIP SoftPhone or another SIP client (Cisco IP Phone) connected to the PBX using SIP Alias and password to make calls.



Click2Dial: this option allows you to use the Click2dial plugin to quickly dial numbers from pages by highlighting the number and pressing Call from the context menu of the mouse. Plugin Details:https://chrome.google.com/webstore/detail/asterisk-click2call



After installation and configuration (only two parameters - Exten and AMI script URL), for a call you need to have a WEB or SIP phone registered, select a number and press Call ,. If everything is configured correctly, then, first, a call will come to the internal number, after the operator answers, the second number will be dialed.

Used to automate the work of the operator when you need to dial a lot of numbers from the list and avoid frequent switching between tabs when Copy / Paste

[CallerID] Tab

Allows you to set outgoing CallerID for internal and external calls. At the same time, external calls will have this CallerID despite the global Cloud option on the external Caller ID (if installed).

[Blocking] Tab

Outbound Calls: It allows you to set rights to outbound routes. Options:

- *All (Default):* all routes are allowed (except for Outgoing routes with the status of “Private”);

- *Only Internal Calls* : only calls to internal numbers;

[outgoing routing table name]: calls are allowed to all internal numbers and only those destinations specified in this table. As a rule, this table has the status of “Private” (see Outgoing Routing Settings), for setting individual routes.

Inbound calls - allow or block outgoing calls;

Anonymouse Calls - allow incoming calls with hidden CalelrID;

Call Block Method - if calls are barred, this option determines which method to pick up calls, options - Busy, send to Voicemail, simulate a call;

Caller Block List: - black list of numbers from which to block calls

[Screenings] Tab

- The screening function offers first to be presented to the caller (the question is asked with the answer: “enter your name and subject of the call”), then dial the extension line, notify of the incoming call and play the record, then give the choice to receive the call, drop the call or voice mail . This feature allows you to “Screening” unwanted calls. There is also a list of rooms– for which it’s always possible to do screening either never.

[Forwarding] Tab

- The tab sets the options for redirecting an incoming call.

Forward Timeout - time after which to redirect (if the call was not accepted). 0 - for unconditional instant redirection of all calls.

Always Forward: always redirect (regardless of status) to voice mail or number (can be either internal or external - will be dialed through external lines)

Forward When Busy: redirect only if the line is busy.

Tag forwarded Call: add text to the name of the caller, so that this mark will be visible on the party receiving the redirected call (for example, FWD will signal that it is a redirected call)

FollowME: This option makes it possible for an incoming call to perform a serial search on a group of numbers. This can be a group of different numbers (both internal and external), which will be used to dial to search for a subscriber. The search will be done before the first response.

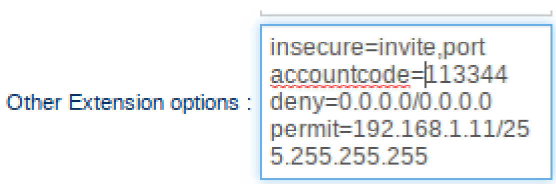
[SIP Device] Tab

Configure SIP Extension:

The main two parameters are SIP User and SIP Password, by which this telephone is registered. Address SIP Registrar and SIP Proxy is the address (domain name) of the server. The remaining parameters are the Asterisk SIP parameters from sip.conf [peers and clients], are optional, and are described here:https://www.voip-info.org/asterisk-config-sipconf/ .

Any parameter can be added to 'Other extensions Options' with the format: key = value.

for instance, this set of parameters implements the following tasks:



a) to allow calls from this device without an additional password request (disables SIP Unauthorized with INVITE),
b) to transfer the account ID when billing outgoing calls by external billing
c) Permissions to use the line only from the local IP: 192.168.1.11

A WEB phone uses the same SIP line parameters as a regular SIP client, using a JavaScript-based WEB client (sipjs, sipml5, jsSip) when loading a CRM page. The operator can use only one type of line at a time - WEB Phone or SIP, the simultaneous operation of two protocols is not supported by the Asterisk system.

To switch calls from the WEB phone to the SIP client and vice versa, the operator only needs to turn it off / on from the CRM interface using the “Power” button. When attempting to make a SIP call by a client with 'WEB phone' turned on, the 'SIP / 488 Not acceptable here' error will be displayed. The WEB phone mode can also be switched by the administrator using the 'WEB Phone' option on the SIP Device tab.

It is located in CRM and you can monitor active calls and call history in any condition of the WEB phone.

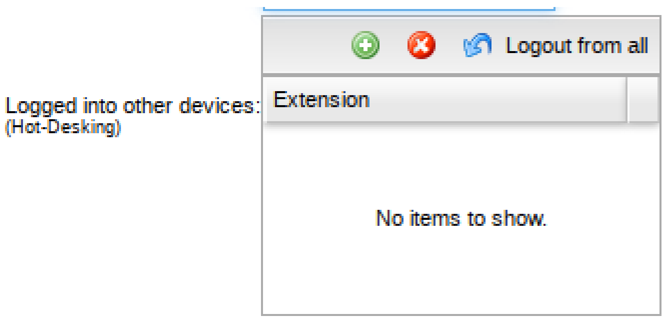
There are two modes of monitoring the state of the WEB phone when the operator enters the CRM interface:

a) automatic inclusion of WEB phone;

b) saving the previous state of the WEB phone

This mode is set by the system administrator.

Host Desking Option



This option allows the operator to receive calls to their extension number from any other registered cloud phone, and included in the list of Host Desk of this operator. In this case, calls to the original line number will be sent to VoiceMail or dropped (according to the settings of this line). Host Desking is managed using DTMF codes and has the following commands:

* 11 - login to your account from the current phone. A login will be requested (the number of the new number for receiving calls, and password (password from the voice mail number). If authorization is successful, incoming calls to the specified number will also come to this line. In this case, the new number should differ from the current number of the selected line .

* 12 - Release this line from HotDesk mode. It occurs without additional questions, all calls of the main line are resumed in the normal mode as before.

[Voice / Video Mail] Tab

Setting up a voicemail box. Voice mail verification is done through DTMF codes (see Feature Codes)

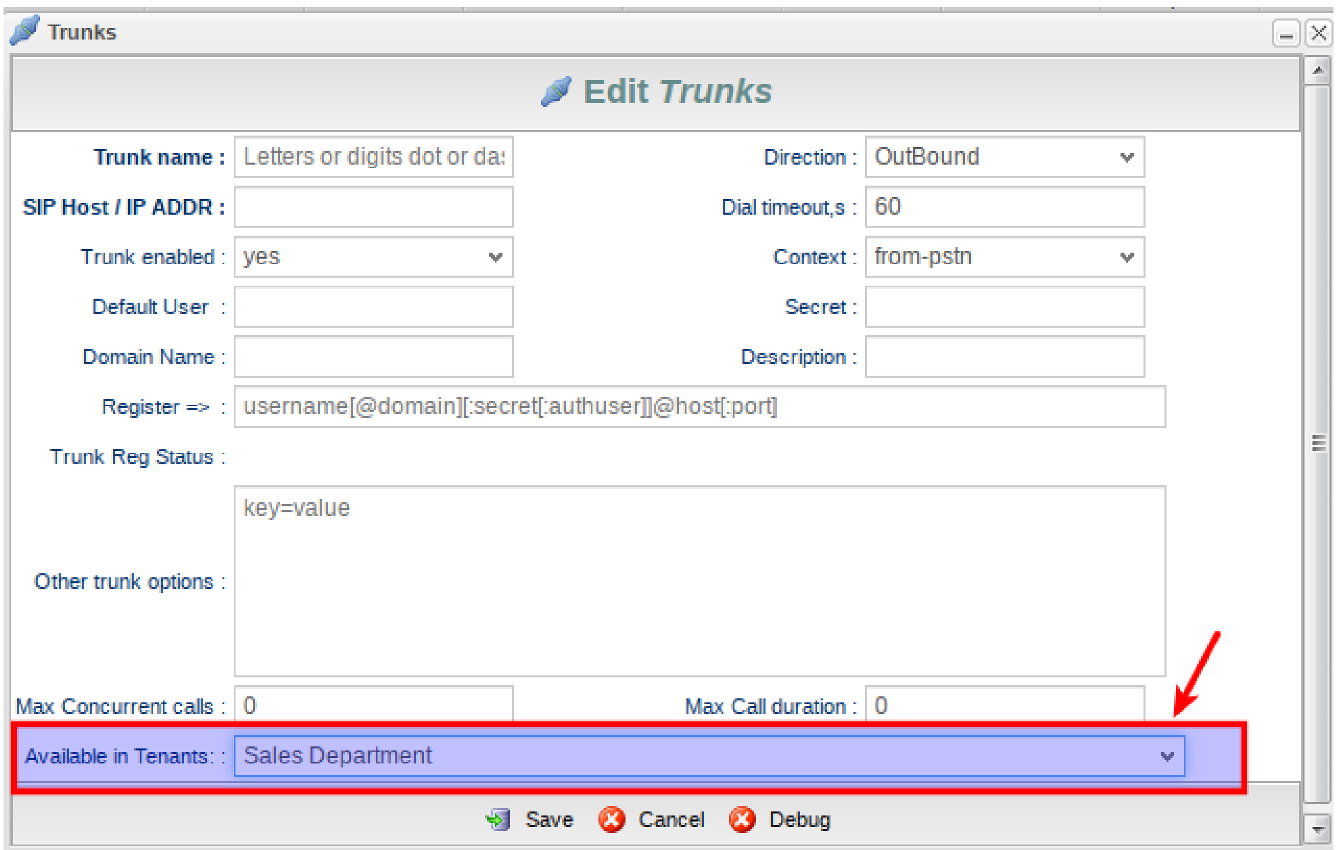
Voip SIP Trunks and Trunks

VOIP SIP Trunks are used to create communication with gateways or VOIP service providers via SIP protocol.

The trunk option set is the same as in Asterisk. In this case, the trunk can both be registered on the remote server with a username and password (Register: => field) and allow the remote line to be registered on the current server (trunk name as login, secret as password)

Field Trunk Name - trunk identifier, can contain only letters and numbers. For an external registration line on the current server, the trunk name must match the user name (for example, the GSM gateway line)

Field **“Other trunk options” you can add any additional options that Asterisk supports.**

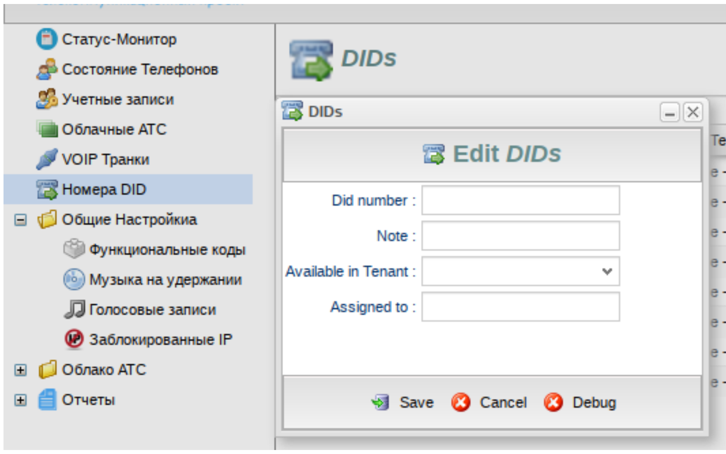


An important option is to define the Cloud for which the trunk will be available. One trunk can be used in several (or all) clouds, and give the opportunity to use the same lines for outgoing calls

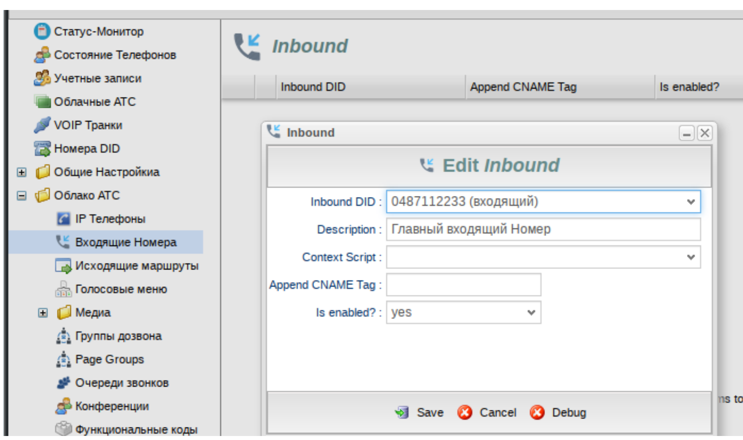
Creating Incoming DID Numbers

Incoming numbers (DID) allow you to receive and distribute calls between internal clouds, lines and services.

The sequence is as follows:

-

-First, a user with Superadmin rights globally adds incoming numbers to the system in the form in which they come from the provider (otherwise there will be no coincidence of the incoming number with the configured one). It is important to set “Available in Tenant”: the DID number can only be accessed for one cloud at a time.



-Then added numbers are configured in the menu of incoming numbers of the specified Cloud (this can already be done by both superadmin and Tenant Admin). If the number is reassigned by the super administrator to another cloud, then in this cloud the record remains with the Inbound DID number empty, and will go back if the appointment is canceled.

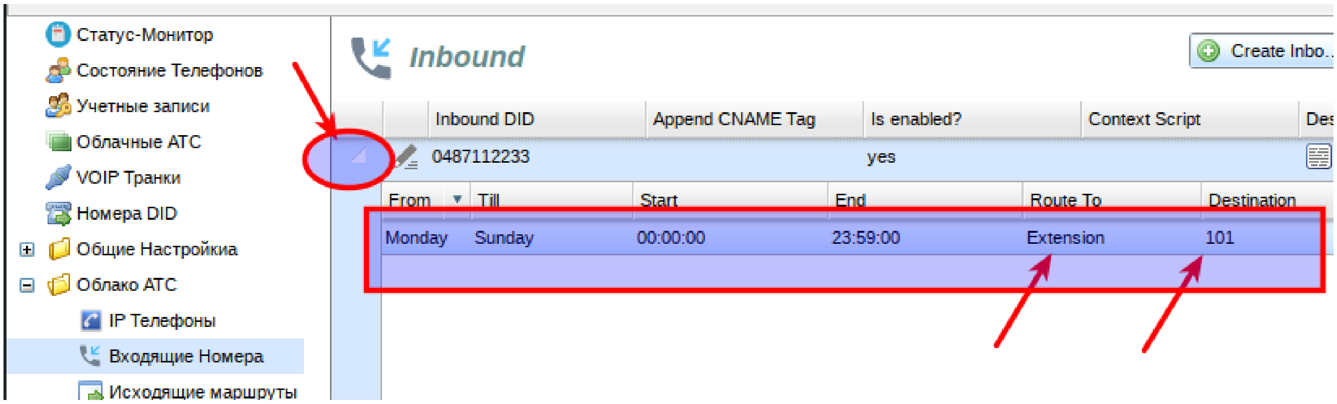
Append CNAME tag: allows you to additionally set a label to the name of the caller, which will be displayed on the incoming call of the received party.

Context Script: allows you to execute a third-party AGI script or transfer temporary execution to the DialPlan context through the GoSub () command. when a call hits this number. Typically, this is an AGI script written in Python or PHP and located on the server in / var / lib / asterisk / agi-bin /, which performs such actions on the call as:

- search for data about the caller by his number in external sources using the API;

- playing information in a line, for example, balance or account status, service operation, etc.

The final step in setting a route is to create inbound routing rules with end events / call recipients.



Inbound Routing Rules **:**

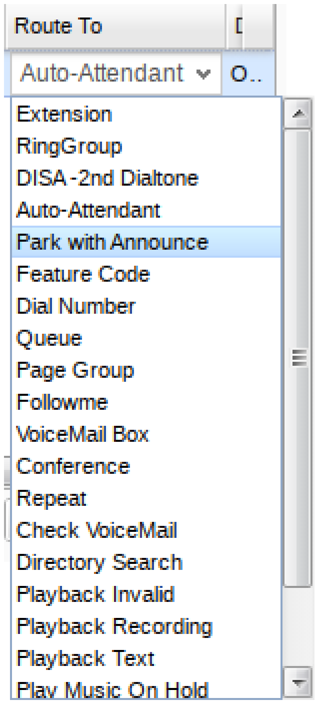
After adding the number to the table - there is an opportunity to set the incoming routing for the number - that is, redirect it to the right place at the right time.

WITHa small button on the left of the number opens a sub-table with rules for routing, with the ability to specify a filter by time. By default, it covers the entire range of the day: 00:00 - 23:59. If necessary, you can set different events for different times.

As a rule, a call is sent either to the Queue or to the Voice menu, or directly to an internal number. For example, indicate 101, which means that all calls will be redirected to this number.

There can be an unlimited number of rules, and they will be executed sequentially (if the time interval coincides in several rules and also when the call is not answered after the first rule is executed), or once (if the rule is selected separately according to the time interval, or when the call breaks after the first rule).

Available options **“Route To [Destinaion]”:**



Extension: The call goes directly to the extension number;

Ringgroup: Sent to a dialer group, usually to search for a specific contact by a group of numbers, and if not dialed, to the Call Queue.

DISA-2nd Dialtone: The function provides the caller with a secondary DialTone for further dialing. This can be either an internal number or any other number that will be dialed on behalf of the current PBX Cloud using outgoing routes. The function allows operators, or other persons, to make calls on behalf of the company remotely;

Auto-Attendant: Virtual Secretary. Separately customizable Menu, described separately;

Park with Announce: parking a call to virtual numbers (usually 7XX), and playing a notification to a specific configured speakerphone device number (configured in the Tenants menu). A parked number can be answered by any operator by dialing a spoken virtual number;

Feature Code: execute a set of DTMF commands;

Dial Number: dial the specified number. The call will be redirected to this number using outgoing call routing;

Queue: send to Call Queue (described separately)

Page Group: send a call to a group of numbers that will automatically answer the call in speakerphone mode. Used for alerts and other purposes;

FollowMe: sequential search in a group of numbers until the first answer. Used to search for a contact that has many dialers and can be accessed under different phones at different times;

VoiceMail Box: send a call to the voicemail box of the selected extension number;

Conference: send a call to the specified Conference, where according to the settings, the caller falls into the group of simultaneously talking contacts;

Repeat: returns the call back to DialPlan;

CheckVoiceMail: the call goes to the voice message verification application of the selected extension. This will ask for a password.

Directory search: run the Directory Search application, which asks the user to enter the first few characters of the contact's name (numeric keypad), and search the names of the voicemail mailboxes for agents. Described in detail on the VoipInfo website:https://wiki.asterisk.org/wiki/display/AST/Directory+Application,

Playback Invalid: play the standard error message set in the cloud settings;

Playback Recording: play the specified media file with the recording;

PlayBack Text: play text using the Text2Speech service, according to the settings specified for the current cloud (language and voice)

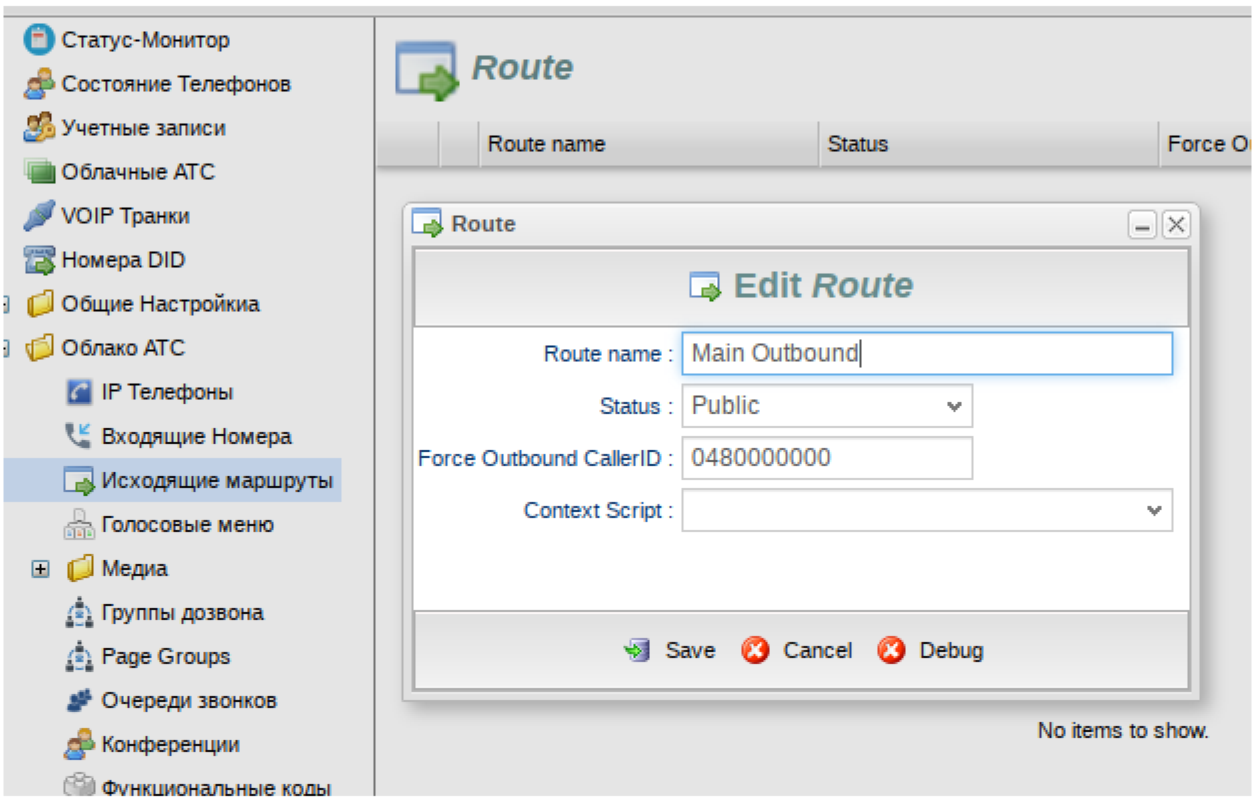
Play music onhold**: Start playing the specified music on hold, and play until the connection is disconnected.**

Outbound routing

Outgoing routing sets the rule for dialing from internal numbers or redirected numbers. Each cloud has its own rules for outgoing routing, but they can use the same trunks.

Routes for internal communications are always available and do not require creation. Dialing an internal number always gets on the telephone of the operator if it was created, regardless of its status (Answer Voicemail if its line is not registered).

Creature routing tables inside the cloud:



Route Name: route name to display

Status:

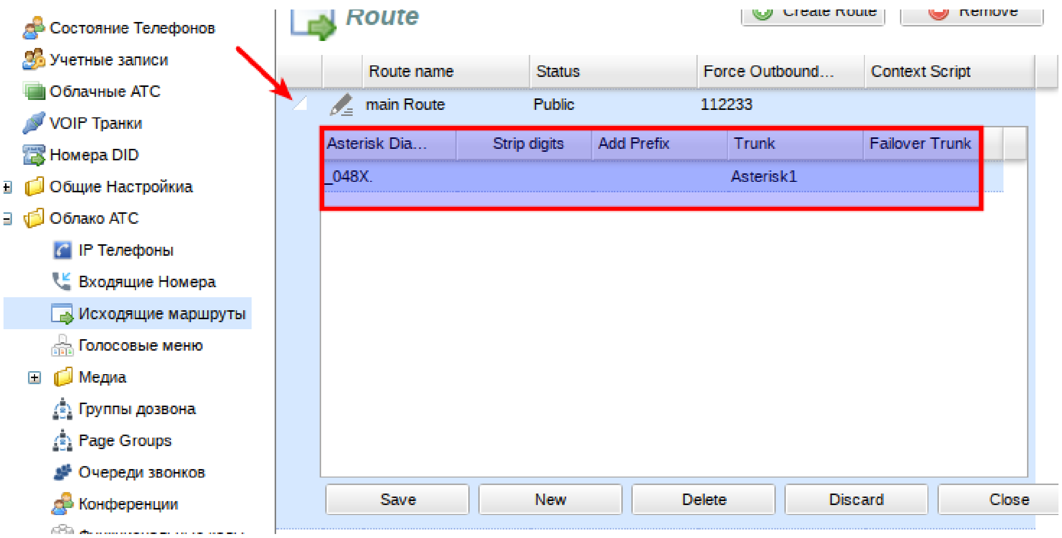
Disabled: ***route unavailable***

Public**: available for the entire cloud (default)**

Private**: Available only to those lines that have chosen it as their route in the [Blocking] masonry.**

Context script : For an outgoing call along this route, it allows you to execute a third-party AGI script or transfer temporarily execution to the DialPlan context through the GoSub () command.

Adding routes:



After creating the routing table, you can add routes to it.

Asterisk Dialplan: this is a mask by which a check is made for coincidence of the number. Description here https://www.voip-info.org/asterisk-dialplan-patterns.

Strip Digits: How many characters to remove from the number. A positive number is starting on the right, a negative number is starting on the left. for instance

Strip digits: 3: 04872232 → 04872

Strip digits: -3: 04872232 → 72132

Add Prefix: prefix to add to the beginning of the number. For example 38 (country code)

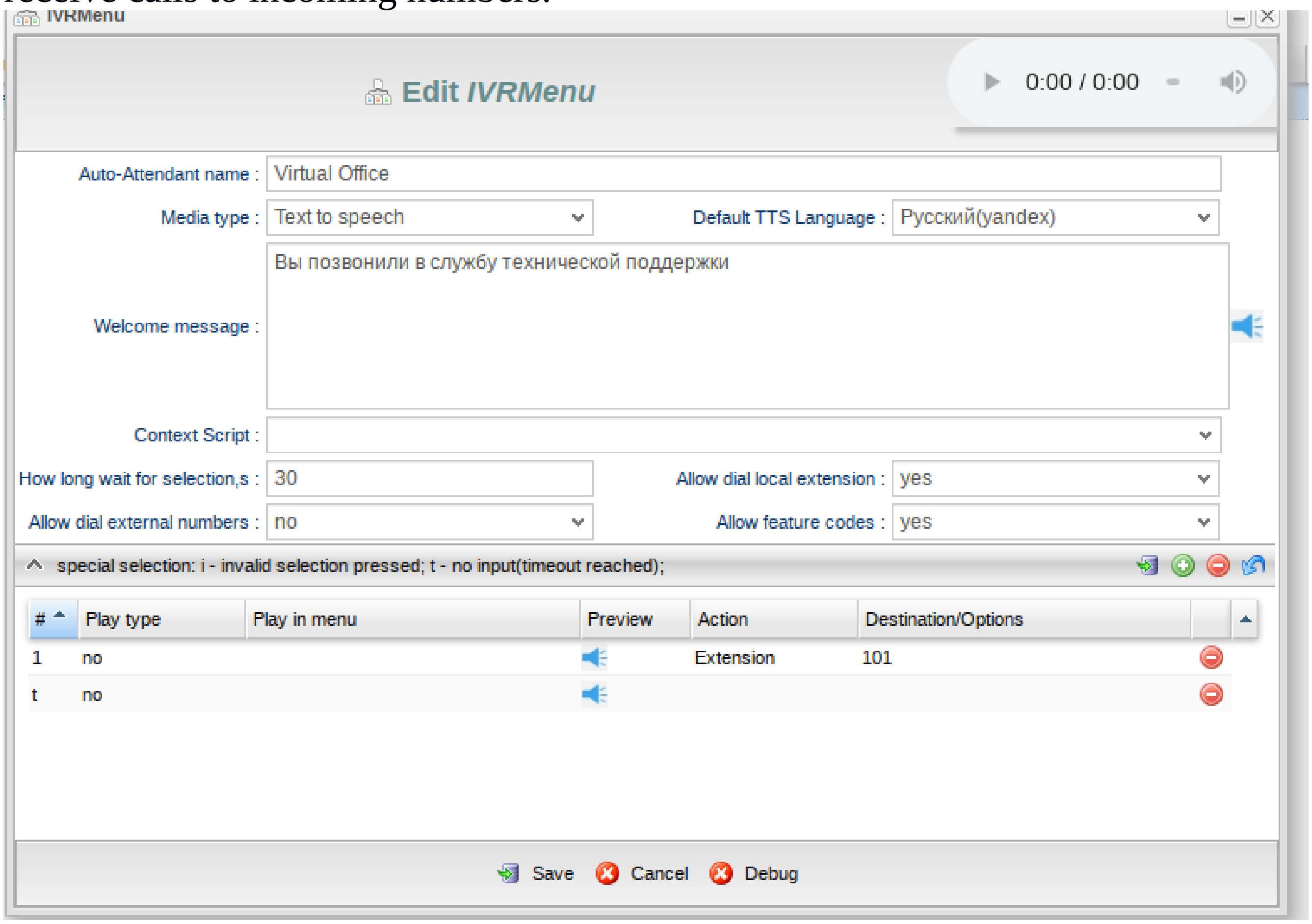
Trunk: Trunk to send a call through.

Failover trunk: If the first trunk is unavailable (dropped the call), then try to call through another.

6. PBX services for processing calls

Golos menus

This function is one of the main mechanisms for receiving and processing calls. The menu structure is created by the administrator one-time, and can be used to receive calls to incoming numbers.



To create voice recordings, the speech synthesis service is used, which allows you to create high-quality voice greetings in different languages with the same parameters of voice, level and intonation.

You can also use custom recordings - Play Recording. You can upload a recording by selecting the Media Type option 'Upload' and specify a file.

The generated voice message may contain a greeting and a list of options available to the caller to select. To listen to the resulting  message - you need to click on the speaker

To add options - click on the + button above the table, specify the DTMF sign (from 1 to 0, * or #), and specify the Action that will be performed when this option is selected (for example, the connection with extension 101 is selected).

Play Type / Play in Menu / Preview options can be suppressed without changes; they additionally make it possible to set a voice message for each option separately.

There are two additional options -

t: when this option is specified, an Action is triggered if the user has not made any choices during the interval specified as “How long to wait for selections”.

e: it works when the option is selected incorrectly (not in the list)

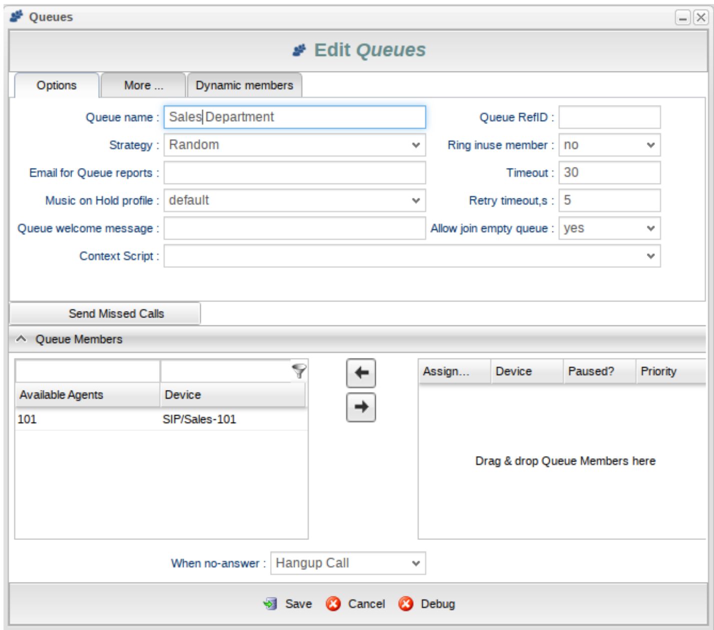
Allow Dial Local Extens: From the menu, it is possible to dial the extension number without selecting a separate item. For example, by the message “If you know the number of the extension you need, dial it now,” the user can dial, for example, a three-digit operator number.

Allow dial external Numbers: In the same way, you can let the caller dial any number - and the call will be made on behalf of the Cloud through outgoing lines. In this way, you can enable employees of the company to make calls on behalf of the company from any phone - by calling this main menu first.

Allow feature codes: allows the caller to use DTMF codes in the voice menu.

Call Queues

Call queues is the main function for servicing the incoming call flow, which distributes them to a group of available operators according to the settings of this queue.



The caller can be in the queue until the first operator is released, while the subscriber plays music on hold. When the time specified as Timeout expires, the subscriber can be transferred to any other place or function using the “When no Answer” queue setting. a list of available functions which is the same as in incoming routing and is described above.

The main settings of the queue:

name: the name of the queue to display in reports and other PBX elements;

Queue RefID: short number of the dialer of the queue, when dialing, the caller can be connected to this queue. Used for quick direct dialing to the queue.

Strategy: operator search strategy:

ringall: *Call all available channels until someone answers.*

random: random choice of operator;

Ring with recent hangup: call the agent who last completed the call;

Ring fewest completed calls: dial the operator with the least number of answered calls;

Ring Linear, one by one: dial sequentially, one by one;

Email for queue reports: when the administrator periodically sends them out, a report is sent to this address daily on the number of received and missed calls of the queue.

Queue welcome message: *lose the greeting before the start of the queue;*

Timeout: time in seconds how much the caller can be in the queue before the response of the event by non-response;

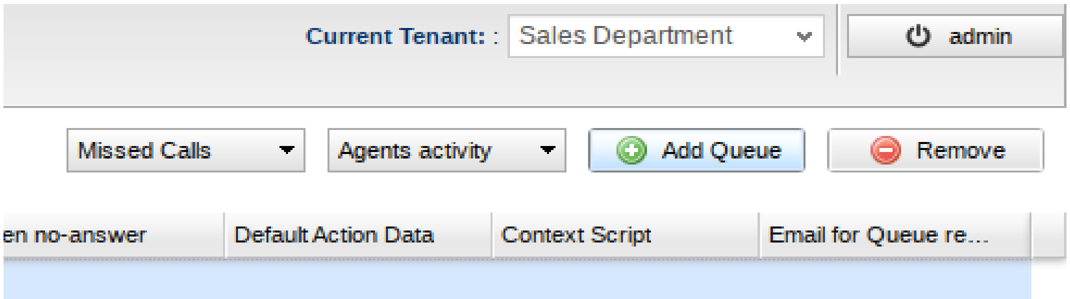
Context script: Passed a call into the queue, allows you to execute a third-party AGI script or transfer temporarily execution to the DialPlan context through the GoSub () command.

Queue members:

Participants in the queue, the operators to whom the calls will be distributed according to the strategy. Additionally, in addition to the strategy, there is a parameter - Penalty (priority) for each operator. With the same priority values, calls will be distributed to everyone evenly and simultaneously. With different priority values, the call will go to the agent with the lowest priority.

A queue participant may have a Paused state, which temporarily stops sending calls to this agent without removing it from the queue.

Change of state Paused / Unpaused generates events on the basis of which reports are generated in PDF format, about the time the agents have been in the queue, which can then be received together with the missed call reports by pressing the corresponding buttons above the queue table for a specific day.



PIf you specify an email address, these reports will be sent daily, when the system is configured to send mail by the system administrator.

Conferences

Conferences provide the opportunity for simultaneous multilateral communication of a group of callers, with the ability to manage the conference by the administrator. The administrator is determined by the entered PIN number at the entrance, where the usual PIN and administrator PIN exist.

Conference number (#room), can be used for direct access to the conference by dialing this number.

Being inside the conference, it is possible to statically set separate menus from DTMF codes for ordinary users and for administrators. Among the menu items the most common are

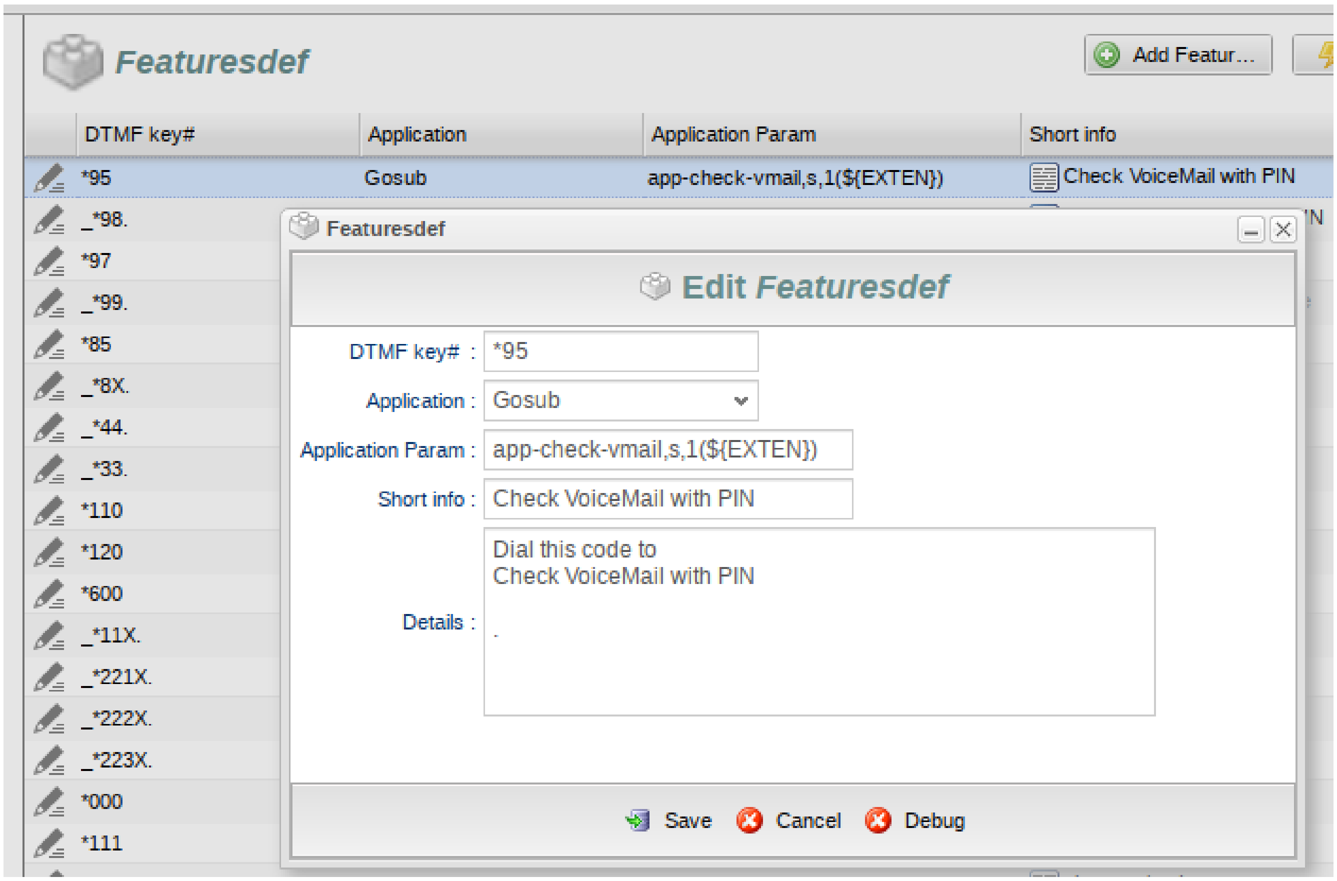
- invite the interlocutor by number;
- set the volume level;
- block, mute the sound for one or all participants;

A more detailed description of the conference function is here. https://www.voip-info.org/asterisk-cmd-conference/

DTMF Feature Set

The system has a basic set of functions that can be called by a set of special sequences, using the characters * and #. The list of functions can be expanded by the administrator both globally for all cloud units, and separately for a particular cloud.

Adding the global DTMF function is carried out by a user with superadmin rights, by adding “General Settings” -Functional Codes in the menu item.



Funkitsa has

-DTMF key: the sequence in which it is called, It can be not only a static code - but also a mask. For example - _ * 11XXX, then when dialing * 11321, or * 11999, the script will be launched, with * 11999 passed as a parameter, where 999 can be processed by the script as an input parameter.

- Application: **Gosub (i**nvoking a context subroutine with increasing control) or AGI - asterisk gateway interface script written in any available programming language and placed in the / var / lib / asterisk / agi-bin / folder.

- Application Param: application call line with parameters (context name with GoSub or script file name with AGI).

- Short Info: A brief description of the function that will be available to tenants when choosing available

- Details: a detailed description of the function

After creating the function, it will be available for all new clouds in the system. To add it to existing ones, you need to click the “**Re-Populate Features To tenants',** and after applying the “Commit Changes” changes, they will be available.

DTMF function example:

Application : app-check-vmail, s, 1 ($ {EXTEN})

Description: the command will be executed

Gosub (app-check-vmail, s, 1 ($ {EXTEN})),

where app-check-vmail: is the name of the context in dialplan (defined in the /etc/asterisk/extensions.condf file),

s is the name exten in the context where the control is transferred (s is the service name, abbreviated from the word “start”),

1 - priority from which execution in a given context begins on a given exten /.

($ {EXTEN}) - in brackets the parameter is indicated, of which there can be more than one, indicated with a comma. In this case, the dialed number is transmitted - that is, the DTMF code itself.

Below is the text of the subroutine itself (context), which checks the voicemail box for the dialed number:

```
[app-check-vmail]
;; Check Video / Voice Mail ;;
exten => s, 1, Answer
exten => s, n, Wait (1)
exten => s, n, GotoIf ($ ["$ {ARG1: 3}" = ""]? channel)
exten => s, n, VoiceMailMain ($ {ARG1: 3} @ $ {tenant} -vmdefault, $
{ARG2})
exten => s, n, Wait (1)
exten => s, n, Hangup
exten => s, n (channel), Gosub (set-variables, s, 1)
exten => s, n, VoiceMailMain ($ {MYEXTENSION} @ $ {tenant}
-vmdefault, $ {ARG2})
exten => s, n, Wait (1)
exten => s, n, Hangup
```

Call redirection

Call forwardingIt works both for incoming calls of the operator and for outgoing, that is, the operator can redirect both the received and dialed calls. Both methods work when the operator uses a WEB phone and a SIP phone.

Unconditional Forwarding

Unconditional call forwarding involves an immediate transfer of the call to the redirected number and the gap the current connection to the agent who initiated the redirect. The caller hears a dial tone and waits for a response. If the redirected number does not answer, the call ends.

To perform redirection, you need to dial a combination in the call mode:

**1NUMBER

after dial ** 1, having heard the invitation to dial the number and the long beep, dial the desired number. At the same time, the call of the operator doing the redirection ends. The caller switches to another number and, depending on the dialing result, connects or not.

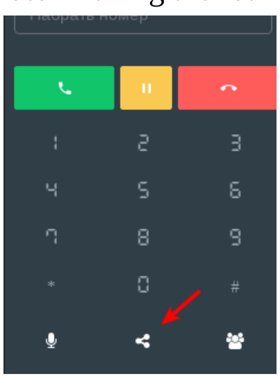
Conditional Forwarding

With conditional call forwarding, the caller is temporarily put on hold (music plays hold), and the operator previously connects himself to the number to which he wants to transfer the call. After the conversation (for example, they will consult or notify / warn about the transfer of the call), the operator simply hangs up, at this moment the caller is removed from the hold and is directly connected to the number to which he was redirected.

To perform redirection, you need to dial a combination in the call mode:

** 2NUMBER

after dial ** 2, having heard the invitation to dial the number and the long beep, dial the desired number and wait for an answer. After the conversation, just hang up. At the same time, the call of the operator making the redirection will end. And the caller will connect directly toredirected number.



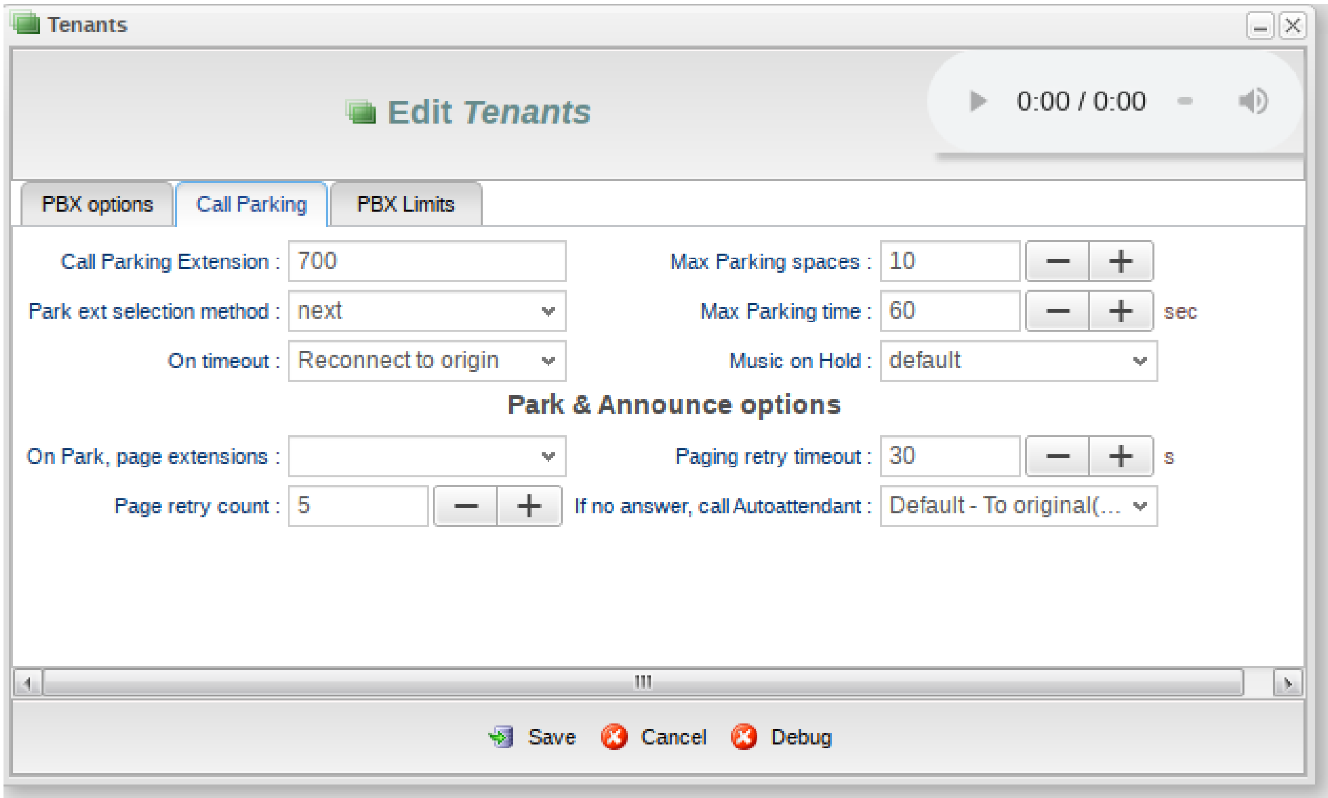
The unconditional redirection button on the WEB phone.

The button appears only during a conversation. When this button is pressed, the program will ask for the number to which to redirect unconditionally. After entering - the system will transfer the caller’s call to this number, and the operator’s call will end

Call parking

The call parking function is to redirect the call to a special virtual number 700 according to the method described above, in this case, a free parking number, usually from 701 to 711, will be automatically selected and played to the operator, which the caller can temporarily stay on hold while the operator makes other calls .

To remove from the parking means simply dial the parking number, and any operator can do this. If no one has removed the caller’s call from the parking lot, he will either return to the operator who parked the call, or simply disconnect. Parking settings are set in the Cloud settings form, 'Call Parking' tab



Call parking extension: Name of the first virtual parking space where all calls park.
Max parking space: number of parking spaces
Park ext. selection method:parking space selection method
Max Parking Time: maximum parking time for calls
On timeout event after the maximum parking time
Music on hold music on hold to be played to the caller

Part & Announce - this function of a notification about a parked call is used to inform staff about the parking number through loud-speaking communication during automatic parking (for example, incoming calls).

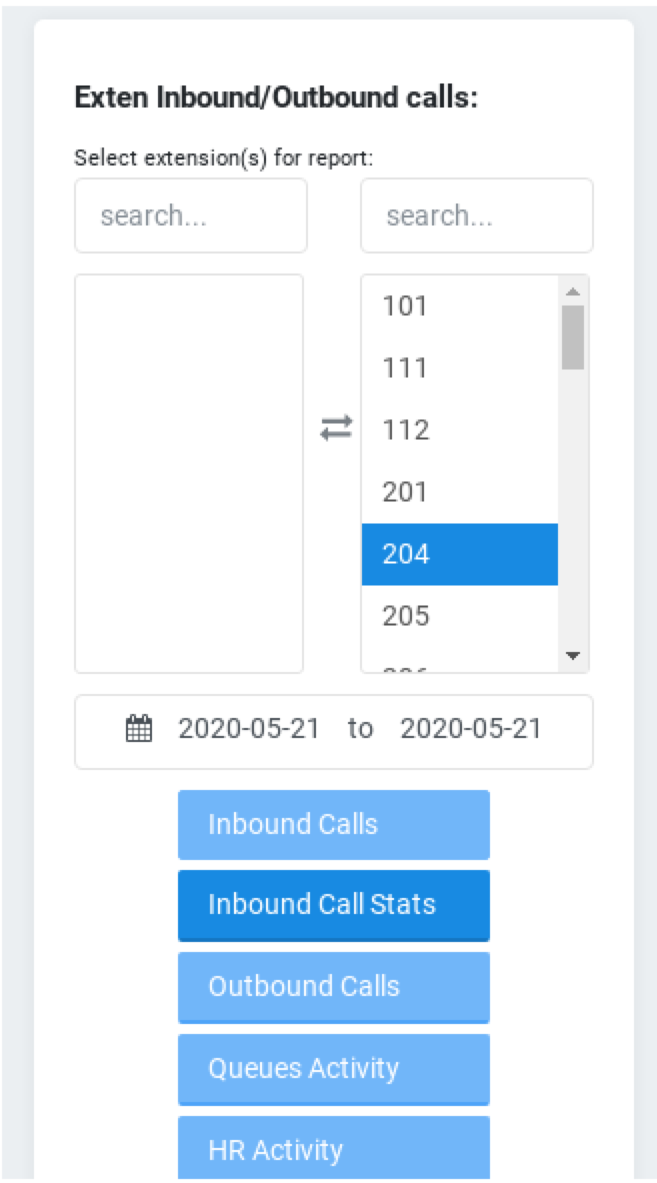
On park, Page Extension: - the extension number to which the speaker is located, and answers all incoming calls through the speaker;

Page retry count [timeout] - how many times and at what interval to repeat the notification

Reports

1. The missed call report displays a list and summary statistics of received and missed calls with a link to the queues, as the main points of receiving and processing the incoming call flow. The report is grouped by day, and can be obtained by double-clicking on the selected day. PDF report

2. Call History. Displays all calls in the search function.
3. Summary reports on internal lines



Under these reports, several types of reports are collected for the selected extension lines and time span.

Inbound Calls - a list of all incoming calls;

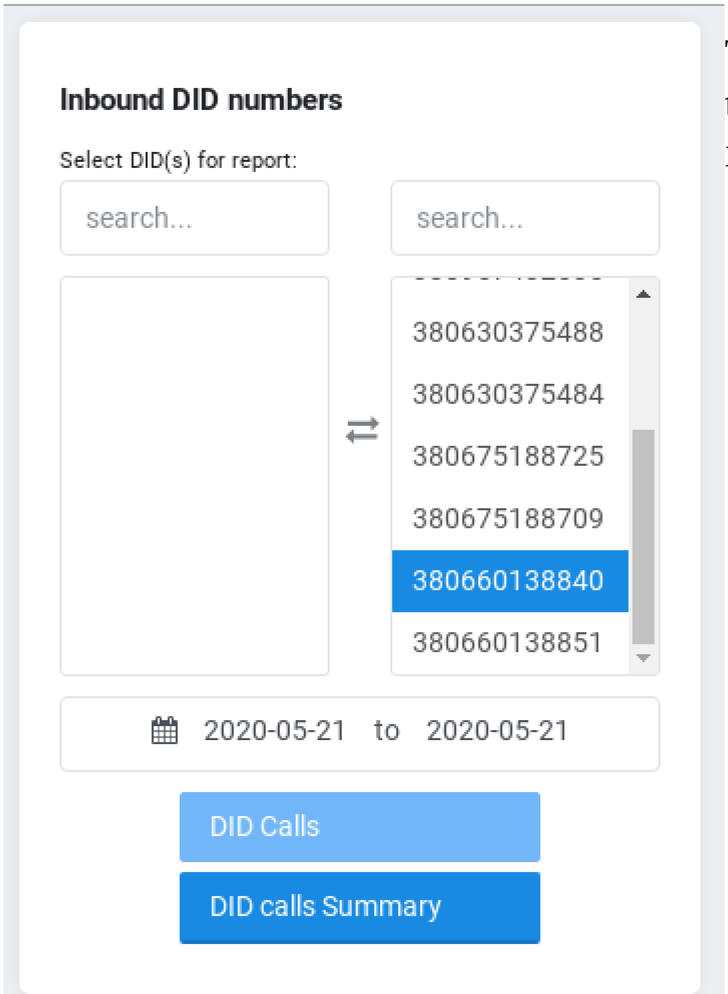
Inbound Call Stats - Summary Summarized Incoming Call Report

OutBound Calls: - List of outgoing calls

Queues Activity report on the activity of agents in the queue (the moment of pausing, entering the queue with the calculation of the time in the queue)

HR Activity - is used together with the DTMF code to register the operator at the workplace and record his working time**.**

4. Incoming line statistics



This report displays a list and the total value of received calls on incoming lines.