
MagnusBilling Documentation

Release source

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CHAPTER 1

Introduction

MagnusBilling is an open source tool written in PHP and JAVASCRIPT, using the EXTJS 6 and YII FRAMEWORK frameworks, aimed at IP telephony providers. MagnusBilling provides a complete and powerful system for anyone to start an IP telephony provider in a matter of minutes. MagnusBilling uses **Asterisk** for connection between customers and providers.

- MagnusBilling can be installed on Linux, Centos 7 or Debian 10.
- MagnusBilling has a modern and robust interface that makes the administration of all functions very easy to understand.
- MagnusBilling has everything that a small, medium and large companies need to start a robust IP PBX system *Youtube Channel*.

CHAPTER 2

Functions

- The system has a lot of functionality to make your server incredibly robust.
- Makes Asterisk easier to manage, and allows you to work without prior Asterisk knowledge, everything is done by the WEB panel.
- Complete WEB panel for the administrator, reseller and customers.

CHAPTER 3

Supported functions

SoftSwitch 4 & 4	SMS	Packages offer	Paypal
VoiceBroadcasting	CallShop	Pre and pos	Pagseguro
CallingCard	CidCallBack	Services	Moip
DID numbers	Ivr	SIP & IAX	GerenciaNet
CallBack	Queus	Reverse URA	and more. . .

CHAPTER 4

Installation

In order to install MagnusBilling you'll need a server with CentOS 7 or Debian 10, minimal install.

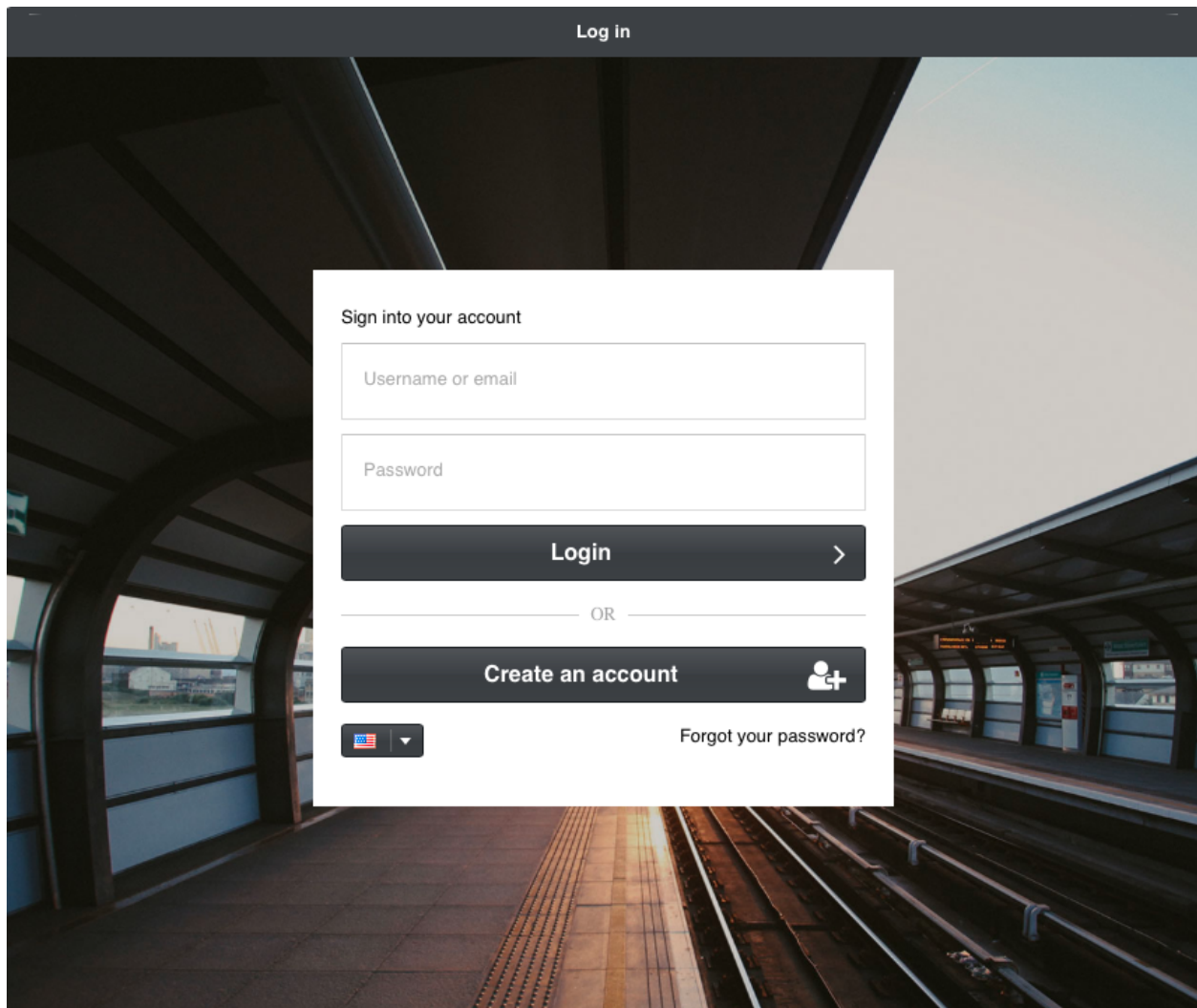
1. Execute the following commands as root to run the script that will install MagnusBilling, Asterisk and all dependencies needed like: IPTables, Fail2ban, Apache, PHP and MySQL.

Install CentOS 7 *minimal*.

```
cd /usr/src/  
yum -y install wget  
wget https://raw.githubusercontent.com/magnussolution/magnusbilling7/source/script/  
↪install.sh  
chmod +x install.sh  
./install.sh
```

2. During the install you'll be asked what language MagnusBilling should use. Choose by typing the number of the language.

```
Install complete. The server will restart automatically..  
  
Use a browser to access the interface.  
Go to: http://000.000.000.000  
User: root  
Password: magnus (Remember to change the password)
```

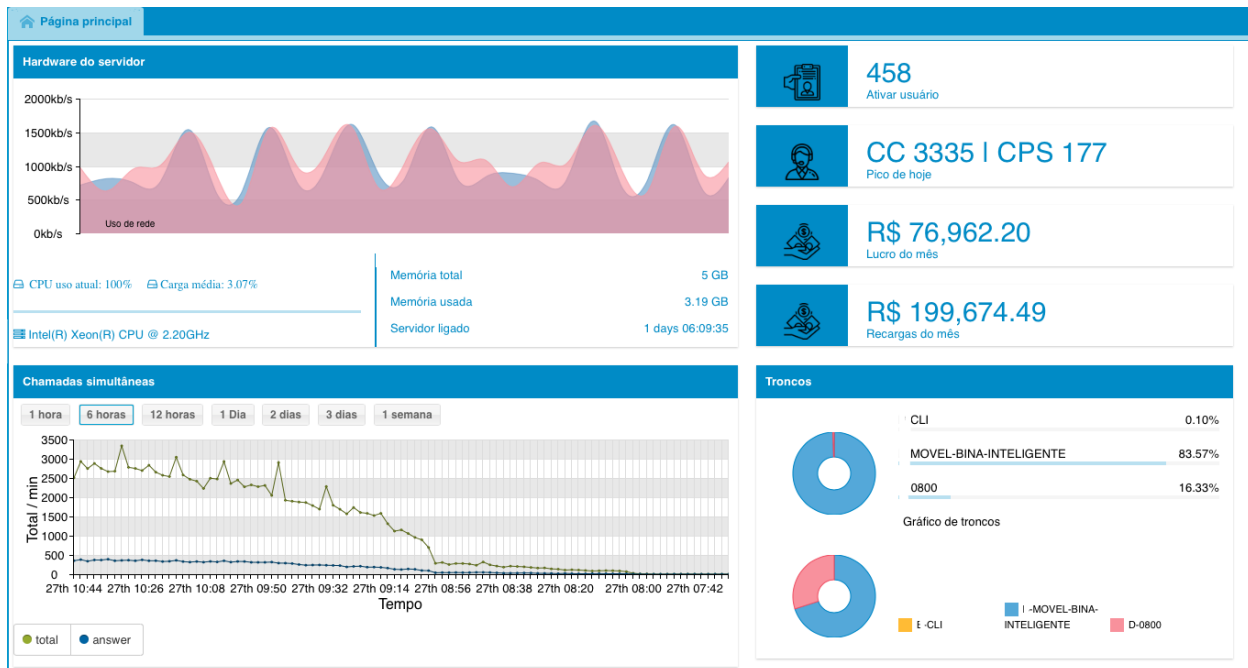


MagnusBilling's interface is made with ExtJS, and is composed by a lateral menu which has all the modules listed with their respective submodules. A main area, where all the modules are opened at, with active and inactive tabs. A suspended menu with the language options, change password, about and leave.

5.1 Monitoring Panel

In the main page we have some graphs and information about the server. This bar is only shown in the Home tab, server status bar.

- Processors model;
- Current processor usage;
- Average processor usage;
- Server total memory;
- Memory in use;
- Link upload and download usage graph;
- Time that the server is online;
- Active users;
- Simultaneous calls and CPS(Calls per second) peak;
- Profit from the month;
- Recharges held in the month.
- Simultaneous Calls graph, with the “show” option last hour, 6 hours, 12 hours, 1 day, 2 days, 3 days or 1 week;
- Graph of the 3 most used trunks.



5.2 Button bar

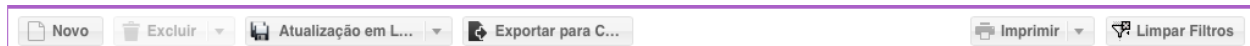
MagnusBilling is a modular system, each menu or submenu corresponds to a module. All the modules are identical in their structure. Each module is composed of one bar of tools composed by this buttons:

Modules buttons bar

- **Add Button.** Adds a new record to the module.
- **Edit Button.** Edits one record, only clicking on top of a register, already open from the formulary to edit.
- **Delete Button.** Selection one or more records and clicking in Delete, will be requested the confirmation, when confirmed the record will be deleted. The delete button have a property called All, found when clicked on top of the arrow besides the button, this property deletes all the records

listed in the module. A very useful tool to delete a large amount of data. OBS: MagnusBilling databank uses Inn-oDB with foreign key, this links every data, therefore is not possible to delete data that are already linked with others. EX: Is not possible to delete an client, if this client made an purchase. Will be necessary to delete all the records linked to this client to be able to delete this client.

- **Batch Update button.** This button is fundamental to the server efficiency. This button is used, as the names suggests, is to update a large amount of records at the same time, making it easy one taks that would be impossible to be done manually in modules with a lot of records, with the tariff module that can easily surpass the 100 thousand records. It's use is pretty simple, search the records that you want to update and click the Batch Update button, will open a formulary with all it's fields blank, just change the fields that you want to update, letting the rest of them in blank, and then press save, this process can take a few seconds, and will update all the records. Batch Update button have a property called Selected, found when clicked on the arrow besides the button, this property will only update the selected records.
- **Print button.** This button generates a Excel file in CSV format of the selected records.
- **Print button.** This button generates a PDF file of the selected records.
- **Clear Filters Button.** This button clears used filters.



5.3 Columns

Each columns in MagnusBilling have nemerous options, very useful for a good use of the system.

To find these options put your mouse pointer on top of the desired column, then press the arrow.

- **Ascending and Descending order.** Classifies the records of the selected column in a ascending or descending manner. This option can be used when clicked on top of the column as well.
- **Columns.** With this option you can show or hide columns, very useful to shown only necessary data. Example, to generate a price table to send to a client including only the name of destination and the sales price.
- **Group by this Column.** Group the records by this column, very useful for the Online Calls modules, making it possible to group the calls per client. To be able to see individually the calls of one client or any other column.
- **Show in groups.** This option is used to disable the previous option.
- **Filter.** This option is fundamental as well, therefore will be written a dedicated tutorial for it.

<input type="checkbox"/>	Nome	Descrição	Crédito	Data de criação
--------------------------	------	-----------	---------	-----------------

5.4 Filter System

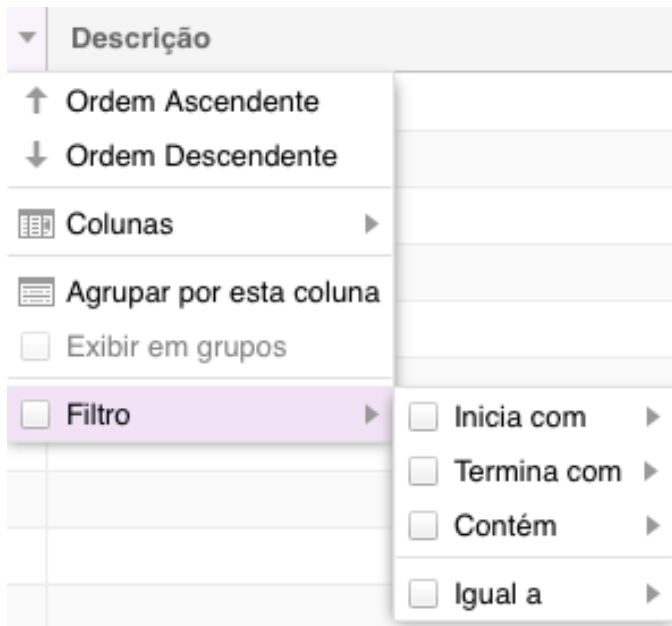
MagnusBilling counts on a powerful filter system to find desired records in each module.

To find this option put your mouse pointer on top of the column that you want to make search, press in the arrow, slide it until Filter, will open a sub-menu containing the option for the column, keep sliding the mouse pointer until the field and put the value that you want to search, then press enter. OBS: The filter system is Case Sensitive.

Exists different filters depending on the type of the column;

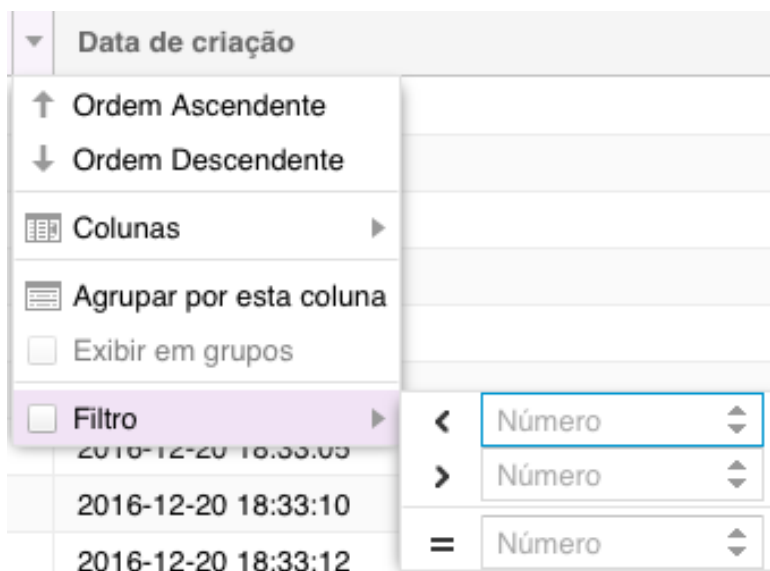
- Text filter.

This filter search for data in the columns with a text that begins with, ends with, contains or that is equal to. For example , if you search for the name that begins with “Peter”, will be listed all the clients that have Peter in the begining of their names.



- Whole numbers filter.

This filter show in columns that contains numeric numbers, like the client credit column.



- Filter by dates.

This filter show the columns that contains dates, this filter is really useful in the reports.

Data de criação	
2016-05-25 09:16:12	↑ Ordem Ascendente
2016-08-25 10:52:16	↓ Ordem Descendente
2016-08-25 10:52:20	Colunas
2016-12-16 18:32:26	Agrupar por esta coluna
2016-12-20 18:33:00	<input type="checkbox"/> Exibir em grupos
2016-12-20 18:33:00	<input type="checkbox"/> Filtro
2016-12-20	<input type="checkbox"/> Antes
2016-12-20	<input type="checkbox"/> Após
2016-12-20	<input type="checkbox"/> Em

- Combo filter.

This filter is used in cases of modules relationships.

Plano	Grupo	Revendedor	Estado
agent1			Ativo
agent1			Ativo
agent1			Ativo
magnus			Ativo
agent1			Inativo
tesdcscds			Ativo
agent1	Client	root	
agent1	Client para...	root	
agent1	Client	root	
agent1	Client	root	
agent1	Client	root	
agent1	administra...	root	
agent1	Client	root	
agent1	Client	root	
agent1	Client	root	
agent1	Client	root	

To undo an filter, remove the selection of filter menu. And to clean all the aplicable filters in a module, click on the

Clean Filters button.

It's always a good idea to have a backup.

The backup don't saves the rejected calls table, since normally it's enourmous.

6.1 Manual Backup

The project already has a script to do backups of the Databank and Asterisk files. In the installation is already added an script in the Linux crontab to perform one backup per day. Default is set to 02:00 am.

6.2 Manually

Execute this command in SHELL of your server.. `php /var/www/html/mbilling/cron.php Backup`

6.3 Crontab

Setting up crontab -e

```
crontab -e
```

Search the line below and change the time as you see fit, or only comment in the line with ; to not make automated backups.

```
0 2 * * * php /var/www/html/mbilling/cron.php Backup
```

6.4 Backup Menu

It's possible to view, download and delete backups via the Backup menu as well. The menu is located in the settings.

CHAPTER 7

Update MagnusBilling

Our team works daily to add new functions and solve problems. You can see the changes made in the link <https://github.com/magnussolution/magnusbilling7/commits/source>.

Therefore, our team recommends that you keep your server up to date.

To update run the following command.

```
/var/www/html/mbilling/protected/commands/update.sh
```

Making your first call

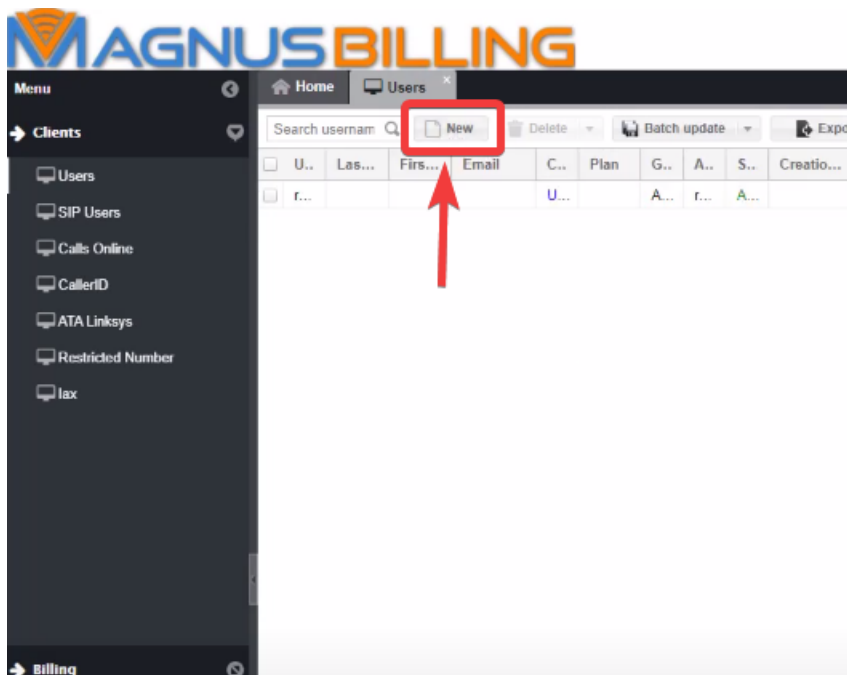
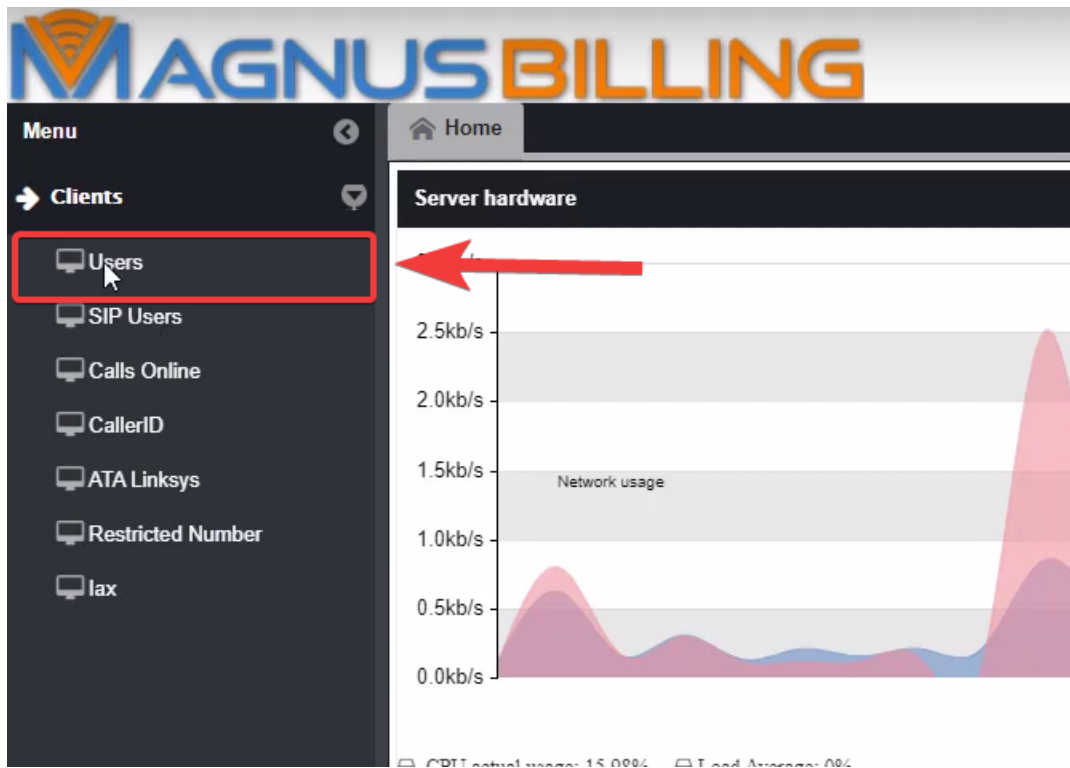
This will be a guide to show you how to configure MagnusBilling to make your first call.

If you prefer a guide in video form, you access this link: <https://www.youtube.com/watch?v=GBm424z5KSE>

8.1 Guide

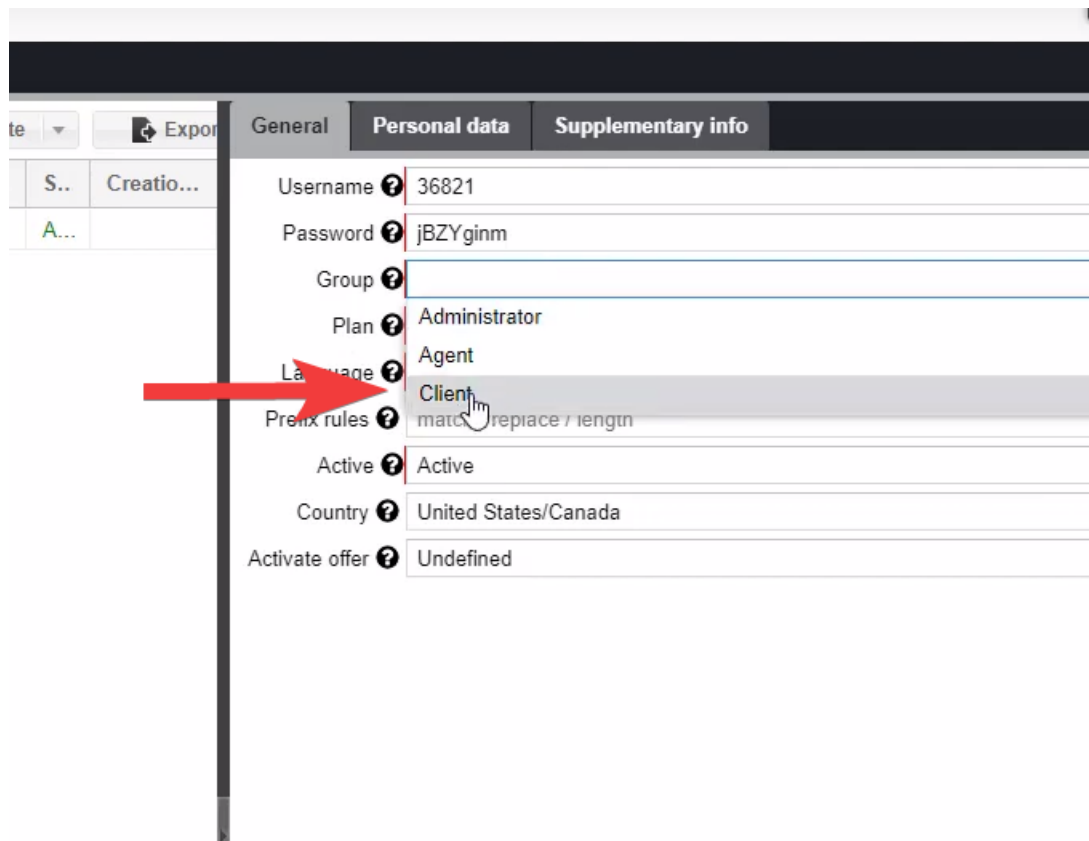
First, you need to login to MagnusBilling, the default username and password is " root " and " magnus ".

After login in, the first thing you need to do is create an User. To do this, you need to click the Clients tab, Users, New.



A new User will be created, the username and password will be created automatically.

Now you need to select the group of the user. In this case we are creating an client, so select the client option.



General	Personal data	Supplementary info
te ▼	Export	
S..	Creatio...	
A...		
Username ?	36821	
Password ?	jBZYginm	
Group ?		
Plan ?	Administrator	
Language ?	Client	
Prefix rules ?	match / replace / length	
Active ?	Active	
Country ?	United States/Canada	
Activate offer ?	Undefined	

We have no plans created, so let's create one so we assign one plan to the client. In order to do that, first click the Rates tab, Plans, New.

For this guide we will only name the plan Golden, and not change any other of the settings.

Back at the Users tab, you select the newly created plan in the search button.

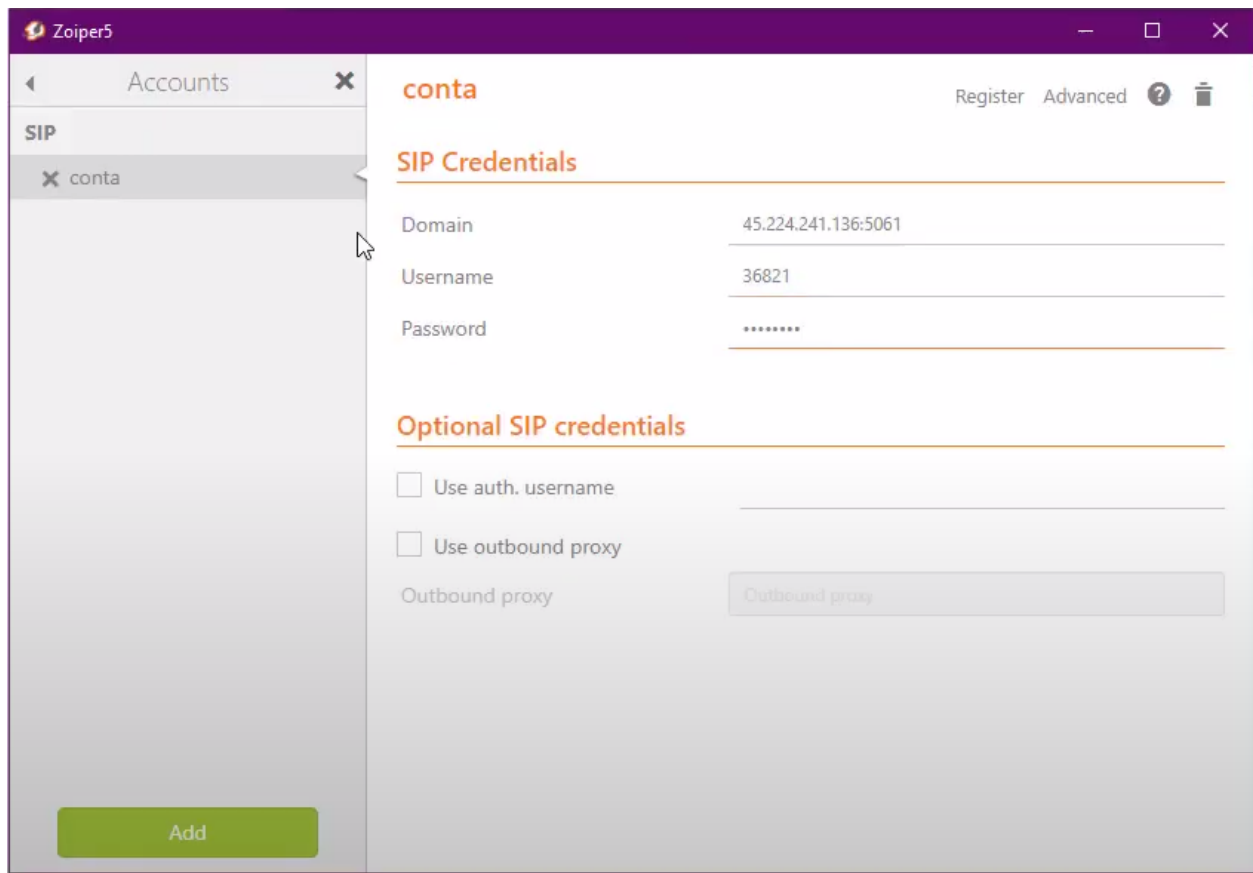


General	Personal data	Supplementary info
Username ?	36821	
Password ?	jBZYginm	
Group ?	Client	
Plan ?		
Language ?	English	
Prefix rules ?	match / replace / length	
Active ?	Active	
Country ?	United States/Canada	
Activate offer ?	Undefined	

After successfully creating an user, MagnusBilling will automatically create a Sip account for this user.

Now we are going to use Zoiper software (you can use your preferred software).

In Zoiper configure the SIP credentials accordingly to the user we just created. In the domain field put your server IP address. *Note: After the server ip (:5061) is the Asterisk Port that in this example is set up for 5061. The default MagnusBilling port is 5060. Also, since 5060 is the default SIP port, you can just use your IP address.



The screenshot shows the Zoiper5 web interface. On the left, a sidebar menu has 'Accounts' selected, and a sub-menu shows 'SIP' with 'conta' highlighted. The main content area is titled 'conta' and contains two sections: 'SIP Credentials' and 'Optional SIP credentials'. The 'SIP Credentials' section has three input fields: 'Domain' with the value '45.224.241.136:5061', 'Username' with the value '36821', and 'Password' with masked characters '*****'. The 'Optional SIP credentials' section has two checkboxes: 'Use auth. username' and 'Use outbound proxy', both of which are unchecked. Below these checkboxes is a disabled 'Outbound proxy' input field. At the bottom left of the sidebar is a green 'Add' button. At the top right of the main content area are links for 'Register', 'Advanced', a help icon, and a trash icon.

If the setup is done correctly, the SIP user status will change to OK.

Now we need to create a Provider. To do this, go to the Routes tab, Providers. You can just name it and click save.

And after that we need to create a Trunk, fill the fields of the trunk with the credentials. Furthermore we need to create an Trunk Group.

Lastly, we need to create an Prefix and a Tariff. Located in the Rates tab. They are pretty straight forward, so configure it to your needs.

Going to back to Zoiper. If we try to make an call, we are going to receive an warning that your balance is empty. To fix that you need to go to the Billing Tab, refills. And create a new refill to the user we created.

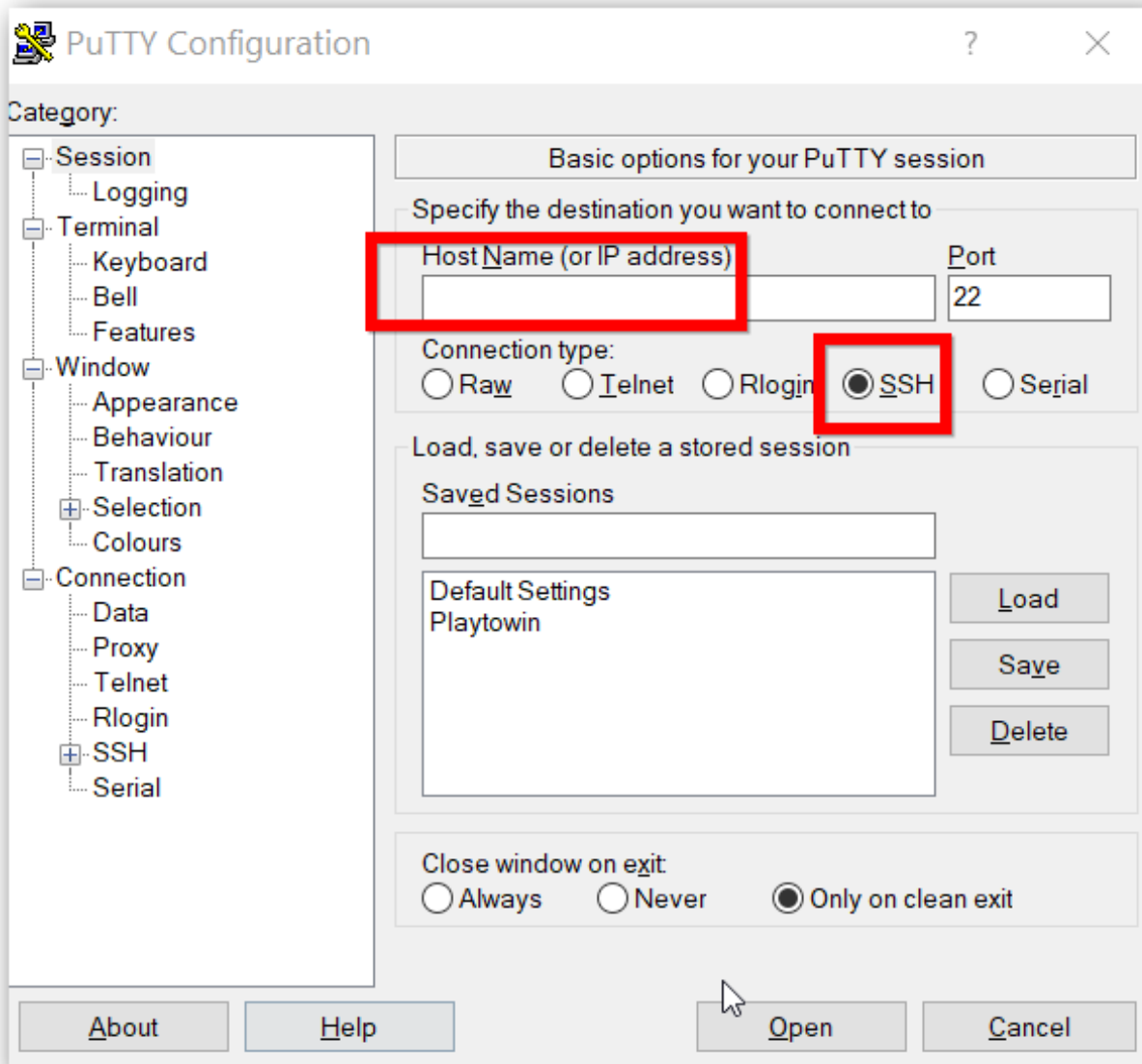
If everything was done correctly, you can make your first call.

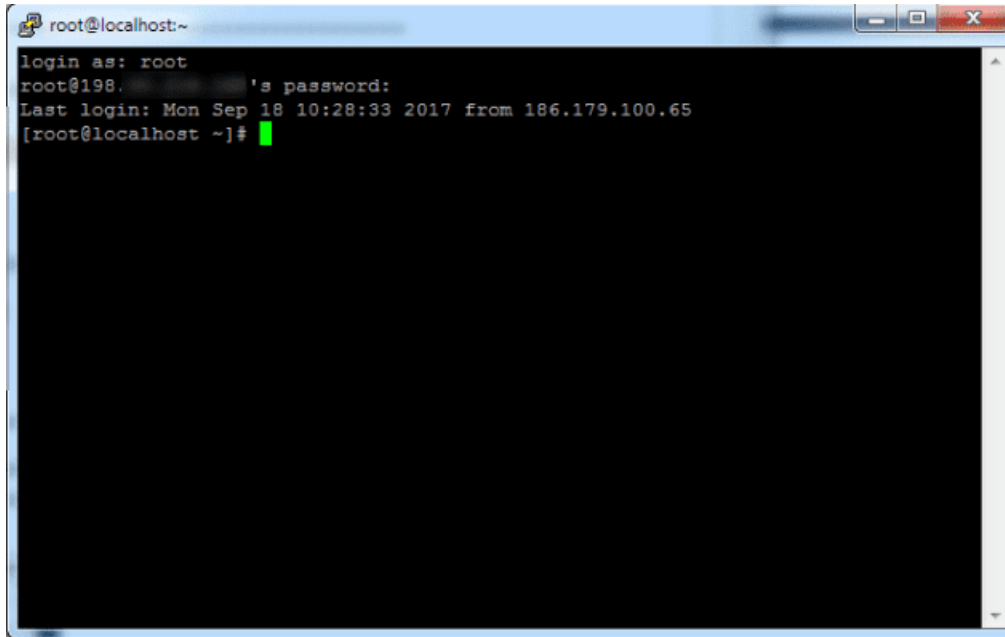
Recover Magnusbilling password

Recover MagnusBilling root password. To recover or change your root password with MagnusBilling, we will basically make some modifications directly to the MagnusBilling MySQL / MariaDB database. You can do this through the console as in this tutorial or through phpMyAdmin (in this case, you can use the same MySQL commands presented in the last steps of this tutorial).

1. JOIN AS ROOT IN THE SSH OF THE MAGNUSBILLING SERVER..

Using an SSH client, for example Putty, we enter the terminal or shell of the MagnusBilling server. We must put the IP or domain of the server and then enter root as user and its password. If you use a user other than root, login with your username, and using the sudo command, log in as root





2. ENTER AS ROOT IN MYSQL DATABASE.

Now we enter the following command:

```
mysql -u root -p mbilling
```

By default the MagnusBilling install script saves the mysql root password in the `/root/passwordMysql.log` file, so you can see the mysql root password with the command

```
cat /root/passwordMysql.log
```

```
[root@localhost ~]# mysql -u root -p
Enter password:
Welcome to the MariaDB monitor.  Commands end with ; or \g.
Your MariaDB connection id is 21584
Server version: 5.5.56-MariaDB MariaDB Server

Copyright (c) 2000, 2017, Oracle, MariaDB Corporation Ab and others.

Type 'help;' or '\h' for help. Type '\c' to clear the current input statement.

MariaDB [(none)]>
```

3. MODIFY THE PASSWORD DIRECTLY IN THE DATABASE.

After entering the password, we find the MySQL monitor, where we will place the commands to change the password from the MagnusBilling admin panel and regain access. Alternatively, you can run these commands in phpMyAdmin if you have this tool on your server.

Note that MagnusBilling uses the SHA1 algorithm to encrypt the password of administrator users.

To enter a new password, since we won't know the current one as it is encrypted, we do the following. In this example we will reset it to the default password that comes with a new installation of MagnusBilling, which is `magnus`. But instead, you can put whatever you want, including numbers, letters, symbols.

```
UPDATE pkg_user SET password = SHA1("magnus") WHERE username = "root";
```

After running this command, your MagnusBilling admin password is reset, and you can log in to your dashboard normally.

CHAPTER 10

Call Price Calculation

This value is calculated based on the adjusted price for the provider which trunk was used to complete this call.

The price is calculated using 4 values.

- Call duration;
- Purchase price;
- Initial block;
- Increment.

Example.

The client calls to the number 5511988443300 and spoke for 45 seconds. And exist one tariff on the Providers Tariff menu for the 55119 prefix, with:

- Purchase price = 0.05
- Initial block = 30
- Increment = 6

The calculation formula is:

If the call durations is inferior than the Initial block value the formula will be

```
(Purchase price / 60 seconds) * Initial block
```

If the call durations is higher than the Initial block value the formula will be

```
(Purchase price / 60 seconds) * Call duration
```

When the call duration is higher than the Initial block, the Increment is calculated as well.

In the last example, the duration was 45 seconds, and the Increment is 6, so the formula become

```
(Call duration / Increment ) the result is 7, with 3 as remainder. Then we need to  
→ add up 3 to the rest, the result is 48. Therefore in this example will be calculated
```

```
(0.05 / 60) * 48 = 0.04
```

The code below is used for the calculus.

```
$cost = 0;
if ($callduration < $initblock) {
    $callduration = $initblock;
}

if (($billingblock > 0) && ($callduration > $initblock)) {
    $mod_sec = $callduration % $billingblock;
    if ($mod_sec > 0) {
        $callduration += ($billingblock - $mod_sec);
    }
}
$cost += ($callduration / 60) * $rateinitial;
```

The variable \$callduration is the call duration; The variable \$initblock is the Initial block; The variable \$billingblock is the Increment.

11.1 How does MagnusBilling select the rate to be used?

When a customer makes an external call, to a landline or cell phone, a verification routine is triggered which consists of:

1. Check the username and password.
2. Checks whether the client is active.
3. Checks if the number dialed is a DID.
4. Check if the dialed number is a SIP Account.
5. Check the customer's plan.
6. Search within the plan's tariffs which fare best fits the number dialed.
7. Search for the trunks that belong to the trunk group of the tariff found.
8. Sends the call to the first trunk, and if it fails it sends to the next.
9. When the call is finished, the purchase and sale price is calculated.
10. Added the call in the CDR report.
11. Discounting the value of the call in the customer's credit.

Rather, how does step 6 work?

It is carried out as follows. Let's say customer XXXXX called 551140045678

In step 5, the system takes the customer's plan, and then in step 6 a "SELECT" is made on the tariff table like this.

```
$ sql = "SELECT * FROM pkg_plan LEFT JOIN pkg_rate ON pkg_plan.id = pkg_rate.id_plan
LEFT JOIN pkg_prefix ON pkg_rate.id_prefix = pkg_prefix.id
LEFT JOIN pkg_trunk_group ON pkg_trunk_group.id = pkg_rate.id_trunk_group
WHERE pkg_plan.id = $ MAGNUS-> id_plan AND pkg_rate.status = 1
AND $ MAGNUS-> prefixclause ORDER BY LENGTH (prefix) DESC LIMIT 1 ";
```

The \$ MAGNUS-> id_plan variable is the customer's plan id, and the \$ MAGNUS-> prefixclause variable is the filter used to find the tariff.

To facilitate understanding we will explain it in another way.

As the customer called 551140045678, the system will search as follows.

Is there a tariff for the prefix 551140045678?

- If yes, we use this rate
- If not, we will search again by removing the last number, in this case 55114004567 and so on until we reach only number 5. If we do not find any rate, the system will return an error and will not continue the call.

Going back to "SELECT" it would look like this.

```
$ sql = "SELECT * FROM pkg_plan LEFT JOIN pkg_rate ON pkg_plan.id = pkg_rate.id_plan
LEFT JOIN pkg_prefix ON pkg_rate.id_prefix = pkg_prefix.id
LEFT JOIN pkg_trunk_group ON pkg_trunk_group.id = pkg_rate.id_trunk_group
WHERE pkg_plan.id = 1 AND pkg_rate.status = 1 AND
(prefix = '551140045678' OR
prefix = '55114004567' OR
prefix = '5511400456' OR
prefix = '551140045' OR
prefix = '55114004' OR
prefix = '5511400' OR
prefix = '551140' OR
prefix = '55114' OR
prefix = '5511' OR
prefix = '551' OR
prefix = '55' OR
prefix = '5')
ORDER BY LENGTH (prefix) DESC LIMIT 1 ";
```

Let's say you have the following 3 tariffs in your customer's plan

55, Brasil Fixo Geral 55119, Brazil SP Celular 5511, Brazil SP Fixed

In this example, "SELECT" would return tariff 5511, Brasil SP Fixo.

But you may be wondering if the customer calls 5511988551234, the "SELECT" would find the prefix 55119 and 5511 and then which one would use? I would use 55119, because in "SELECT" there is the parameter ORDER BY LENGTH (prefix) DESC and also the parameter LIMIT 1. That is, it will order the results by placing first the prefix that has the largest number of digits.

CHAPTER 12

TTS Configuration

MagnusBilling supports TTS via URL, next some tested providers and their respective configurations.

Reminding that the URL variables, independently of the provider, must be edited conform your necessities and with the right API according to the provider, the only part relating MagnusBilling is the variable \$name, which is where MagnusBilling will replace the registered names in the Numbers menu.

12.1 Vocalware

The first step to set up is to create your account in this site <https://www.vocalware.com/> Now we need to configure the API URL in the Settings menu -> Adjustments sub menu. Locate the option TTS URL and change the URL to:

https://www.vocalware.com/tts/gen.php?EID=3&LID=6&VID=1&TXT=\protect\T1\textdollaname&EXT=mp3&FX_TYPE=&FX_LEVEL=&ACC=YOUR_ACC&API=YPUR_API&SESSION=&HTTP_ERR=&CS=&SECRET=YOUR_SECRET

In the example above, is already configured in Brazilian Portuguese.

EID=3 is Engine ID = 3 LID=6 is Portuguese VID=1 is Helena's voice

12.2 Voicerss

The first step to set up is to create your account in this site: <http://www.voicerss.org> Now we need to configure the API URL in the Settings menu -> Adjustments sub menu. Locate the option TTS URL and change the URL to:

http://api.voicerss.org/?key=YOUR_API&hl=pt-br&src=\protect\T1\textdollaname&f=8khz_16bit_mono

In this company you only need to put your API. In this example is already configured in Brazilian Portuguese.

12.3 Google

You can look for more information about Google TTS in this link: https://cloud.google.com/text-to-speech?hl=pt_br

The Google URL is:

https://translate.google.com/translate_tts?ie=UTF-8&q=\protect\T1\textdollarname&tl=pt-BR&total=1&idx=0&textlen=5&client=tw-ob&tk=\protect\T1\textdollartoken

MagnusBilling will work like any other TTS provider that accepts the integration via URL.

This service can be charged by the provider.

13.1 What is free packages?

Free packages are for giving customers a number of calls at no charge, for a monthly fee or with no cost whatsoever.

13.2 How to configure?

The configuration of the free packages needs several steps.

1 - Create the package.

See the list of descriptions for each field:

- offer-label
- offer-packagetype
- offer-freetimetocall
- offer-billingtype
- offer-price

2 - Select the tariffs that you want to include in the packages.

It's necessary to inform which prefixes will be included in the free packages. To do this, go to the tariffs menu in the tariffs submenu, click on the tariff you want to include in the packages and put the option "Include in free packages" instead.

3 - Activate the free packages for users. Go to the Customers menu, users submenu, click on the customer who wants to activate the package, and select the option "Activate free package" the package you want to activate for the customer.

13.3 How it works?

We will use the following example:

A free package called SPAIN FIXED was created, with 6000 free seconds, that is 100 minutes, charging monthly and with the price of USD 5,00.

It was configured to include in the free packages the Spain fixed tariffs, of the Gold plan.

And the free package was activated on the 24315 user, and this customer was placed on the Gold plan.

And we left this client 24315 with a balance of USD 10.00.

Process performed by the system.

Customer 24315 calls the number 551140040001, the system will check the customer's plan, and will soon search for the most appropriate rate for the number dialed, in this case it will be the rate 55114, São Paulo Prefix.

Now it will check if this tariff has the option "Include in free packages", if it does, and in our example it does, MagnusBilling will check if the customer has already used 100 minutes from the day of activation, considering if the type of charging is monthly or week, in our example it is monthly. If the 100 minutes have not yet been used, the system will allow the customer to call even if the customer's balance is 0. And if the customer has a balance, the call will be free.

If the tariff is included in the packages, but the customer has already exceeded the package limit, MagnusBilling will only allow the call if the customer has credit.

Continuing with our example, if the customer dials any other number that does not start with 55114, he can only call if he has credit. In our Gold plan, we only select the precise 55114 prefix to include in the packages.

13.4 How is the monthly fee charged?

Using the same example as before, and that the plan was activated on the 15th, the system will automatically charge the customer for the value of the package, in this example USD 5.00 from the customer's credit every 15th.

There is an option in the settings submenu settings menu called "Notification of Offer Packages", by default it is 5 days. This means that 5 days before the expiration date of the plan, MagnusBilling will try to charge the value of the package, in our example it would be on the 10th.

If the customer does not have enough credit for MagnusBilling to discount the USD 5.00, an "plan_unpaid" email will be sent.

Magnusbilling will try to charge the value of the free package until it succeeds , or until the package expires..

If the customer places credit, or already has enough credit for payment, MagnusBilling will mark the plan as paid for another month, and create a refill in the plan price, and will send the "plan_paid" email.

If it arrives on the 16th and the customer still does not have enough credit for the payment, the plan will be deactivated in the user's account, and an "plan_released" email will be sent.

The emails can be found, and edited, in the settings menu submenu email templates.

13.5 Where to see the consumption of each customer?

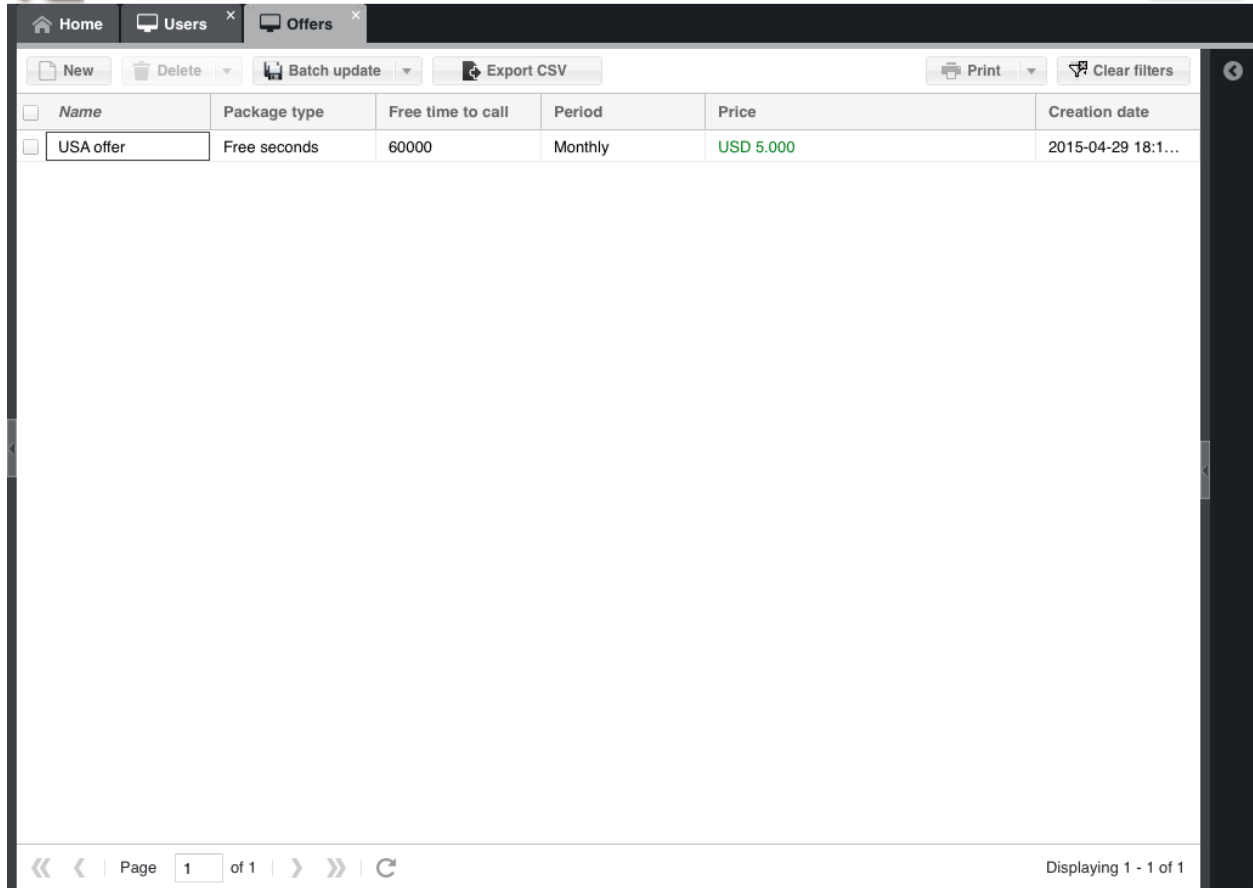
All calls made using a package will be added to the tariff menu submenu package report.

OBS: * It is not possible to activate more than one package per customer. * It is not possible to create combined packages, for example, 100 minutes for fixed, and 50 minutes for mobile. * Calls made using packages will always be rounded up to minutes, regardless of the minimum time and block of tariff time.

13.6 Example image

See some images of the configuration of our example.

The free package.



<input type="checkbox"/>	Name	Package type	Free time to call	Period	Price	Creation date
<input type="checkbox"/>	USA offer	Free seconds	60000	Monthly	USD 5.000	2015-04-29 18:1...

Tariffs.

Home

Tariffs

Search prefix:

New

Delete

Batch update

Export CSV

Import CSV

Print

Clear filters

<input type="checkbox"/>	Prefix	Destination	Sell price	Initial bl...	Billing ...	Trunk groups	Plan	Include in offer
<input type="checkbox"/>	1250232	Canada Northwest Territories	R\$ 0.005	1	1	teste2	Ouro	No

This column is hidden

«

«

Page

1

of 1

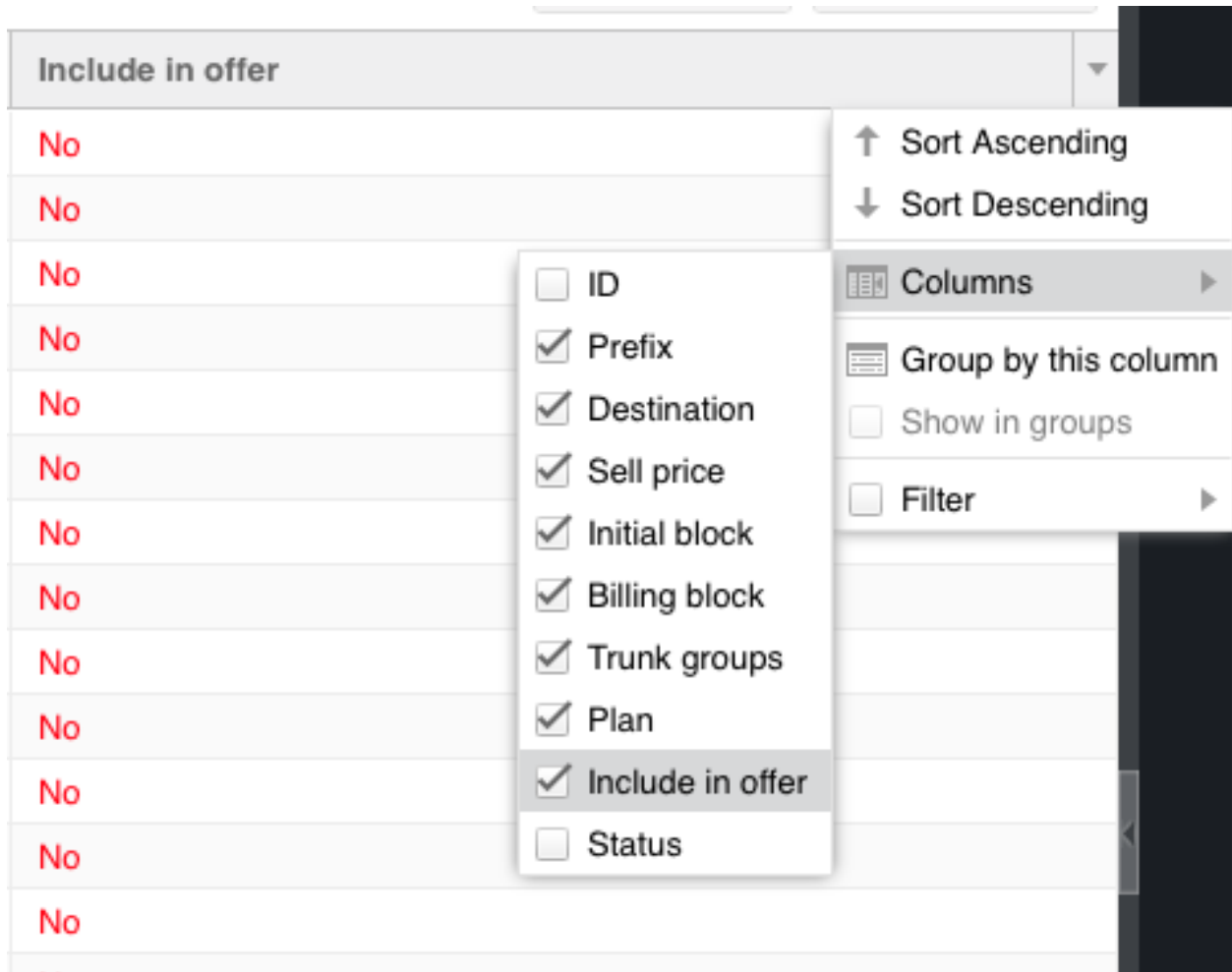
»

»

↺


Displaying 1 - 1 of 1


Show the Include in free packages column.




User activation

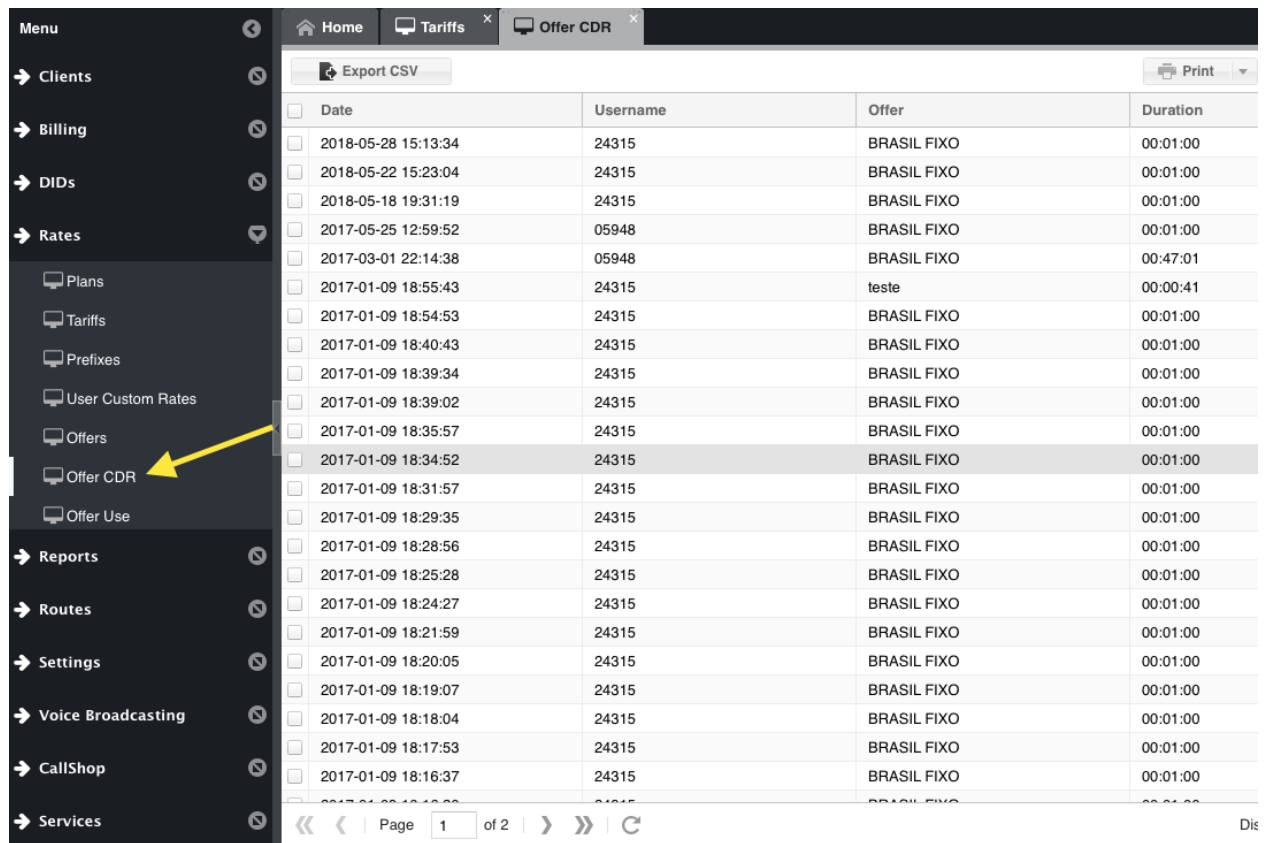
General	Personal data	Supplementary info
Username: 24315		
Password: *****		
Group: Client		
Plan: Wholesale		
Language: English		
Prefix rules: match / replace / length		
Active: Active		
Country: Brazil		
Activate offer: USA offer		

 Resend Email

 Save

 Cancel

Report



The screenshot displays the MagnusBilling web interface. On the left is a dark sidebar menu with various navigation options. The main content area at the top has tabs for 'Home', 'Tariffs', and 'Offer CDR'. Below the tabs, there's a table with columns 'Date', 'Username', 'Offer', and 'Duration'. A yellow arrow points to the 'Offer CDR' option in the sidebar menu. The table contains 20 rows of data, mostly for 'BRASIL FIXO' offers. At the bottom, there's a pagination bar showing 'Page 1 of 2' and a 'Dis' label on the right.

Menu

- Clients
- Billing
- DID's
- Rates
 - Plans
 - Tariffs
 - Prefixes
 - User Custom Rates
 - Offers
 - Offer CDR**
 - Offer Use
- Reports
- Routes
- Settings
- Voice Broadcasting
- CallShop
- Services

Home Tariffs Offer CDR

Export CSV Print

<input type="checkbox"/>	Date	Username	Offer	Duration
<input type="checkbox"/>	2018-05-28 15:13:34	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2018-05-22 15:23:04	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2018-05-18 19:31:19	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-05-25 12:59:52	05948	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-03-01 22:14:38	05948	BRASIL FIXO	00:47:01
<input type="checkbox"/>	2017-01-09 18:55:43	24315	teste	00:00:41
<input type="checkbox"/>	2017-01-09 18:54:53	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:40:43	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:39:34	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:39:02	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:35:57	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:34:52	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:31:57	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:29:35	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:28:56	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:25:28	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:24:27	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:21:59	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:20:05	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:19:07	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:18:04	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:17:53	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:16:37	24315	BRASIL FIXO	00:01:00
<input type="checkbox"/>	2017-01-09 18:16:37	24315	BRASIL FIXO	00:01:00

« ‹ Page 1 of 2 › » ↺

Dis

Email Templates

Home

Users

Offers

Emails Templates

Clear filters

Type	Status	Langua...	Subject
<input type="checkbox"/> plan_unpaid	Active		Aviso de Vencimento do Plano Mensal
<input type="checkbox"/> plan_paid	Active		Pagamento do plano recebido
<input type="checkbox"/> plan_released	Active		Plano cancelado
<input type="checkbox"/> plan_unpaid	Active		Aviso de Vencimiento del Plan Mensual
<input type="checkbox"/> plan_paid	Active		Confirmación de pago del plan mensual rec...
<input type="checkbox"/> plan_released	Active		Su plan fue dado de baja.
<input checked="" type="checkbox"/> plan_unpaid	Active		Balance Due Alert for your Monthly plan
<input type="checkbox"/> plan_paid	Active		Confirmation of call plan payment
<input type="checkbox"/> plan_released	Active		Your call plan has been suspended

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Page 1 of 1

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Displaying 1 - 9 of 9

General

Email body

Variables

Helvetica

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☰

Dear \$firstname\$ \$lastname\$.

We are sending you this message to notify that your call plan is approaching the due date and your account does not have enough credit balance to make the payment.

Please, login in you account area, select: Menu, Buy Credits, and recharge your account with at least \$plancost\$ \$base_currency\$.

You can login the client are with the following data:

http://your_ip/mbilling

Username=\$login\$

Password=\$password\$

Please recharge your account or your call plan will be deactivated automatically on the due date.

Greetings

Save

Cancel

CHAPTER 14

How to use VOUCHER

To use a Voucher, the first thing to do is to create the Vouchers in the financial menu, Voucher submenu.

See the list of descriptions for each field:

- voucher-credit
- voucher-id-plan
- voucher-language
- voucher-prefix-local
- voucher-quantity
- voucher-tag
- voucher-voucher

After generating the new Vouchers, it will be necessary to add permissions to the customer group so that they can view and create the Voucher, as shown in the image below.

Group Users

General

Permissions

Module	Menu	Create	Update	Delete	Desktop	Quick init.
▶ <input checked="" type="checkbox"/> → Clients	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▼ <input checked="" type="checkbox"/> → Billing	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Refills	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
<input checked="" type="checkbox"/> Payment Methods	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input checked="" type="checkbox"/> Voucher	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/> Refill Providers	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
<input type="checkbox"/> Fatura	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input checked="" type="checkbox"/> → DIDs	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input checked="" type="checkbox"/> → Rates	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input checked="" type="checkbox"/> → Reports	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input type="checkbox"/> → Routes	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input type="checkbox"/> → Settings	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input checked="" type="checkbox"/> → Voice Broadcasting	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input type="checkbox"/> → CallShop	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
▶ <input checked="" type="checkbox"/> → Services	<input checked="" type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Save

Cancel

There are three ways to use a voucher.

1 - Through the panel of an existing customer.

Now customers in possession of a Voucher that has not been used, and who belong to a group of customers who are allowed to view and create a Voucher, will be able to log in to their customer panel, go to the Voucher menu, click on new, and place the Voucher. Once everything is correct with the voucher, the system will release the credit to the user.

2 - Through a DID.

Create a DID and leave it without destination, or with destination Calling Card. When calling the DID, you will be asked for a PIN number, so enter the voucher number. A new user will be created, with the Voucher settings, and credit will be added to the account.

3 - Call from a SIP account to the code * 120

When calling the code * 120 and entering the Voucher number when the PIN is requested, the value of the Voucher will be added to the user.

A recharge will be created for the customer with the Voucher credit value in the 3 previous options.

In the administrator's Voucher menu, the user who used the voucher and the date of use will appear.

Applicable Iptables rules in the installation.

15.1 Basic Rules

```
iptablesF
iptablesA INPUT -p icmp --icmp-type echo-request -j ACCEPT
iptablesA OUTPUT -p icmp --icmp-type echo-reply -j ACCEPT
iptablesA INPUT -i lo -j ACCEPT
iptablesA INPUT -m state --state ESTABLISHED,RELATED -j ACCEPT
iptablesA INPUT -p tcp --dport 22 -j ACCEPT
iptablesP INPUT DROP
iptablesP FORWARD DROP
iptablesP OUTPUT ACCEPT
iptablesA INPUT -p udp -m udp --dport 5060 -j ACCEPT
iptablesA INPUT -p udp -m udp --dport 10000:20000 -j ACCEPT
iptablesA INPUT -p tcp -m tcp --dport 80 -j ACCEPT
```

15.2 Optional Rules

```
OPENVPN: iptables -A INPUT -p udp --dport 1194 -j ACCEPT
ICMP: iptables -A INPUT -p icmp --icmp-type echo-request -j ACCEPT
IAX: iptables -A INPUT -p udp -m udp --dport 4569 -j ACCEPT
HTTPS: iptablesA INPUT -p tcp -m tcp --dport 443 -j ACCEPT
```

15.3 Friendly Scanner

Rules to block not so friendly scanner

```
iptables -I INPUT -j DROP -p tcp --dport 5060 -m string --string "friendly-scanner" --  
↪ algo bm  
iptables -I INPUT -j DROP -p tcp --dport 5080 -m string --string "friendly-scanner" --  
↪ algo bm  
iptables -I INPUT -j DROP -p udp --dport 5060 -m string --string "friendly-scanner" --  
↪ algo bm  
iptables -I INPUT -j DROP -p udp --dport 5080 -m string --string "friendly-scanner" --  
↪ algo bm
```

Optional

```
iptables -I INPUT -j DROP -p tcp --dport 5060 -m string --string "VaxSIPUserAgent" --  
↪ algo bm  
iptables -I INPUT -j DROP -p udp --dport 5060 -m string --string "VaxIPUserAgent" --  
↪ algo bm  
iptables -I INPUT -j DROP -p udp --dport 5080 -m string --string "VaxSIPUserAgent" --  
↪ algo bm  
iptables -I INPUT -j DROP -p tcp --dport 5080 -m string --string "VaxIPUserAgent" --  
↪ algo bm
```

15.4 Show iptable rules

```
sudo iptables -L -v
```

15.5 Show line numbers

```
iptables -L -v --line-numbers
```

15.6 Delete a line

Delete line 2

```
iptables -D INPUT 2
```

15.7 Block IP address

```
iptables -I INPUT -s 62.210.245.132 -j DROP
```

15.8 Save Changes

Centos

```
service iptables save
```

Debian / Ubuntu

```
apt-get install iptables-persistent  
service iptables-persistent save  
dpkg-reconfigure iptables-persistent
```


CHAPTER 16

Menu Clients

This is the list of all fields with their description of the menu Clients

16.1 Users

This menu has the following fields

16.1.1 Username

Username used to login into the panel.

16.1.2 Password

Password used to login into the panel.

16.1.3 Group

There are 3 groups: admin, agent and client. You can create more or edit any of these groups. Each group can have specific permissions. Check the menu Configuration->User Group.

16.1.4 Group for agent users

Select the group that the clients of this retailer used.

16.1.5 Plan

Plan that will be used to charge the clients.

16.1.6 Language

Language. This language is used for some system function, but not for the panel language.

16.1.7 Prefix rules

Prefix rules. You can see more details at the link https://www.magnusbilling.org/local_prefix.

16.1.8 Active

Only active users can login into the panel and make calls

16.1.9 Country

Used to CID Callback. The country prefix code will be added before the CID to convert the CID to E164

16.1.10 Activate offer

Used to give free minutes. It's necessary to inform the tariffs that will belongs to the free packages.

16.1.11 CPS Limit

CPS(calls per second) limit to this client. The calls that exceed this limit will be send CONGESTION.

16.1.12 Description

We did not write the description to this field.

16.1.13 Company website

Company website. Also used to agent panel customization. To agent, set the domain without http or www.

16.1.14 Company name

Company name. Also used to agent panel customization. Whether is a agent this name will be used on the login panel. Need set the company website and use the agent domain to working the customization

16.1.15 Commercial name

Brand name.

16.1.16 State number

State number.

16.1.17 Last name

Lastname.

16.1.18 First name

Firstname.

16.1.19 City

City.

16.1.20 State

State.

16.1.21 Address

Address.

16.1.22 Neighborhood

Neighborhood.

16.1.23 Zip code

Zipcode.

16.1.24 Phone

Landline phone.

16.1.25 Mobile

Mobile phone.

16.1.26 Email

Email, it's necessary to send system notifications.

16.1.27 DOC

Client document.

16.1.28 VAT

Used in some payment methods.

16.1.29 Contract value

We did not write the description to this field.

16.1.30 DIST

We did not write the description to this field.

16.1.31 Type paid

Pos-paid clients can stay with negative balance until the credit limit informed in the field below.

16.1.32 Credit notification daily

Enable this option to customer receive daily balance notification Email. You can customize the email on Configuration menu, submenu Email Templates

16.1.33 Credit limit

If the user is Post-paid, the user will be able to make calls until he reaches this limit.

16.1.34 Credit notification

If the client credit get lower than this field value, MagnusBilling will send an email to the client warning that he is with low credits. IT'S NECESSARY HAVE A REGISTERED SMTP SERVER IN THE SETTINGS MENU.

16.1.35 Enable expire

Activate expire. It's necessary to inform the expiry date in the "Expiry date" field.

16.1.36 Expiration date

The date that the user will not be able to make calls anymore.

16.1.37 Call limit

The amount of simultaneous calls allowed for this client.

16.1.38 Limit error

Warning to be send if the call limit is exceeded.

16.1.39 Record call format

Format used to record calls.

16.1.40 Callshop

Activate the CallShop module. Only active if you really are going to use it. It's necessary give permission to the selected group.

16.1.41 Disk space

Insert the amount disk space available to record, in GB. Use -1 to save it without limit. It's necessary to add in the cron the following php command `/var/www/html/mbilling/cron.php UserDiskSpace` .

16.1.42 SIP account limit

The amount of VoIP accounts allowed by this user. Will be necessary give permission to the group to create VoIP accounts.

16.1.43 CallingCard PIN

Used to authenticate the CallingCard.

16.1.44 Restriction

Used to restrict dialing. Add the numbers in the menu: Users->Restricted numbers.

16.1.45 Use

Which number will be used to check the restriction. This option is valid only to outbound calls.

16.1.46 Profit

This function is not available in Brazil. It's only used to mobile refills in some countries.

16.1.47 Profit

This function is not available in Brazil. It's only used to mobile refills in some countries.

16.1.48 Profit

This function is not available in Brazil. It's only used to mobile refills in some countries.

16.1.49 Enable DBBL/Rocket

This function is not available in Brazil. It's only used to mobile refills in some countries.

16.1.50 Profit

This function is not available in Brazil. It's only used to mobile refills in some countries.

16.1.51 Show selling price

This function is not available in Brazil. It's only used to mobile refills in some countries.

16.1.52 Email

We did not write the description to this field.

16.1.53 Services email notification

We did not write the description to this field.

16.1.54 DID email notification

We did not write the description to this field.

16.2 SIP Users

This menu has the following fields

16.2.1 Username

User that this SIP user is associated with.

16.2.2 SIP user

Username used to login in a Softphone or any SIP device.

16.2.3 SIP password

Password to login in a Softphone or any SIP device.

16.2.4 CallerID

The Caller ID number that will be shown in their destination. Your trunk needs to accept CLI.

16.2.5 Alias

Alias to dial between sip accounts from the same AccountCode (company).

16.2.6 Disallow

Disallow all codecs and then select the codecs available below to enable them to the user.

16.2.7 Codec

Select the codecs that the trunk will accept.

16.2.8 Host

Dynamic is an option that allows the user to register their account under any IP. If you want to authenticate the user via IP, put the client IP here, let the password field blank and set it to “insecure” to por/invite in the Additional Informations tab.

16.2.9 Group

When sending an call from DID, or campaign to a group, will be called all SIP users that are in the group. You can create the groups with any name.

Is used as well to capture calls with *8, need to configurate the option “pickupexten = *8” in the file “feature.comf”.

16.2.10 Videosupport

Activate video calls.

16.2.11 Block call regex

Block calls using REGEX. To block calls from cellphones, just put it ^55\d\d9. You can see more details at the link <https://regex101.com..>

16.2.12 Record call

Record calls of this SIP user.

16.2.13 Tech prefix

Useful option for when it’s necessary to authenticate more than one client via IP that uses the same IP. Common in BBX multi tenant.

16.2.14 Description

We did not write the description to this field.

16.2.15 NAT

Nat. You can see more details at the link <https://www.voip-info.org/asterisk-sip-nat/>.

16.2.16 Directmedia

If enabled, Asterisk tries to redirect the RTP media stream to go directly from the caller to the callee.

16.2.17 Qualify

Sent the “OPTION” package to verify if the user is online.
Syntax:

qualify = xxx | no | yes

Where the XXX is the number of milliseconds used. If “yes”, the time configurated in sip.conf is used, 2 seconds is the standard.

If you activate “qualify”, the Asterisk will sent the command “OPTION” to SIP peer regularly to verify if the device is still online.

If the device don’t answer the “OPTION” in the set period of time, Asterisk will consider the device offline for future calls.

This status can be verified with the funcion “sip show peer XXXX”, this funcion will only provide informations of status for the SIP peer that possess “qualify = yes.

16.2.18 Trunk groups

::::::WARNING::::::. By selecting a trunk group here, the trunk group will be ignored from tariffs and this trunk group will always be used. Only select a trunk group here if you really want all calls from this SIP user to be sent to this trunk group

16.2.19 Context

This is the context that the call will be processed, “billing” is the standard option. Only change configuration if you have knowledge of Asterisk.

16.2.20 Dtmfmode

DTMF type. You can see more details at the link [https://www.voip-info.org/asterisk-sip-dtmfmode/..](https://www.voip-info.org/asterisk-sip-dtmfmode/)

16.2.21 Insecure

This option need to be “NO” if the host is dynamic, so the IP authentication changes to port,invite.

16.2.22 Deny

You can limit SIP traffic of a determined IP or network.

16.2.23 Permit

You can allow SIP traffic of a determined IP or network.

16.2.24 Type

Standard type is “friend”, in other words, can make and receive calls. You can see more details at the link [https://www.voip-info.org/asterisk-sip-type/..](https://www.voip-info.org/asterisk-sip-type/)

16.2.25 Allowtransfer

Enable this VOIP account to do tranference. The code to transfer is *2 + ramal. It's necessary to activa the option atxfer => *2 in the file "features.conf" of Asterisk.

16.2.26 Ring false

Activate false ring. Add rR of the "Dial" command.

16.2.27 Call limit

Maximum simultaneous calls allowed for this SIP user.

16.2.28 MOH

Waiting music for this SIP user.

16.2.29 URL events notify

.

16.2.30 Addparameter

The parameters set in here will replace the system default parameters, as well of the trunks, if there's any.

16.2.31 AMD

.

16.2.32 Forward type

Resend destination type. This resend will not work in queues.

16.2.33 IVR

Select the IVR that you want to to send to calls if the SIP user don't answer.

16.2.34 Queue

Select the queue that you want to to send to calls if the SIP user don't answer.

16.2.35 Sip user

Select the SIP users that you want to send to calls if the SIP user don't answer.

16.2.36 Destination

Click for more details

We have 3 options, conform the selected type, group, number or custom.

* Group, the group name set here, needs to be exactly the same group of SIP users that wants to receive the calls, is going to call all SIP users in the group.

* Custom, it's possible to execute any valid option of the DIAL command of Asterisk, example: SIP/contaSIP,45,tTr

* Number, can be a landline number or mobile number, needs to be in the 55 DDD format

16.2.37 Dial timeout

Timeout in seconds to wait for the call to be picked-up. After the timeout will be execute the channeling if it's configurated.

16.2.38 Enable voicemail

Activate voicemail. It's necessary the configuration of SMTP in Linux to receive the email with the message. You can see more details at the link <https://www.magnusbilling.org/br/blog-br/9-novidades/25-configurar-ssmtp-para-enviar-voicemail-no-asterisk.html>.

16.2.39 Email

Email that will be send the email with the voicemail.

16.2.40 Password

Voicemail password. It's possible to enter in the Voicemail typing *111

16.2.41 Parameters

We did not write the description to this field.

16.2.42 Peer

sip show peer

16.2.43 CNL zone

We did not write the description to this field.

16.3 Calls Online

This menu has the following fields

16.4 CallerID

This menu has the following fields

16.4.1 Username

Select user.

16.4.2 CallerID

The number to CID authenticate with CallingCard. Use the exact format that you received the CallerID from your DID provider.

16.4.3 Name

Optional.

16.4.4 Description

CallerID description.

16.4.5 Status

Status of the CallerID.

16.5 ATA Linksys

This menu has the following fields

16.5.1 Serial

LinkSys serial number

16.5.2 MAC

LinkSys MAC address

16.5.3 User password

Username to login in LinkSys settings

16.5.4 Admin password

Password to login in LinkSys settings

16.5.5 Antireset

Be cautious.*73738# command prevents resetting LinkSys.

16.5.6 Enable_Web_Server

Beware! If deactivated, will not be able to login in the Linksys settings.

16.5.7 Dial Tone

We did not write the description to this field.

16.5.8 Proxy

Proxy 1.

16.5.9 Username

SIP user username that will be used in ATA line 1.

16.5.10 Password

SIP user password

16.5.11 Use_Pref_Codec

Only use the preferred CODEC

16.5.12 Codec

Set the preferred CODEC

16.5.13 Register expires

Interval in seconds that LinkSys will send a REGISTER to your server. Useful to avoid a loss of connection when you receive a call.

16.5.14 Dial plan

Read LinkSys documentation

16.5.15 NAT Mapping

It's recommended to activate this option if ATA is behind NAT.

16.5.16 NAT keep alive

It's recommended to activate this option if ATA is behind NAT.

16.5.17 Proxy

Proxy 2.

16.5.18 Username

SIP user username that will be used in ATA line 1.

16.5.19 Password

VOIP account password.

16.5.20 Use_Pref_Codec

Only use preferencial codec.

16.5.21 Codec

Settings of preferencial codec.

16.5.22 Register expires

Time in seconds that Linksys sends “REGISTER” to the server. If it’s going to get calls in this line, it’s better set it up between 120 and 480 seconds.

16.5.23 Dial plan

Read linksys documentation

16.5.24 NAT Mapping

It’s recommended to activate this option if ATA is behind NAT.

16.5.25 NAT keep alive

It’s recommended to activate this option if ATA is behind NAT.

16.5.26 Enable STUN

Activate STUN server.

16.5.27 STUN Test

Validate STUN server periodically..

16.5.28 Substitute VIA Addr

Replace public IP in the VIA.

16.5.29 STUN Server

STUN server domain.

16.6 Restricted Number

This menu has the following fields

16.7 Callback

This menu has the following fields

16.8 Buy Credit

This menu has the following fields

16.9 Iax

This menu has the following fields

16.9.1 Username

The user whose IAX account will belong

16.9.2 IAX user

The user that will be used to authenticate in the softphone

16.9.3 IAX password

The Password that will be used to authenticate in the softphone

16.9.4 CallerID

This is the CallerID that will be shown in their destination, in external calls the provider will need to permit CLI to be correctly identified in their destination.

16.9.5 Disallow

In this option will be possible to deactivate codecs. To deactivate all the codecs and letting available to the user only what you select below, use “Use all”

16.9.6 Codec

Codecs that will be accepted.

16.9.7 Host

“Dynamic” is an option that will let the user register his account in any IP. If you want to to authenticate the user by their IP, fill here the IP of the client, let the password field blank and put “insecure” for the port/invite in the tab “Additional Information”

16.9.8 NAT

The client is behind NAT. You can see more details at the link <https://www.voip-info.org/asterisk-sip-nat/>..

16.9.9 Context

This is the context that the call will be processed, by default is set to “billing”. Only alter if you have knowledge of Asterisk.

16.9.10 Qualify

Sent the “OPTION” package to verify if the user is online.

Syntax:

qualify = xxx | no | yes

Where the XXX is the number of milliseconds used. If “yes”, the time configured in sip.conf is used, 2 seconds is the standard.

If you activate “qualify”, the Asterisk will sent the command “OPTION” to SIP peer regularly to verify if the device is still online.

If the device don’t answer the “OPTION” in the set period of time, Asterisk will consider the device offline for future calls.

This status can be verified with the function “sip show peer XXXX”, this function will only provide status informations to the SIP peer that have “qualify = yes”.

16.9.11 Dtmfmode

Type of DTMF. You can see more details at the link <https://www.voip-info.org/asterisk-sip-dtmfmode/>..

16.9.12 Insecure

If the host is set to “dynamic”, this option will need to be set to “no”. To authenticate via IP and alter to port. You can see more details at the link <https://www.voip-info.org/asterisk-sip-insecure/>..

16.9.13 Type

Default type is “friend”, in other words they can make and receive calls. You can see more details at the link <https://www.voip-info.org/asterisk-sip-type/>..

16.9.14 Call limit

Total of simultaneous calls allowed for this IAX account.

16.10 Send Credit

This menu has the following fields

16.11 User History

This menu has the following fields

CHAPTER 17

Menu Billing

This is the list of all fields with their description of the menu Billing

17.1 Refills

This menu has the following fields

17.1.1 Username

User that will be realized the refill.

17.1.2 Credit

Refill amount. Can be a positive or negative value, if the value is negative will remove from the total amount of credit of the client.

17.1.3 Description

Description to the calendar, only for self control.

17.1.4 Add payment

This setting is only to your control, the credit will be released to the user anyway if set to Payment NO

17.1.5 Date

We did not write the description to this field.

17.1.6 Invoice number

Invoice number.

17.1.7 Payment receipt

We did not write the description to this field.

17.2 Payment Methods

This menu has the following fields

17.3 Voucher

This menu has the following fields

17.3.1 Credit

Voucher price. You can see more details at the link
https://wiki.magnusbilling.org/en/source/how_to_use_voucher.html..

17.3.2 Plan

Plan that will be linked to the client that will use this VOUCHER.

17.3.3 Language

Language that will be used.

17.3.4 Prefix rules

Rule that will be used in the “prefix rule” field

17.3.5 Quantity

The amount of VOUCHERS to be generated.

17.3.6 Description

Description to the calendar, only for self control.

17.3.7 Voucher

VOUCHER number.

17.4 Refill Providers

This menu has the following fields

17.4.1 Provider

Providers name.

17.4.2 Credit

Refill value.

17.4.3 Description

Used for internal control.

17.4.4 Add payment

This option is to your control only. The credit authorized to the client even if it's set to "NO".

CHAPTER 18

Menu DIDs

This is the list of all fields with their description of the menu DIDs

18.1 DIDs

This menu has the following fields

18.1.1 DID

The exact number coming from the context in Asterisk. We recommend you to always use the E164 format. Also you can create intervals. Example 13605040001-13605040009. In this example will be created the DIDs 13605040001 until 13605040009.

18.1.2 Record call

Record calls for this DID. Recorded regardless of destination.

18.1.3 Status

Only active numbers can receive calls.

18.1.4 Callerid name

Use this field to set a CallerID name or leave it blank to use the received CallerID from the DID provider.

18.1.5 Setup price

Activation cost. This value will be deducted from the client the moment that the DID is associated with the user.

18.1.6 Monthly price

Monthly price. This value will be deducted automatically every month from the user's balance. If the client doesn't have enough credit the DID will be cancelled automatically.

18.1.7 Connection charge

This is the value that will be charged for each call. Simply by picking up the call, this value will be deducted.

18.1.8 Minimum time to charge

We did not write the description to this field.

18.1.9 Buy price initblock

We did not write the description to this field.

18.1.10 Buy price increment

We did not write the description to this field.

18.1.11 Minimum time to charge

Minimum time to tariff the DID. If you set it to 3 any call that with lower duration will not be charged for.

18.1.12 Initial block

Minimum time in seconds to buy. If you set it to 30 and the call duration is 10, the call will be billed as 30.

18.1.13 Billing block

This defines the block in which the call billing time will be incremented, in seconds. If set to 6 and call duration is 32, the call will be billed as 36.

18.1.14 Charge who

The user that will be charged for the DID cost.

18.1.15 Channel limit

Maximum simultaneous calls for this DID.

18.1.16 Description

Set here the destination!

18.1.17 Regular expression

This is a regular expression to tariff the DID depending on who is calling it.

Lets analyze a real example:

Suppose we want to charge 0.10 when we receive a call from a landline and 0.20 if its a mobile phone and block any other format.

In this example we will create rules to identify the CallerID in the format 0 + area code + number, area code + number, or 55 + area code + number.

Take a look at the following image on what the result would look like:

.. image:: ../img/did_regex.png :scale: 100%

Regular expression for mobile

```
^[1-9][0-9]9\d{8}$|^0[1-9][0-9]9\d{8}$|^55[1-9][0-9]9\d{8}$
```

Regular expression for landline

```
^[1-9][0-9]\d{8}$|^0[1-9][0-9]\d{8}$|^55[1-9][0-9]\d{8}$
```

18.1.18 Buy price per min

We did not write the description to this field.

18.1.19 Sell price per min

Price per minute if the number matches the above regular expression.

18.1.20 Block calls from this expression

Set to yes to block calls that matches with the above regular expression.

18.1.21 Send the call to callback

Send this call to CallBack if it matches with the above regular expression.

18.1.22 Regular expression

Same as the first expression. You can see more details at the link
<https://wiki.magnusbilling.org/en/source/modules/did/did.html#did-expression-1..>

18.1.23 Buy price per min

We did not write the description to this field.

18.1.24 Sell price per min

Price per minute if the number matches the above regular expression.

18.1.25 Block calls from this expression

Set to yes to block calls that matches with the above regular expression.

18.1.26 Send the call to callback

Send this call to CallBack if it matches with the above regular expression.

18.1.27 Regular expression

Same as the first expression. You can see more details at the link
<https://wiki.magnusbilling.org/en/source/modules/did/did.html#did-expression-1..>

18.1.28 Buy price per min

We did not write the description to this field.

18.1.29 Sell price per min

Price per minute if the number matches the above regular expression.

18.1.30 Block calls from this expression

Set to yes to block calls that matches with the above regular expression.

18.1.31 Send the call to callback

Send this call to Callback if it matches with the above regular expression.

18.1.32 Callback pro

Enables Callback Pro.

18.1.33 Use audio

Execute an audio.

18.1.34 Maximum trying

How many times will the system try to return the call?

18.1.35 Interval between trying

Time interval between each try, in minutes.

18.1.36 Early media

Execute an audio before the call is answered. Your DID provider needs to allow early media.

18.1.37 Mon-Fri

Example: if your company only callbacks to the callee if the call was placed in between 09:00-12:00 and 14:00-18:00 MON-FRY, between this time interval the workaudio is going to be played and then callback to the callee. You can use multiple time intervals separated by |.

18.1.38 Sat

The same but for Saturday.

18.1.39 Sun

The same but for Sunday.

18.1.40 Work audio

Audio that will be executed when a call is received at the time interval.

18.1.41 Out work audio

Audio that will be executed when a call is received out of the time interval.

18.1.42 Country

We did not write the description to this field.

18.1.43 Server

We did not write the description to this field.

18.2 DID Destination

This menu has the following fields

18.2.1 DID

Select the DID that you want create new destination for.

18.2.2 Username

User that will be the owner of this DID.

18.2.3 Status

Only active destinations will be used.

18.2.4 Priority

You can create up to 5 destinations for your DID. If a try is made and a error is received, MagnusBilling will try to send the call to the next destination priority available. Only works with the “SIP call” type.

18.2.5 Type

Type of destination.

18.2.6 Destination

Set here the destination!

18.2.7 IVR

Select a IVR to send the call to. The IVR needs to belong to the owner of the DID aswell.

18.2.8 Queue

Select a Queue to send the call to. The Queue needs to belong to the owner of the DID aswell.

18.2.9 Sip user

Select a SIP user to send the call to. The SIP user needs to belong to the owner of the DID aswell.

18.2.10 Context

In this field you may use a context in the format supported by Asterisk
Example:

```
_X. => 1,Dial(SIP/sipaccount,45)
same => n,Goto(s-${DIALSTATUS},1)
```

```
exten => s-NOANSWER,1,Hangup
exten => s-CONGESTION,1,Congestion
exten => s-CANCEL,1,Hangup
exten => s-BUSY,1,Busy
exten => s-CHANUNAVAIL,1,SetCallerId(4545454545)
exten => s-CHANUNAVAIL,2,Dial(SIP/sipaccount2,,T)
```

You should NOT set a name for the context because the name will be set automatically as [did-number-of-the-did]

You may take a look at the context at /etc/asterisk/extensions_magnus_did.conf

18.3 DIDs Use

This menu has the following fields

18.4 IVRs

This menu has the following fields

18.4.1 Name

Name of the IVR

18.4.2 Username

User who owns the IVR

18.4.3 MonFri intervals

Weekly interval of attendance, can be configurated with multiples shifts.

Example:

Supposing that the attendance hours are 08h to 12h and 14h to 19h. In this case the rule would be

08:00-12:00|14:00-19:00

18.4.4 Saturday intervals

Interval of attendance in saturdays, can be configurated with multiple shifts

Example:

Supposing that the attendance hours in the saturdays are 08h to 13h. In this case the rule would be

08:00-13:00

18.4.5 Sunday intervals

Interval of attendance in sundays, can be configurated with multiple shifts

Example:

Supposing that theres no attendance hours in the sundays. In this case the rule would be

00:00-00:00

18.4.6 Use holidays

If this option is activated then the system will check if there is a holiday registered for the day, if so, then the audio, not working, will be played.

18.4.7 Work audio

Audio to play in the attendance hours.

18.4.8 Out work audio

Audio to play when it's not attendance hours

18.4.9 Option 0

Select the destination if the option 0 is pressed. Let it in blank if don't want any action

18.4.10 Option 1

Select the destination if the option 1 is pressed. Let it in blank if don't want any action

18.4.11 Option 2

Select the destination if the option 2 is pressed. Let it in blank if don't want any action

18.4.12 Option 3

Select the destination if the option 3 is pressed. Let it in blank if don't want any action

18.4.13 Option 4

Select the destination if the option 4 is pressed. Let it in blank if don't want any action

18.4.14 Option 5

Select the destination if the option 5 is pressed. Let it in blank if don't want any action

18.4.15 Option 6

Select the destination if the option 6 is pressed. Let it in blank if don't want any action

18.4.16 Option 7

Select the destination if the option 7 is pressed. Let it in blank if don't want any action

18.4.17 Option 8

Select the destination if the option 8 is pressed. Let it in blank if don't want any action

18.4.18 Option 9

Select the destination if the option is pressed. Let it in blank if don't want any action

18.4.19 Default option

Select the destination if none of the options was selected.

18.4.20 Enable known SIP user

Activating this option will be able to type an SIP user to call it directly.

18.4.21 Option 0

Select the destination if the option 0 is pressed. Let it in blank if don't want any action

18.4.22 Option 1

Select the destination if the option 1 is pressed. Let it in blank if don't want any action

18.4.23 Option 2

Select the destination if the option 2 is pressed. Let it in blank if don't want any action

18.4.24 Option 3

Select the destination if the option 3 is pressed. Let it in blank if don't want any action

18.4.25 Option 4

Select the destination if the option 4 is pressed. Let it in blank if don't want any action

18.4.26 Option 5

Select the destination if the option 5 is pressed. Let it in blank if don't want any action

18.4.27 Option 6

Select the destination if the option 6 is pressed. Let it in blank if don't want any action

18.4.28 Option 7

Select the destination if the option 7 is pressed. Let it in blank if don't want any action

18.4.29 Option 8

Select the destination if the option 8 is pressed. Let it in blank if don't want any action

18.4.30 Option 9

Select the destination if the option 9 is pressed. Let it in blank if don't want any action

18.4.31 Default option

Select the destination if none of the options was selected.

18.5 Queues

This menu has the following fields

18.5.1 Username

User that owns the queue.

18.5.2 Name

Queue name.

18.5.3 Language

Queue language.

18.5.4 Strategy

Queue strategy.

18.5.5 Ringinuse

Call or not the agents of the queue that are in call.

18.5.6 Ring for

How long the phone will ring until timeout

18.5.7 Time for another agent

The amount of time in seconds that will retry the call.

18.5.8 Time for another call

Time in seconds until the agent can receive another call.

18.5.9 Weight

Queue Priority.

18.5.10 Periodic announce

A set of periodic announcements can be created by separating each announcements to reproduce whit commas. E.g.: queue-periodic-announce,your-call-is-important,please-wait. This data need to be in /var/lib/asterisk/sounds/ directory.

18.5.11 Frequency

How often to make a periodic announcement.

18.5.12 Announce position

Informs the postition in the queue.

18.5.13 Announce holdtime

Should we include an estimated hold time in the position announcements?

18.5.14 Announce frequency

How often to announce queue position and/or estimated holdtime to caller 0=off

18.5.15 Join empty

Allow calls when there's no one to answer the call.

18.5.16 Leave when empty

Hang the calls in queue when there's no one to answer.

18.5.17 Max wait time

Maximum wait time on the queue

18.5.18 Max wait time action

SipAccount, IVR, QUEUE or LOCAL channel to send the caller if the maximum wait time is reached. Use: SIP/sip_account, QUEUE/queue_name, IVR/ivr_name OR [LOCAL/extension@context](#).

18.5.19 Ring or playing MOH

Play waiting music or tone when the client is in the queue.

18.5.20 Audio musiconhold

Import one waiting music to this queue.

18.6 Queues Members

This menu has the following fields

18.7 Buy DID

This menu has the following fields

18.8 Queue DashBoard

This menu has the following fields

18.9 DIDww

This menu has the following fields

18.10 Holidays

This menu has the following fields

18.10.1 Name

Holiday name

18.10.2 Date

Day of holiday

18.11 DID History

This menu has the following fields

CHAPTER 19

Menu Rates

This is the list of all fields with their description of the menu Rates

19.1 Plans

This menu has the following fields

19.2 Tariffs

This menu has the following fields

19.2.1 Plan

The plan that you want to create a tariff for.

19.2.2 Destination

The prefix that you want create a tariff for.

19.2.3 Trunk groups

The group of trunks that will be used to send this call.

19.2.4 Sell price

The amount that you want to charge per minute.

19.2.5 Initial block

Minimum time in seconds to buy. E.g., if set to 30s and the call duration is 21s, will be charged for 30s.

19.2.6 Billing block

This defines how the time is incremented after the minimum. E.g, if set to 6s and call duration is 32s, will be charged for 36.

19.2.7 Minimum time to charge

Minimum time to tariff. If it's set to 3, will only tariff calls when the time is equal or more than 3 seconds.

19.2.8 Additional time

Additional time to add to all call duration. If it's set to 10, will be added 10 seconds to all call time duration, this affects tariffs.

19.2.9 Connection charge

We did not write the description to this field.

19.2.10 Include in offer

Set to yes if you want to include this tariff to a package offer.

19.2.11 Status

Deactivating Tariffs, MagnusBilling will completely desconsider this tariff. Therefore, deleting or deactivating will have the same effect.

19.3 Prefixes

This menu has the following fields

19.3.1 Prefix

Prefix code. Prefix will be used to tariff and bill the calls.

19.3.2 Destination

Destination name.

19.4 User Custom Rates

This menu has the following fields

19.5 Offers

This menu has the following fields

19.5.1 Name

Free package name

19.5.2 Package type

Type of package, there's 3 types. Unlimited calls, free calls or free seconds.

19.5.3 Free time to call

In this field is where the package available quantity configuration will occur.

Example:

- * Unlimited calls: In this option the field is blank, because will be allowed to call without any control.
- * Free calls: Configure the amount of free calls that you want to give.
- * Free seconds: Configure the amount of seconds that you want to allow the client to call.

19.5.4 Billing type

This is the period that the package will be calculated.

Look the description:

- * Monthly: The system will verify the day of the plan activation + 30 days that the client reached the package limit.
- * Weekly: The system will verify the day of the plan activation + 7 days that the client reached the package limit.

19.5.5 Price

Price that will be charged monthly to the client.

If on the expiry day the client don't have the sufficient funds to pay the package MagnusBilling will automatically cancel the package.

In the settings menu, adjusts, exist one option named Package Offer Notification, this value means how many days are left until the expiration of the package, the system will try to charge the subscription, if the client don't have the balance, MagnusBilling will send an email to the client informing the lack of funds.

The email can be edited in the menu, Email models , type, plan_unpaid, Expiry of Monthly Plan Notice subject.

To send emails it's necessary the configuration of SMTP in the SMTP menu.

To learn how free packages works: <https://wiki.magnusbilling.org/en/source/offer.html>.

19.5.6 Initial block

Minimum time in seconds to sell. This value will subscribe the tariffs of the client's plan.

19.5.7 Billing block

This defines how the time is incremented after the minimum. This value will subscribe the tariffs of the client's plan.

19.5.8 Minimum time to charge

Minimum time to tariff. If it's set to 3, will only tariff calls when the time is equal or more than 3 seconds.

19.6 Offer CDR

This menu has the following fields

19.7 Offer Use

This menu has the following fields

This is the list of all fields with their description of the menu Reports

20.1 CDR

This menu has the following fields

20.1.1 Date

Start time of the call

20.1.2 Sip user

SIP user that made the call

20.1.3 CallerID

Number sent to the trunk as the identifier of the call.

If the trunk accepts the sent CallerID, then this number will be used as the identifier.

In order to this work its going to be necessary to have the Fromuser field in the trunk empty.

20.1.4 Number

Number dialed by the client.

20.1.5 Destination

Name of the destination, this name is a relation to the prefix menu.

20.1.6 Username

User that made the call, the one who the call cost was taken from.

20.1.7 Trunk

Trunk that was used to complete the call.

20.1.8 Duration

Duration of the call in seconds.

20.1.9 Buy price

Buy cost. You can see more details at the link https://wiki.magnusbilling.org/en/source/price_calculation.html..

20.1.10 Sell price

Sell price, the value that was taken from the client. You can see more details at the link https://wiki.magnusbilling.org/en/source/price_calculation.html..

20.1.11 Sell price

Sell price, the value that was taken from the client. You can see more details at the link https://wiki.magnusbilling.org/en/source/price_calculation.html..

20.1.12 Uniqueid

Unique ID generated by Asterisk, this field is also the start time of the call in Epoch Unix.

20.2 CDR Failed

This menu has the following fields

20.3 Summary per Day

This menu has the following fields

20.4 Summary Day User

This menu has the following fields

20.5 Summary Day Trunk

This menu has the following fields

20.6 Summary Day Agent

This menu has the following fields

20.7 Summary per Month

This menu has the following fields

20.8 Summary Month User

This menu has the following fields

20.9 Summary Month Trunk

This menu has the following fields

20.10 Summary per User

This menu has the following fields

20.11 Summary per Trunk

This menu has the following fields

20.12 Call Archive

This menu has the following fields

20.13 Send Credit Summary

This menu has the following fields

20.14 Summary Month DID

This menu has the following fields

This is the list of all fields with their description of the menu Routes

21.1 Providers

This menu has the following fields

21.1.1 Name

Provider name

21.1.2 Credit

The amount of credit you have in your provider's account. This field is optional.

21.1.3 Credit control

If you set to YES and your provider credit is < 0 , all trunks from this provider will be deactivated.

21.1.4 Description

Description to the calendar, only for self control.

21.2 Trunks

This menu has the following fields

21.2.1 Provider

Provider which the trunk belongs.

21.2.2 Name

Trunk name, must be unique.

21.2.3 Username

Only used if the authentication is via username and password.

21.2.4 Password

Only used if the authentication is via username and password.

21.2.5 Host

IP or Trunk domain.

21.2.6 Add prefix

Add a prefix to send to your trunk.

21.2.7 Remove prefix

Remove a prefix to send to your trunk.

21.2.8 Codec

Select the codecs that are allowed in this trunk.

21.2.9 Provider tech

You need install appropriate drive to use card like dgv extra Dongle.

21.2.10 Status

If the trunk is inactive, Magnusbilling will sent the call to the backup trunk.

21.2.11 Go to backup if 404

Send call to the next trunk if receive error 404.

21.2.12 Register trunk

Only active this if the trunk is authenticated via username and password.

21.2.13 Register string

<user>:<password>@<host>/contact.

“user” is the user ID for this SIP server (ex 2345).

“password” is the user password

“host” is the SIP server domain or host name.

“port” send an solicitation of the register to this host port. Standard for 5060

“contact” is the extension of Asterisk contact. Example 1234 is set in the contact header of the SIP register message. The contact ramal is used by the SIP server remotely when it’s needed to send one call to Asterisk.

21.2.14 Fromuser

Several providers demand this option to authenticate, primarily when it’s authenticated via user and password. Let it blank to send the CallerID of the SIP user of From.

21.2.15 Fromdomain

Defines the FROM domain: in the SIP messages when act like a UAC SIP (client).

21.2.16 Language

Default language used in any Playback()/Background().

21.2.17 Context

Only change this if you know what you are doing.

21.2.18 Dtmfmode

DMTF type. You can see more details at the link <https://www.voip-info.org/asterisk-dtmf/>.

21.2.19 Insecure

Insecure. You can see more details at the link <https://www.voip-info.org/asterisk-sip-insecure/>..

21.2.20 Max use

Maximum simultaneous calls for this trunk.

21.2.21 NAT

Is the trunk behind NAT. You can see more details at the link <https://www.voip-info.org/asterisk-sip-nat/>..

21.2.22 Directmedia

If activated, Asterisk will try to send the RTP media directly between your client and provider. It's necessary to active on the trunk as well. You can see more details at the link <https://www.voip-info.org/asterisk-sip-canreinvite/>..

21.2.23 Qualify

Sent the "OPTION" package to verify if the user is online.

Syntax:

qualify = xxx | no | yes

Where the XXX is the number of milliseconds used. If "yes", the time configurated in sip.conf is used, 2 seconds is the standard.

If you activate "qualify", the Asterisk will sent the command "OPTION" to SIP peer regularly to verify if the device is still online.

If the device don't answer the "OPTION" in the set period of time, Asterisk will consider the device offline for future calls.

This status can be verified with the funcion "sip show peer XXXX", this funcion will only provide informations of status for the SIP peer that possess "qualify = yes.

21.2.24 Type

Default type is "friend", in other words they can make and receive calls. You can see more details at the link <https://www.voip-info.org/asterisk-sip-type/>..

21.2.25 Disallow

In this option is possible to deactivate codecs. Use "Use all" to deactive all codectcs and make it avaiable to the user only what you selected below.

21.2.26 Sendrpid

Defines if one Remote-Party-ID SIP header task to be send.
The default is “no”.

This field is frequently used by VoIP wholesalers providers to supply the callers identity, independently of the privacy settings (From SIP header).

21.2.27 Addparameter

These parameters will be added in the final AGI command - Dial command, where is in the ajust settings menu.

By default the DIAL command is:

```
,60,L(%timeout%:61000:30000)
```

Let's say that you wanted to add an MACRO in the trunk, therefore in this field you will add the parameter, set it up M(macro_name) and create your MACRO in the Asterisk extensions.

21.2.28 Port

If you want to use a different port than 5060, you will need open the IPTABLES port.

21.2.29 Link SMS

URL to send SMS. Replace the number variable to %number% and text per %text%. EXAMPLE. Your SMS URL is http://trunkWebSite.com/sendsms.php?user=magnus&pass=billing&number=XXXXXX&sms_text=SSSSSSSSSS. replace XXXXXX per %number and SSSSSSSSSS per %text%

21.2.30 SMS match result

Leave it blank to not wait the provider answer. Or write the text that needs to consist in the providers answer to be considered SENT.

21.2.31 Parameters

Valid format of Asterisk sip.conf, one option per line.

Example, let's say that you need to put the useragent parameter, so put it in this field:

```
useragent=my agent
```

```
.
```

21.2.32 Enable CNL

We did not write the description to this field.

21.3 Trunk Groups

This menu has the following fields

21.4 Provider Rates

This menu has the following fields

21.5 Servers

This menu has the following fields

21.5.1 Name

Server name.

21.5.2 Server IP

Server IP. You can see more details at the link
<https://magnussolution.com/br/servicos/auto-desempenho/servidor-slave.html..>

21.5.3 Public IP

Public IP.

21.5.4 Username

User to connect to the server.

21.5.5 Password

Password to connect to the server.

21.5.6 Port

Port to connect to the server.

21.5.7 SIPport

SIP port that the server will use.

21.5.8 Type

Server type.

21.5.9 Weight

This option is to balance the calls by weight.

Example.

Let's say there's 1 MagnusBilling server and 3 slave servers, and you want to send the double of calls to each slave, proportionally to the MagnusBilling server. Then just set the MagnusBilling server to weight 1, and for the slave servers weight 2.

21.5.10 Status

The proxy will only send calls to active servers and with weight higher than 0.

21.5.11 Description

Used for internal control.

21.6 Trunk Errors

This menu has the following fields

21.7 Provider CNL

This menu has the following fields

This is the list of all fields with their description of the menu Settings

22.1 Menus

This menu has the following fields

22.1.1 Text

Menu name

22.1.2 IconCls

Icon, default font “awesome V4”.

22.1.3 Main menu

Menu which this menu belongs. In case the menu is blank, it’s a main menu

22.1.4 Order

Order that the menu will be shown in the menu

22.2 Group Users

This menu has the following fields

22.3 Configuration

This menu has the following fields

22.3.1 Value

Value. You can see more details at the link <https://wiki.magnusbilling.org/en/source/config.html>..

22.3.2 Description

Description. You can see more details at the link <https://wiki.magnusbilling.org/en/source/config.html>.

22.4 Emails Templates

This menu has the following fields

22.5 Log Users

This menu has the following fields

22.6 SMTP

This menu has the following fields

22.6.1 Host

SMST domain

You need to verify if the datacenter where the server will be hosted don't block the ports used by SMTP.

22.6.2 Username

Username used to authenticate the SMTP server.

22.6.3 Password

Password used to authenticate the SMTP server.

22.6.4 Port

Port used by the SMTP server.

22.6.5 Encryption

Encryption type.

22.7 Fail2ban

This menu has the following fields

22.7.1 IP

IP Address.

22.7.2 Perm ban

With this option marked on YES, the IP will be placed on the ip-blacklist list of fail2ban and will be blocked forever. The option will NOT block the IP momentarily according the parameters of the file `/etc/fail2ba/jail.local`.

By default the IP is going to stay blocked for 10 minutes

22.7.3 Description

These informations are captured from the log file `/var/log/fail2ban.log`

It's possible to track this LOG with the command

```
tail -f /var/log/fail2ban.log
```

22.8 API

This menu has the following fields

22.8.1 Username

You need use the MagnusBilling API from <https://github.com/magnussolution/magnusbilling-api-php>. The username owner this API

22.8.2 Api key

This apy key will be necessary to execute the API

22.8.3 Api secret

This apy secret will be necessary to execute the API

22.8.4 Status

You can activete or inactivate this API

22.8.5 Permissions

Which action the user will have execute

22.8.6 Restriction IPs

What IPS you want allow access this API. Leave blank to allow any IP. It is very recomendet set the IPS

22.9 Dashboard

This menu has the following fields

22.10 Call per Minutes

This menu has the following fields

22.11 Extra2

This menu has the following fields

22.12 Group to Admins

This menu has the following fields

22.13 Extra3

This menu has the following fields

22.14 Backup

This menu has the following fields

22.15 Alarms

This menu has the following fields

22.15.1 Type

We did not write the description to this field.

22.15.2 Period

We did not write the description to this field.

22.15.3 Condition

We did not write the description to this field.

22.15.4 Amount

We did not write the description to this field.

22.15.5 Email

We did not write the description to this field.

22.15.6 Status

We did not write the description to this field.

22.15.7 Subject

We did not write the description to this field.

22.15.8 Message

We did not write the description to this field.

22.16 Extra

This menu has the following fields

Menu Voice Broadcasting

This is the list of all fields with their description of the menu Voice Broadcasting

23.1 Campaigns

This menu has the following fields

23.1.1 Username

User that owns the campaign.

23.1.2 Plan

What plan do you want to use to bill this campaign?

23.1.3 Name

Name of the campaign.

23.1.4 CallerID

We did not write the description to this field.

23.1.5 Status

Status of the campaign.

23.1.6 Starting date

The campaign will start from this date.

23.1.7 Expiration date

The campaign will stop in this date.

23.1.8 Type

Choose Voice or SMS. If you choose Voice you will need to import audio. If you choose SMS you will need to set the text in the SMS tab.

23.1.9 Audio

Available to massive calling. The audio needs to be compatible with Asterisk. The recommended format is GSM or WAV(8k hz mono).

23.1.10 Audio 2

If you use TTS, the name will be executed between Audio and Audio2.

23.1.11 Restrict phone

Activating this option, MagnusBilling will check if the number that will be sent the call is registered in the Restrict Phone menu, if it has, the system will change the status of the number to blocked and will not send the call.

23.1.12 Auto reprocess

If there are no active numbers in this campaign phone book, reactivates all pending numbers.

Select one or more phonebooks to to be used.

23.1.13 Number to forward

Do you want to forward the call after the audio? E.g, if the callee presses 1, he gets sent to SIP user XXXX. Set Number to Forward = 1, Forward Type = SIP and select the SIP user to send the callee to. Set -1 to disable.

23.1.14 Forward type

Choose the type of redirect. This will send the call to the chosen destination.

23.1.15 IVR

Choose a IVR to send the call to. The IVR needs to belong to the owner of the campaign.

23.1.16 Queue

Choose a Queue to send the call to. The Queue needs to belong to the owner of the campaign.

23.1.17 Sip user

Choose a SIP user to send the call to. The SIP user needs to belong to the owner of the campaign.

23.1.18 Destination

Click for more details

There are two options available.

- *Group, the group name should be put here exactly as it is in the SIP users that should receive the calls.

- *Personalized, you may execute any valid option via Asterisk's DIAL command. Example:
SIP/sipaccount,45,tTr.

23.1.19 Record call

Record the calls of the campaign. They only will be recorded if the call is transferred.

23.1.20 Daily start time

Time that the campaign will start sending.

23.1.21 Daily stop time

Time that the campaign will stop sending.

23.1.22 Monday

Activating this option the system will send calls on Mondays.

23.1.23 Tuesday

Activating this option the system will send calls on Tuesdays.

23.1.24 Wednesday

Activating this option the system will send calls on Wednesdays.

23.1.25 Thursday

Activating this option the system will send calls on Thursdays.

23.1.26 Friday

Activating this option the system will send calls on Fridays.

23.1.27 Saturday

Activating this option the system will send calls on Saturdays.

23.1.28 Sunday

Activating this option the system will send calls on Sundays.

23.1.29 Call limit

How many numbers will be processed per minute?

This value will be divided by 60 seconds and the calls will be sent every minute at the same time.

23.1.30 Maximum call limit

This is the maximum value that the client will be able to set. If you set it to 50 the user will be able to change to any value that is 50 or less than 50.

23.1.31 Audio duration

Used to control the max completed calls.

23.1.32 Toggle max completed calls

If activated MagnusBilling will check how many calls were already made and have a duration total bigger than the audios. If the quantity is equal or bigger than the value set in the field, the campaign will be deactivated.

23.1.33 Max completed calls

Maximum amount of complete calls. You need to activate the field above to use this.

23.1.34 Description or SMS Text

This field has different uses if the campaign is sending Voice or SMS.

Uses:

* Voice: This field is simply a description of the campaign.

* SMS: The text in here is going to be sent to the numbers. You may use the var `%name%` where you want to use the name of the customer. Example:

Hello `%name%`

23.1.35 Audio 1 TTS

With this setting the system will generate the audio 1 for the campaign via TTS.

In order for this to work, you will need to set the TTS URL under Settings, Configuration, TTS URL.

23.1.36 Audio 2 TTS

Same setting as the previous field but for audio 2. Keep in mind that in between audio 1 and 2, the TTS executes the name imported with the number.

23.2 Phonebooks

This menu has the following fields

23.3 Phonenumbers

This menu has the following fields

23.4 Polls

This menu has the following fields

23.5 Polls Reports

This menu has the following fields

23.6 Restrict Phone

This menu has the following fields

23.7 SMS

This menu has the following fields

23.7.1 Username

User that sent/received the SMS.

23.7.2 Number

Number in the E164 format.

23.7.3 SMS

SMS text.

23.7.4 From

If your SMS provider accepts the submission of FROM, put it here. This value will be replaced for the variable %from% in the trunk URL.

23.7.5 Provider result

We did not write the description to this field.

23.8 Quick Campaign

This menu has the following fields

23.9 Campaigns DashBoard

This menu has the following fields

23.10 Campaign Report

This menu has the following fields

CHAPTER 24

Menu CallShop

This is the list of all fields with their description of the menu CallShop

24.1 Booths

This menu has the following fields

24.2 Booths Report

This menu has the following fields

24.3 Booths Tariffs

This menu has the following fields

24.4 Summary per Day

This menu has the following fields

CHAPTER 25

Menu Summary Month Trunk

This is the list of all fields with their description of the menu Summary Month Trunk

CHAPTER 26

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