

## QueueMetrics 20.11 WebRTC SoftPhone - FreePBX PJSIP setup



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In this tutorial we will go through the necessary steps to setup the latest version of the QueueMetrics Softphone. With the latest version we strongly recommend you switch to PJSIP extensions, following the recommendations of various PBX manufacturers.

#### QueueMetrics

QueueMetrics is a highly scalable monitoring software that lets you track agent productivity, payrolls, measure targets, conversion rates, ACD, IVR, Music on hold, generate outbound campaign statistics and monitor realtime processes with customizable wallboards.

You can measure all contact centre activities with more than 200 different metrics and manage realtime processes with extensions and calls control, live alarms, whisper mode, spy and barge mode.

#### Prerequisites

A QueueMetrics Live instance or a QueueMetrics on-premise installation (20.11.8 at the time of writing), with SSL configured.

If you are not sure how to setup SSL for your QueueMetrics on-premise, please follow [our manual](#).

A valid SSL certificate should also be added to FreePBX's Certificate Manager. [A Let's Encrypt certificate](#) is a valid solution if you don't know where to start.

To add a certificate to FreePBX, please follow the [relevant documentation](#).

#### FreePBX Setup

##### TLS/WSS Setup

First of all, we need to enable WSS transport for PJSIP extensions.

To do this, navigate to Settings => Asterisk SIP Settings => General SIP Settings

And make sure your NAT settings are correct (if any). You can use the Detect Network Settings button to automatically configure these settings.

Once this is done, navigate to the pjsip tab:

SIP Settings [chan\_pjsip]

Here we need to specify that we want our PJSIP extensions to use TLS and set the transport protocol to WSS. We also need to specify the certificate we want to use to guarantee the connection.

First, let's select the certificate mentioned in the prerequisites as the TLS transport certificate.

TLS/SSL/SRTP Settings => Certificate Manager

Then, down below, enable WSS transport.

WSS - 0.0.0.0 – All = Yes

Once you have done this, you can submit and save your changes.

Extension Setup

Let's create a new PJSIP extension by navigating to

Applications => Extensions => Add Extension => Add New SIP [chan\_pjsip] Extension

Extension: 100

General Voicemail Find Me/Follow Me Advanced Pin Sets Other

Edit Extension

This device uses PJSIP technology listening on Port 5060 (UDP)

Display Name 100

Outbound CID

Emergency CID

Secret

Language

Language Code Default

User Manager Settings

Linked to User 100

Select User Directory: PBX Internal Directory

Link to a Different Default User: 100 (Linked)

Username  ☐ Use Custom Username

Password For New User

Groups

We set the Display Name and Secret for our new extension, then navigate to the Advanced tab.

Here, we need to set the following options:

Option	Value
Enable AVPF	Yes
Enable ICE Support	Yes
Enable rtcp Mux	Yes
Media Encryption	DTLS-SRTP
Transport	0.0.0.0-wss
Enable DTLS	Yes
Enable ICE Support	Yes
Use Certificate	The same certificate mentioned in the <b>prerequisites</b>
DTLS Verify	Fingerprint
DTLS Setup	Act/Pass
DTLS Rekey Interval	0

Qualify Frequency 60

Transport 0.0.0.0-wss

Enable AVPF

Enable ICE Support

Enable rtcp Mux

Self Secure

Media Encryption DTLS-SRTP (not recommended)

Session Timers Yes

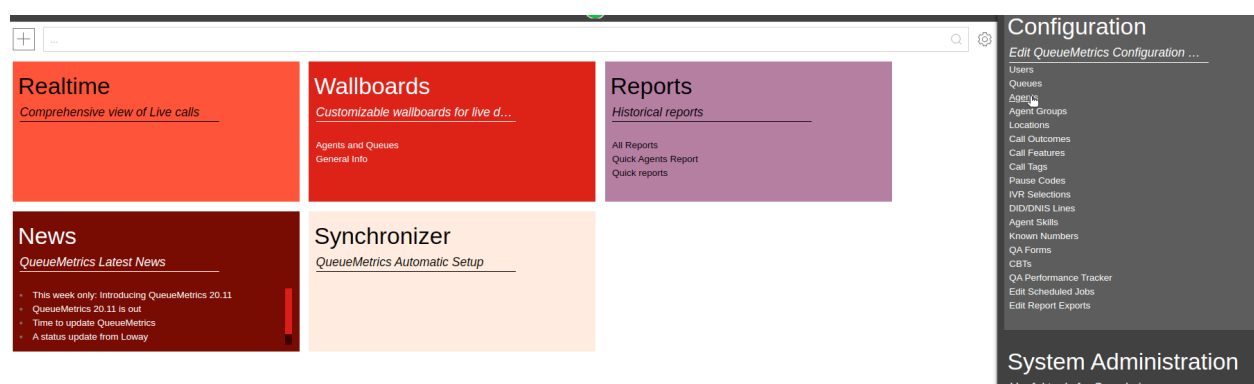
DTLS

Enable DTLS	No Yes
Auto Generate Certificate	No Yes
Use Certificate	default
DTLS Verify	Fingerprint
DTLS Setup	Act/Pass
DTLS Rekey Interval	0

Once we have done this, the extension is ready. We can then save and apply our changes.

## QueueMetrics Setup

To Create and agent, linked to the Asterisk Extension we created earlier (100 in our case), go to Agents → Create New.



We call the agent Agent/100 and fill in the Agent Description field as we please. We then fill the following fields:

WebPhone Username with the extension username we previously set (in this case 100).

WebPhone Password with the extension secret we previously set (in this case mysecretpassword).

WebPhone Realm with the address of our FreePBX machine. (in this case myfreepbx.com).

Current terminal:

Instant messenger address:  TEST IT

WebPhone Username:

WebPhone Password:

WebPhone Realm:

WebPhone SIP Uri:

Auto Answer:

We click on Save then go back to the Homepage.

Now we need to create a User for Agent/100, so we go to Users → Create New.

We set the Login field to 100, the Password to 100 and we set the Class field to AGENTS.

## User Detail

User Id:	<input type="text"/>
Login:	<input type="text" value="agent/100"/>
Password:	<input type="password" value="..."/>
Confirm Password:	<input type="password" value="..."/>
Real name:	<input type="text" value="WebRTC 100"/>
Enabled:	<input type="text" value="Yes"/>
E-mail:	<input type="text"/>
Masterkey:	<input type="text" value="No"/>
Class:	<input type="text" value="AGENTS"/>

We click on Save and we go back to the homepage.

The last thing we need to do is to set one parameter in the Explore System Parameters page. Scroll down to the end of the page until you see the following parameter: Web Socket URL for the connection.

We need to set this, using the correct IP address for our FreePBX machine. Then we Save our file.

## Agent Page Softphone Settings

Maximum interval in seconds between WebSocket reconnection attempts. Default value is 30.

Minimum interval in seconds between WebSocket reconnection attempts. Default value is 2.

Web Socket URL for the connection.

Now we're good to go! We log out of our QueueMetrics Administrator Account and we login as Agent/100.

From the Widget Menu we select Soft Phone and here is our WebRTC client.

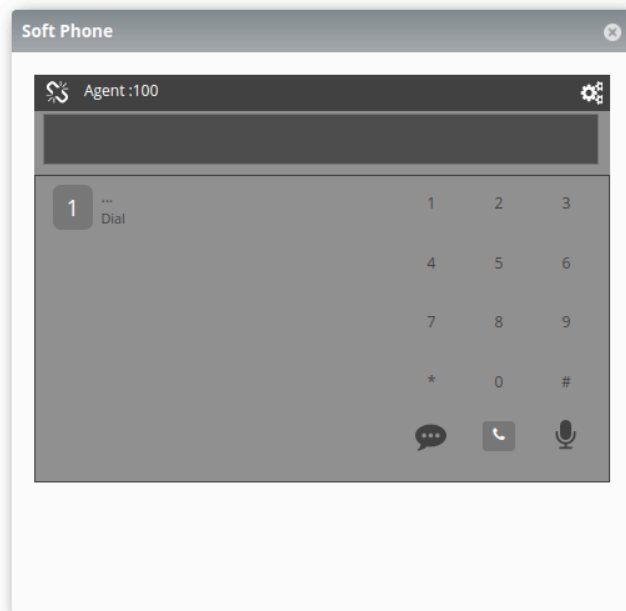


Now you can make and receive calls, as long as you give the webpage permission to use your microphone and headphones/speakers, when prompted.

#### Troubleshooting

The Softphone is not registering

If the softphone is unable to register with the PBX, it will look like this:



Usually this can happen due to various reasons:

- 1) The username or password for the extension are not correct

2) The PBX address is incorrect

3) The extension is a PJSIP extension, but the websocket is currently listening for SIP connections.

1 and 2 are easy to fix, as we just need to check all our settings again and make sure they are correct.

Number 3 is a fairly common occurrence, that can be fixed by following these steps.

NOTE: To fix this issue you will need to restart the PBX, so make sure you do this during an appropriate maintenance window.

Navigate to

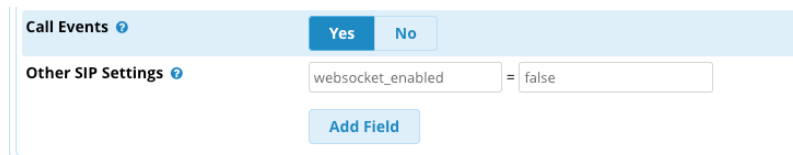
Settings => Asterisk Sip Settings => Chan SIP Settings => Advanced General Settings => Other SIP Settings.

and add the following value in the left field>

websocket\_enabled

and this other value in the right field

false



The screenshot shows the Asterisk SIP Settings interface. At the top, there are tabs for 'Call Events' (with a help icon) and 'Yes' (selected) and 'No'. Below this, the 'Other SIP Settings' section is visible, containing a table with one row: 'websocket\_enabled' in the left column, an equals sign in the middle, and 'false' in the right column. An 'Add Field' button is located below the table.

Save, apply and then we will need to restart the PBX.

NOTE: This will interrupt the PBX operations so make sure you are outside of production time

```
asterisk -rx 'core restart now'
```

Now the PBX should have PJSIP listening on the WSS WebSocket.

Refresh the agent page and check if the phone is now able to register correctly.

The Softphone is registering but calls hang up immediately

This can happen if the browser does not have access to the Microphone or the Speakers/Headphones. Make sure you have assigned the permissions correctly, and that you do not have another tab in the same browser that is currently using the Microphone.

Another reason for calls hanging up immediately, could be that the extension settings regarding AVPF, encryption and the like, specified in this tutorial, have not been set correctly.

The Softphone can make calls, but no audio is heard

This is usually due to NAT settings. If your PBX has NAT enabled, we need to set at least one ICE Server(STUN) to make sure that the audio streams are exchanged correctly. To do this we need to first choose a STUN server, for example:

```
stun.l.google.com:19302
```

and add it to both the Softphone configuration in QueueMetrics, and FreePBXs SIP configuration.

To set it in QueueMetrics, go to Explore System Parameters and set the List of ICE Servers to use. parameter under Agent Page Softphone Settings (if you are using the SoftPhone from the Wallboard page you will need to edit the same setting under Wallboard Softphone Settings.

once you find the correct option, set it to:

```
stun:stun.l.google.com:19302
```

Save and go to FreePBX's GUI. Here we need to navigate to Settings => Asterisk Sip Settings => General Sip Settings => Stun Server Address

and add the server like this:

```
stun.l.google.com:19302
```

NOTE: In QueueMetrics you need to prefix the server with "stun:".

Save and apply your changes in the PBX, and reload the agent page. You should now hear audio on your calls.

### QueueMetrics References

QueueMetrics software is available on premise or as a cloud hosted service for FreePBX, Yeastar S PBX, Grandstream, Issabel, FusionPBX and many other Asterisk distros.

For more technical information please refer to the [User Manual](#).

Visit [www.queuemetrics.com](http://www.queuemetrics.com) for a free 30 days full featured trial.

Attend our [Free Webinars](#) for a live demonstration of QueueMetrics.