

Enabling WebRTC

This page reorganizes the operational steps for **Enabling WebRTC**.

WebRTC is not enabled by default and you need to follow carefully the following steps to make it work.

- Install an SSL certificate on Asterisk. This needs to be a real SSL certificate. Check the Asterisk FAQ on how to install an SSL Certificate in Asterisk
- Install the Opus codec and add to the web interface. You can check the instructions in the Setup Guides section Installing OPUS codec
- Activate the http daemon for asterisk by editing the http.conf file as following:

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088
tlsenable=yes
tlsbindaddr=0.0.0.0:8089
tlscertfile=/etc/asterisk/certificates/demo.mirtapbx.com.pem
```

The tlscertfile needs to point to your certificate in pem format

You can provide separate cert and key:

```
tlscertfile=/etc/ssl/demo.mirtapbx.com.crt
tlsprivatekey=/etc/ssl/demo.mirtapbx.com.key
```

It may be needed to set the following parameters in sip.conf too:

```
dtlsenable=yes
dtlsverify=fingerprint
dtlscertfile=
dtlsprivatekey=
```

- Activate the ws and wss transport in sip.conf, set the realm

```
transport=udp,tcp,tls,ws,wss
realm=demo.mirtapbx.com
```

- Set the path for your certificate in the Web interface - Admin/Settings, Security section, WebRTC SSL path field
- In your extension enable Opus Codec, WebRTC support, RTP Encryption and Any Transport

Verifying WebRTC

To verify the WebRTC configuration, you can try to register and place calls using the SipML5 by visiting: <https://www.doubango.org/sipml5/>

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Current Verification

After applying the change, verify the related MiRTA PBX page, the Asterisk logs, and the relevant Status menu entry. Recheck tenant selection before testing tenant-specific behavior.

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