Enabling 16kHz audio for all FreeSWITCH services in sipXecs 4.4

To enable 16kHz audio for all FreeSWITCH services in sipXecs 4.4, edit the following files:

/etc/sipxpbx/freeswitch/local_stream.conf.xml.vm

Change all instances of:

```
<param name="rate" value="8000"/>
```

to:

```
<param name="rate" value="16000"/>
```

/etc/sipxpbx/freeswitch/sofia.conf.xml.vm

change:

```
<param name="inbound-codec-negotiation" value="scrooge"/>
```

to:

```
<param name="inbound-codec-negotiation" value="generous"/>
```

/etc/sipxpbx/freeswitch/conf/autoload_configs/conference.conf.xml

change all instances of:

```
<param name="rate" value="8000"/>
```

to:

```
<param name="rate" value="16000"/>
```

Applying settings

Log into sipXconfig then browse to **System >> Servers >> {server_name} >> Media Services** then click **OK**. sipXecs will ask you to restart media services. Perform the Media Services restart then try out a conference bridge and you should hear HD audio

Alleviating Polycom G.722 audio garble for voicemail

To alleviate audio garble caused by Polycom phones, set **SPEEX** as the top codec in **System >> Servers >> {server_name} >> Media Services** then restart Media Services. The audio quality won't be quite as good as with G.722 for continuous streams but will help alleviate the garbled audio issue reported here: http://track.sipfoundry.org/browse/XTRN-1064