

DTMF support

Overview

DTMF - Dual-Tone Multi-Frequency signaling

This signaling method is used for telecommunication over analogue telephone lines in the voice-frequency band between telephone handsets and other communication devices and the switching center.

Methods of sending DTMF

There are 3 methods of sending DTMF in the SIP environment:

1. `SIP INFO` packets.
2. As specially marked events in the RTP stream - see: [RFC 2833](#).
3. Inband as normal audio tones in the RTP stream with no special coding or markers.

WCS supports DTMF with 1 and 2.

DTMF settings

Using `SIP INFO` packets

To send DTMF using `SIP INFO` packets, the following parameter in [flashphoner.properties](#) file should be set

```
dtmf=INFO
```

PBX configuration

To receive DTMF using `SIP INFO` packets, PBX should be configured as follows (Asterisk for example)

```
[general]
bindport=5060
bindaddr=0.0.0.0
context=default
dtmfmode=info
allow=all
```

```
[2000]
type=friend
secret=2000
host=dynamic
canreinvite=no
dtmfmode=info
```

Note that Asterisk uses `inband` method by default which is not supported by WCS.

DTMF by RFC2833

To send DTMF as specially marked events according to RFC 2833, the following parameter in `flashphoner.properties` file should be set

```
dtmf=RFC2833
```

Also, `telephone-event` codec should be enabled, for example

```
codecs =opus, alaw, ulaw, g729, speex16, g722, mpeg4-
generic, telephone-event, h264, vp8, flv, mpv
codecs_exclude_sip =mpeg4-
generic, flv, mpv, alaw, ulaw, g729, speex16, g722, vp8
codecs_exclude_streaming =telephone-event
codecs_exclude_sip_rtmp =opus, g729, g722, mpeg4-generic, vp8, mpv
```

In the case above, `H264` for video, `opus` for audio and `telephone-event` for DTMF are enabled.

PBX configuration

To receive DTMF according to RFC 2833, PBX should be configured as follows (Asterisk for example)

```
[general]
bindport=5060
bindaddr=0.0.0.0
context=default
dtmfmode=rfc2833
allow=all

[2000]
type=friend
secret=2000
host=dynamic
canreinvite=no
dtmfmode=rfc2833
```

Note that Asterisk uses `inband` method by default which does not supported by WCS.