Audio Codecs

Transcodable Audio Codecs

The following codecs can be used when setting codec_string and absolute_codec_string.

- **OPUS**
  - opus@48000h@10i - Opus 48khz using 10 ms ptime (mono and stereo)
  - opus@48000h@20i - Opus 48khz using 20 ms ptime (mono and stereo)
  - opus@48000h@40i - Opus 48khz using 40 ms ptime
  - opus@16000h@10i - Opus 16khz using 10 ms ptime (mono and stereo)
  - opus@16000h@20i - Opus 16khz using 20 ms ptime (mono and stereo)
  - opus@8000h@10i - Opus 8khz using 10 ms ptime (mono and stereo)
  - opus@8000h@20i - Opus 8khz using 20 ms ptime (mono and stereo)
  - opus@8000h@40i - Opus 8khz using 40 ms ptime
  - opus@8000h@60i - Opus 8khz using 60 ms ptime
  - opus@8000h@80i - Opus 8khz using 80 ms ptime
  - opus@8000h@100i - Opus 8khz using 100 ms ptime
  - opus@8000h@120i - Opus 8khz using 120 ms ptime
  - provided by mod_opus

- **iSAC** provided by mod_isac

- **CODEC2 2550bps** 8000hz 20ms
  - provided by mod_codec2

- **SILK Skype Audio codec.**
  - provided by mod_silk.

- **iLBC@30i** - iLBC using mode=30 which will win in all cases.
  - Provided by mod_ilbc

- **Speex**
  - speex@8000h@20i - Speex 8kHz using 20ms ptime.
  - speex@16000h@20i - Speex 16kHz using 20ms ptime.
  - speex@32000h@20i - Speex 32kHz using 20ms ptime.
  - Provided by mod_speex

- **BroadVoice.**
  - BV32 - BroadVoice 16kHz, 32kb/s wideband
  - BV16 - BroadVoice 8kHz, 16kb/s narrowband
  - Provided by mod_bv.

- **Siren**
  - G7221@16000h - G722.1 16kHz (aka Siren 7)
  - G7221@32000h - G722.1C 32kHz (aka Siren 14)
  - Provided by mod_siren

- **CELT wideband.**
  - CELT@32000h - CELT 32kHz, only 10ms supported
  - CELT@48000h - CELT 48kHz, only 10ms supported
  - Provided by mod_celt

- **DVI**
  - DVI4@8000h@20i - IMA ADPCM 8kHz using 20ms ptime. (multiples of 10)
  - DVI4@16000h@40i - IMA ADPCM 16kHz using 40ms ptime. (multiples of 10)
  - GSM@80i - GSM 8kHz using 40ms ptime. (GSM is done in multiples of 20, Default is 20ms)
  - G722 - G722 16kHz using default 20ms ptime. (multiples of 10)
  - Provided by mod_spandsp

- **G.726**
  - G726-16 - G726 16bit adpcm using default 20ms ptime. (multiples of 10)
  - G726-24 - G726 24bit adpcm using default 20ms ptime. (multiples of 10)
  - G726-32 - G726 32bit adpcm using default 20ms ptime. (multiples of 10)
  - G726-40 - G726 40bit adpcm using default 20ms ptime. (multiples of 10)
  - AAL2-G726-16 - Same as G726-16 but using AAL2 packing. (multiples of 10)
  - AAL2-G726-24 - Same as G726-24 but using AAL2 packing. (multiples of 10)
  - AAL2-G726-32 - Same as G726-32 but using AAL2 packing. (multiples of 10)
  - AAL2-G726-40 - Same as G726-40 but using AAL2 packing. (multiples of 10)
  - LPC - LPC10 using 90ms ptime (only supports 90ms at this time in FreeSWITCH)
  - Provided by mod_spandsp.

- **G.729** - G729 in transcoding mode
  - provided by mod_g729

- **PCMUL** - G711 8kHz ulaw using default 20ms ptime. (multiples of 10)
- **PCMUL** - G711 8kHz alaw using default 20ms ptime. (multiples of 10)
- **L16** - L16 isn't recommended for VoIP but you can do it. L16 can exceed the MTU Rather quickly.
  - Provided in core PCM module.

Pass-through Audio Codecs

- **G.729** - G729 in passthrough mode. (mod_g729 / mod_com_g729)
  - Provided by mod_g729 for passthrough mode and mod_com_g729 for commercial license (10USD per channel)
- **G.723** - G723.1 in passthrough mode. (mod_g723_1)
- Provided in `mod_g723.1`
- AMR - AMR in passthru mode. (`mod_amr`)
  - Provided by `mod_amr`
- AMR-WB (G.722.2) - AMR-WB in passthru mode. (`mod_amr_wb`)
  - Provided by `mod_amr_wb`