Media Handling in FreeSWITCH

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Agenda

• Audio Codecs
• Transcoding
• Codec Negotiation
• Bypass Media
• Proxy Media
• Sangoma Transcoding
Audio Codecs
Audio Codecs

- Codecs encode and decode voice for network transmission
  - Algorithm (compression technology)
  - Bit rate
  - Sampling rate
  - Packetization

- Algorithm is the core of the codec
- Bit rate defines bandwidth required. (how many bits per second)
- Sampling rate defines the quality. (all other things being equal)
- Packetization affects latency and bandwidth overhead
Audio Codecs

- **G.711 (PCMU/PCMA, Ulaw/Alaw)** – Narrowband.
  - 64kbps per second (Bit rate)
  - 8kHz (Sampling rate)
  - 10ms, 20ms, 30ms, 40ms … + (Packetization)

- **G.722** – Wideband
  - 48kbps, 56kbps and 64kbps
  - 16kHz (IANA clocks it at 8kHz due to historical error in RFC1890)
  - 10ms, 20ms, 30ms, 40ms … +

- **G.722.1 Annex C - Ultra-wideband**
  - 48kbps
  - 32kHz
  - 20ms, 40ms, 60ms
FreeSWITCH supports a wide range of codecs

- Narrowband (G.711, G.726, G.723.1, G.729AB, Speex …)
- Wideband (G.722, G.722.1, G.722.2, Speex …)
- Ultra-wideband (G.722.1C, Speex)
- CD-quality (CELT)

FreeSWITCH core requires the media to be in L16 (signed linear, raw digital audio) format for manipulation (mixing, tone detection etc)

Codec modules encode and decode from/to L16 format

Pass-thru codec modules are dummies (mod_g729, mod_g723_1)
FreeSWITCH Audio Codecs

```
root@sigchld:~# show codecs

type, name, ikey
codec, ADPCM (IMA), mod_spandsp
codec, G.711 a-law, CORE_PCM_MODULE
codec, G.711 u-law, CORE_PCM_MODULE
codec, G.722, mod_spandsp
codec, G.723.1 6.3k, mod_g723_1
codec, G.726 16k, mod_spandsp
codec, G.726 16k (AAL2), mod_spandsp
codec, G.726 24k, mod_spandsp
codec, G.726 24k (AAL2), mod_spandsp
codec, G.726 32k, mod_spandsp
codec, G.726 32k (AAL2), mod_spandsp
codec, G.726 40k, mod_spandsp
codec, G.726 40k (AAL2), mod_spandsp
codec, G.729, mod_g729
codec, GSM, mod_spandsp
codec, LPC-10, mod_spandsp
codec, PROXY PASS-THROUGH, CORE_PCM_MODULE
codec, PROXY VIDEO PASS-THROUGH, CORE_PCM_MODULE
codec, Polycom(R) G722.1/G722.1c, mod_siren
codec, RAW Signed Linear (16 bit), CORE_PCM_MODULE
codec, Sangoma G729, mod_sangoma_codec
codec, Speex, mod_speex
codec, iLBC, mod_ilbc

23 total.
```

Click to add notes
Transcoding
Transcoding

- Required when endpoints have no codec in common
- FreeSWITCH must stay in the media path
- Increases CPU usage (particularly if done in software)
- Is a must if you need:
  - Call recording
  - Tone detection
  - Play announcements or tones
FreeSWITCH Transcoding

- Transcoding in one-legged call
FreeSWITCH Transcoding

- Transcoding 2 SIP legs

![Diagram](image.png)
FreeSWITCH codec pass-thru

- Pass-thru codecs do not do transcoding
Codec Negotiation
Codec Negotiation

• Decisions to be made to choose a codec for a call

• From a list of codecs, pick one!

• You can choose when this happens (early vs late)

• Early happens before call hits the dial plan

• Late will happen when the leg is answered (or in pre-answer)
Codec Negotiation

• 3 inbound negotiation algorithms
  – generous
  – greedy
  – Scrooge (Bah HUMBUG!)

• Use `inbound-codec-negotiation` in SIP profile

• Use `sip_codec_negotiation` variable in the dial plan
Codec Negotiation

INVITE SDP codec list:
- ilBC
- G.726
- G.722

FreeSWITCH codec preference list:
- G.729
- ilBC
- G.722

Loaded codecs (show codecs):
- G.722
- G.729
- G.722
- G.723.1
- G.726
- G.723.1
Early Negotiation

• Default negotiation mode in FreeSWITCH

• The codec is chosen matching SDP vs inbound-codec-prefs in the SIP profile

• “disable-transcoding” offers the same codec chosen for the inbound leg to the outbound leg

• absolute_codec_str is a good brute-force approach
Early Negotiation

SIP UA

INVITE

codec negotiation

Dialplan

INVITE

SIP UA
Late Negotiation

- “Smarter” approach to codec negotiation
- “inbound-late-negotiation” set to “true” in the SIP profile
- Call will hit the dial plan without looking at codecs
- Negotiation will occur when incoming leg is answered (or requires early media)
Late Negotiation

- You can examine the incoming SDP and re-write SDP to fit your own needs

- “inherit_codec” variable is available to try to use the codec from the B leg for the A leg

- “ep_codec_string” contains the codecs offered by the endpoint
Late Negotiation

- SIP UA
- INVITE
- Dialplan
- codec negotiation
- bridge()
- 183 / 200 OK
- INVITE
- SIP UA
Media Modes
Bypass Media

- Media goes around FreeSWITCH (not through) directly between the endpoints
- SIP signaling stays in FreeSWITCH
- Enable by setting variable “bypass_media=true” before bridging
- Set inbound-no-media or inbound-bypass-media in the SIP profile for a permanent solution
Bypass Media

• You can still play files! (uuid_broadcast)

• uuid_media [off] can re-invite FreeSWITCH on/off the media path

• Recording will fail unless you manually put back FreeSWITCH on the media path
Bypass Media
Proxy Media

• Also called “transparent proxy mode” for the RTP

• No media capabilities enabled

• Only the “c=” part in the SDP is modified

• Allows FreeSWITCH to pass-thru codec media that does not support
Proxy Media

- Set “proxy_media=true” variable before the bridge to enable it
- Set “inbound-proxy-media” in the SIP profile for a permanent solution
- You most likely want to have “late negotiation” enabled
Proxy Media
Sangoma Transcoding
Sangoma Transcoding

• Wider codec support in the industry
• Seen as an ethernet interface by the operating system
• SOAP interface for transcoding control
• Multiple servers can use a single card
• Firmware upgradable on the field
• License upgradable (from 30 licenses to 400)
## Capacity

<table>
<thead>
<tr>
<th>Codec/P Time</th>
<th>10 ms</th>
<th>20 ms</th>
<th>30 ms</th>
<th>40 ms</th>
<th>50 ms</th>
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<tr>
<td>G.729 AB</td>
<td>300</td>
<td>440</td>
<td>459</td>
<td>462</td>
<td>466</td>
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<tr>
<td>G.722</td>
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<td>388</td>
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<tr>
<td>PCM/A</td>
<td>310</td>
<td>390</td>
<td>420</td>
<td>440</td>
<td>460</td>
</tr>
</tbody>
</table>
Single Server Setup

FreeSWITCH
- I/O core
- codec module
- SOAP client (libsngtc-node)
- SOAP server
- API (libsng-tc)

Ethernet Driver

RTP (Voice)

Control (SOAP TCP connection)

Board discovery at Install time
Distributed Setup

- **FreeSWITCH**
  - I/O core
  - codec module
  - SOAP client (libsngtc-node)

- **Transcoding Server**
  - SOAP server
  - API (libsng-tc)
  - Ethernet Driver

- **App Server**
  - SOAP client (libsngtc-node)
Supported Codecs

- G.729
- G.726-32
- G.722
- G.722.1
- G.723.1
- iLBC
- AMR

- *more codecs supported by the D-series cards but not yet implemented by FreeSWITCH codec module*
THANK YOU

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