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Calling Instructions

Wednesday November 3rd at 1700 UTC (1200 CST / 1pm EST)
sip:888@conference.freeswitch.org or via the good old PSTN at +1-919-386-9900
gtalk:conf+888@conference.freeswitch.org Or click on this link
Or call Skype the skype user "skypiax5", you'll be automatically connected (max 20 concurrent users).
Codecs: PCMU/PCMA, G.722, CELT, Speex, Skype, among others
Press 0 to mute/unmute your self. Press * to deaf/undeaf.

- What are the Beeps in the conference? One Beep - someone has joined, two beeps - someone has left.

Agenda

Please add agenda items as needed. If you have a question you'd like to ask then edit the **Items Needing Discussion** section below. Be sure to put your name/nick on your question.

News, Notes, & Miscellaneous Fun Stuff

- bkw is back! Yay!
- anthm added a new session var:

```
<action application="set" data="record_post_process_exec_api=some_api_app:api_app args" />
<action application="set" data="record_post_process_exec_app=some_app:app args" />
```

If you record calls, and the a_leg hangs up, your done, so you have no way to post process the recording - this adds that capability.

Featured Presentation

- DRK: Implementing Carrier-Grade Solution with FreeSWITCH and .NET Tools

Upcoming Presentations

- All Open - Please let me (MSC) or NormT know if you have a topic or idea.

Questions For Developers

- From Chad: what's the difference between "originate_timeout" and "call_timeout"?

Items Needing Discussion

- ClueCon videos update - Mishehu working still, will give you details as they become available

Janitorial Items

- Add items here

Items Needing Documentation

- Add items here

Stuff started but needs some community input

- Add items here

Stuff yet to be documented

- New mod_sofia chan var: sip_force_audio_fmtp (commit msg)
- New mod_sofia chan var: sip_copy_multipart (commit msg)
- New FIFO param: taking_calls (commit msg)
- New FIFO flag: no_media (commit msg)
- New FIFO chan vars: fifo_bridged, fifo_manual_bridged (commit msg)
- New chan var: last_bridge_to (commit msg)
- New stats (in CDRs) rtcp_packet_count and rtcp_octet_count (commit msg)

User Tips & Tricks

If you have something you'd like to share with the community then by all means add it here and we will give you a few minutes on the conference call to discuss it!

Getting Magic Variables

- Wrong: uuid_getvar <uuid> state
- Right: eval uuid:<uuid> state

XMLCDRD

Tricks to receive CDRs http://wiki.freeswitch.org/wiki/Xmlcdrd

Need some FREE text to speech?

```
<extension name="Free_Google_Text_To_Speech">
  <condition field="destination_number" expression="^2115$">
    <action application="answer" data=""/>
    <action application="playback" data="shout://translate.google.com/translate_tts?tl=en&q=Buy+C
  </condition>
</extension>
```

-- NormT

Suggestions For Future Meetings & Future To Dos

- Math: Sofia internals

Presenters Needed For These Topics

- SIP 101 (beginning SIP, how does it work, how to look at different SIP packets)
- mod_shout/shoutcast, esp with one-way conferences for scalability

- Codec negotiation - early vs. late, why you need it, how to do it
- T.38 - what it is, how to use it, etc.
- Multi-tenancy (bounties welcome)

FreeSWITCH High Availability Testing

Weekly FreeSWITCH High Availability testing during the conference call. At the end of the weekly conference call the lead developer of FreeSWITCH (anthm) will crash the conference server. This tests the high Availability that is built into FreeSWITCH.

Depending on what type of NAT device will determine if your call is "Re-Invited" back to the conference.

Need SIP Trunking or DIDs?

If you are looking for SIP Trunking, Origination or Termination please visit www.VoiceNetwork.ca

- Unlimited Incoming DIDs from $3.95
- Per Minute DIDs from $0.99 @ 0.01 per minute
- Outbound Canadian Termination from $0.005 and USA from $ 0.0125

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