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Calling Instructions

Wednesday February 2nd at 1800 UTC (1200 CST / 1pm EST)
sip:888@conference.freeswitch.org or via the good old PSTN at +1-919-386-9900 or +44-3300-100-295
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

- FS Dev dinner tonight! :)
    - Thanks to everyone for pitching in - it is much appreciated

Featured Presentation

- None - community scrum & free for all

Upcoming Presentations

- J. Oquendo: PBX Honeypot and the VoIP abuse project (Has been delayed because of crazy weather in New England)
    - No updated date yet... stay tuned
- All Upcoming Dates Open - Please let me (MSC) or NormT know if you have a topic or idea.

Questions For Developers

- Ask your questions here...

Janitorial Items

- Add your items here...

Items Needing Documentation

- conference_enter_sound : committed 651acc62 (2010-12-21) -> Add a chan var conference_enter_sound to override conference enter-sound param on the profile

Stuff started but needs some community input

- bind_digit_action has cool featureness, per anthm

```
anthm: is it doc'd how you can consume the events fired by bind_digit_action in either the inputc
scripts or over ESL with divert_events enabled and if the name has exec: in it its a shortcut to
```

Stuff yet to be documented

- New FIFO member_add param: taking_calls (commit msg)
- New FIFO flag: no_media (commit msg)

- New FIFO chan vars: fifo_bridged, fifo_manual_bridged (commit msg)
- New FIFO param: outbound_ring_timeout (commit msg)
- New FIFO param: default_lag (commit msg)

- New sip profile param and chan var: manual-rtp-bugs and manual_rtp_bugs for our friends at Sonus (commit msg)

- New sip profile param: presence-probe-on-register (commit msg)

- Chan var I didn't know about: verbose_sdp (set in vars.xml) (See anthm's post on mailing list)

- "Dramatic Jitterbuffer Changes"
    - jitterbuffer API (commit msg)
    - STFU updates (commit msg)

- New API: uuid_buglist (commit msg)

- New API and app: recovery_refresh (commit msg)

- New trick: use * in sofia_contact

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!

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)
- Steve Underwood - SpanDSP, T.38, etc. with FreeSWITCH
- mod_fifo vs. mod_callcenter - why use one or the other? Strengths and weaknesses of each
- embedding FS in other applications (libfreeswitch)

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 intentionally crash and restart the conference

server. This tests the high Availability that is built into FreeSWITCH. Your call should continue once the server is restarted.

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

See Also

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