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Calling Instructions

Wednesday August 25th 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

Featured Presentation

- Philip R Zimmermann - ZRTP with FreeSWITCH

Upcoming Presentations

The dates for the following presentations are subject to change, so you need to check back.

- September 1, 2010 - Moc will be talking about a new modules called mod_CallCenter
- September 8, 2010 - intralanman - mod_xml_curl
- September 15, 2010 - DRK
- September 22, 2010 - OPEN (MSC out of town)
- September 29, 2010 - OPEN

Questions For Developers

If you have questions for Tony, Mike, or Brian please add them here...

- what are the benefits from the management interfaces found in show management?

Items Needing Discussion

- After PRZ presentation...
- Followup on the "FS CPU Usage" mega email thread
    - Who wants to be among the "unbiased" people reporting their usage stats?
    - Add/review stuff on Real-world_results wiki page
    - FS devs will recuse themselves - we are content with 50 CPS ;)
- what about ZRTP negotiation? how does a device negotiates ZRTP security?
- does ZRTP needs a certificate?
- does QoS that drops normal RTP traffic can detect ZRTP also?

Janitorial Items

- uuid_break and break have been documented
- uuid_limit_release and limit_interval_release have been documented
- mod_redis has been documented
- disable_hold variable has been documented
- disable-hold SIP profile param has been documented and here
- session_loglevel app has been documented
- hash_remote has been documented but could use a little more love
- FIFO variables fifo_timestamp and fifo_role have been documented
- conference param terminate_on_silence has been documented
- Variables sip_acl_authed_by and sip_acl_token have been documented.
- Variable conference_member_id has been documented

- verbose_events has been documented
- Thanks to cartes for updating expr and Mod_expr

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)

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!

- Don't forget about the contrib folders! They are now in their own repository. See Git_Tips#Initial_Checkout for information on how to check it out.

Suggestions For Future Meetings & Future To Dos

- Add your thoughts here

Presenters Needed For These Topics

- SIP 101 (beginning SIP, how does it work, how to look at different SIP packets)
- NAT traversal debugging in FS environments (when to use autonat, how to diagnose your NAT routers, etc.)
- 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.

Need SIP Trunking or DIDs?

If you are looking for SIP Trunking, Origination or Termination please visit http://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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