Contents

- 1 Calling Instructions
- 2 Agenda
  - 2.1 News, Notes, & Miscellaneous Fun Stuff
  - 2.2 Featured Presentation
  - 2.3 Upcoming Presentations
  - 2.4 Questions For Developers
  - 2.5 Janitorial Items
  - 2.6 Items Needing Documentation
    - 2.6.1 Stuff started but needs some community input
  - 2.7 Suggestions For Future Meetings & Future To Dos
    - 2.7.1 Presenters Needed For These Topics
- 3 Need SIP Trunking or DIDs?
- 4 See Also

Calling Instructions

Wednesday August 3rd, 2011 at 1700 UTC (1200 CDT / 1pm EDT)
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

- Thank you to all those who've registered for ClueCon and booked their hotel rooms - we've filled our block at the hotel!
- ClueCon has kept us really busy, so I will get to the ChangeLog once we get back
- New speaker at ClueCon: jmspeex (Jean-Marc Valin) discussing OPUS codec!!

Featured Presentation

- Community discussion - new project called "Homer" - SIP logging.

Upcoming Presentations

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

- Eliot Gable's HA stuff (no date yet)
- Polycom (still working on a date)

Questions For Developers

- I would really like to get rid of the 400 "Missing Contact Header" response when a SIP REGISTER without Contact header is sent to FS. According to RFC3261 this should be allowed, as this is a way for a UAC to fetch all the current bindings from a registrar for a certain user (I have some devices that do this every 1800 sec and each try gives me a nice red error line on my cli). I created a Jira ticket (FS-2875), wrote a patch and added a trace that shows what changed. Can someone check it out, give feedback and if it's ok apply and commit it ? Thanks, Leon (ledr on irc)
- Add more questions

Janitorial Items

- Community members organizing a GMV sounds order - let MSC know if you want to pitch in.

Items Needing Documentation

- Add your items here

Stuff started but needs some community input

- Add your items here

Suggestions For Future Meetings & Future To Dos

- Math: Sofia internals
- Eliot Gable: mod_ha_cluster
    - What is mod_ha_cluster?
        - In short: N+X (N Masters + X slaves) HA replacement for "Pacemaker + Corosync managed FS"
    - What is planned for mod_ha_cluster?

        - In short: manage interfaces, IPs, firewall rules, master/slave status; eventually, share registrations and call states without shared DB and do 'sofia recover' on failover
    - When will it be available?
        - In short: Master/slave, interface, IPs, and firewall management within the next couple of months (if I have the amount of time available that I expect to have)
    - For additional details, or if you would like to beta test, be on the call
- Jeff Lenk: developing for FS in Windows environment

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)
- IPv6 - what it is, how to use, differences with IPv4, how to configure FS

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
- Free iNum's for each account

See Also

Return to main meeting page