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Calling Instructions

Wednesday May 4th, 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.

Note to Collins: RECORD THE CONFERENCE!!!

News, Notes, & Miscellaneous Fun Stuff

- Sign up for ClueCon... or ELSE!

Featured Presentation

- None - Community gets to speak

Upcoming Presentations

- Eliot Gable's HA stuff (no date yet)
- Polycom (still working on a date)
- All Upcoming Dates Open - Please let me (MSC) or NormT know if you have a topic or idea.

Questions For Developers

- Add your questions here

Janitorial Items

- Add your items here

Items Needing Documentation

- Dialstring feature: ^^ allows you to use a separator other than comma in variable values in dialstring (useful in { } and [] blocks) (commit note)
    - e.g. <action application="bridge" data="[execute_on_ring=info,absolute_codec_string=^^:PCMA:G722]sofia/192.168.10.1/sip:2103@1
    - code comment: "this is stupid but necessary: if the value begins with ^^ take the very next char as a delim, increment the string to start the next char after that and replace every instance of the delim with a ,"
- uuid_preprocess in mod_commands
- 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

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

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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