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Calling Instructions

Wednesday November 9th, 2011 at 1700 UTC (1200 CDT / 1pm EDT)
sip:888@conference.freeswitch.org or via the good old PSTN at +1-919-386-9900
iNum:+44-3300-100-295
Australia: +61 7 3188 7519 (Thanks to Jay Binks - NetSIP.com.au)
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.

Screen sharing: when we do screen sharing we frequently use this:

http://remote.voipproviderinfo.net/

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

- NYC FreeSWITCH meetup - http://bit.ly/sYmtx2
- FSCB Update
- CC Videos update
- FS Wiki: "ConfirmAccounts" - If you are (or can be) Wiki administrator let me know and I'll show you how it works

Featured Presentation

- Review: MC's "channel variables script"
- SIP 101: NDLB settings (force-rport and connectile-dysfunction)

Questions For Developers

- Is there a performance benefit for a directory or dialplan module to directly update user and channel variables in Core FS as opposed to replying back to FS with an XML? We are in the process of deving a module that establishes a direct link between FS and a backend DB . We started out successfully by modifying good ole mod_xml_curl by replacing the HHTP/HTTPS component with a plain Jane C API conduit to respective DBs. We basically reply back with an XML prepared by the DB query. I just cannot help thinking, "What a waste of resources", we contsruct a XML just for FS to parse through the XML and place the items in user and channel classes. I wonder if we can write direct to FS and make it perform a little better by not having XML in the picture at all.

Janitorial Items

- Still need Drupal/PHP helper or even someone w/ HTML/JS/CSS skill

Items Needing Documentation

- Add your items here

Stuff started but needs some community input

- Add your items here

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)

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

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