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Calling Instructions

Wednesday May 2n, 2012 at 1700 UTC (1200 CDT / 1300 EDT)
sip:888@conference.freeswitch.org or via the good old PSTN at +1-919-386-9900
UK:+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.

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

- Coming soon: Giovanni Maruzzelli talking GSM.
- Coming soon: Fred Dixon and the gang from Big Blue Button
  - They love FreeSWITCH! Will tell us about their experiences with Asterisk vs. FS and how FS makes their software possible

Featured Presentation

- Vestec discussing latest version of their ASR engine.

Questions For Developers

- Add your questions here

Janitorial Items

- Add your questions here

Items Needing Documentation

- Add your items here

Items Needing Discussion

kimc wonders about:

How to use the dialing of destination numbers to set/unset variables which set options in a dialplan.

Ex. dial 8201 for day mode, 8200 for night mode

Similar to the example in dialplan/default.xml for add-group/del-group

Stuff started but needs some community input

- Add your items here

Upcoming Presentations

- All future dates are open
- "Coming Soon"
    - Giovanni Maruzzelli & GSM Open
    - Plivo update
    - Vestec ASR 2.0

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
- Echo
    - What/Why/Where
    - Troubleshooting
- contributing
    - How non C programmers can help out - its had to write docs when you dont really know whats going on and referred to the code.

Presenters Needed For These Topics

- Internals of modules like mod_callcenter and mod_conference
- 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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