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Calling Instructions

Wednesday January 11th, 2012 at 1800 UTC (1200 CST / 1300 EST)
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

- Assistance needed: Mailing List
  - We could use some help answering questions on the ML. Please subscribe and help us answer the flood of questions coming in. :)
- New module: mod_httpi - Hyper Text Telephony API (Think Twilio) for FreeSWITCH. Still in beta; if you have a chance to tinker with it please do so.
  - All docs are in src/mod/applications/mod_httpi/mod_httpi_doc.txt.
  - We will put on wiki once we have a little experience with this new module
  - Feel free to submit your code snippets and ideas
- Check out Linux builds on build.freeswitch.org:8111
  - If you want to add your own build agent let me know (It's easy once you know what steps to take.)
- Need more ideas for future conference calls. We have some openings coming up. If you have a topic or idea please let me know.

Featured Presentation

- Moises Silva from Sangoma, discussing PRI card setup, new GSM goodies
  - Presentation file: http://www.moythreads.com/congresos/fsconfcall2012/

Questions For Developers

- Add your questions here

What does "Failed DTMF sanity check" really mean?

Janitorial Items

- Add your items here

Items Needing Documentation

- Add your items here

Stuff started but needs some community input

- Add your items here

Upcoming Presentations

- Jan 18 - Moc: mod_voicemail_ivr
- Jan 25 - Muhammed Naseer, vBilling
- Feb 1 - Robert Daniel from GWU, discussing cybersim.net; "Using Games to Teach Telecom"
- Feb 8 - OPEN
- Feb 15 - 2600hz Project
- Feb 22 - Stefan Wintermeyer, Gemeinschaft 4

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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