Contents

- 1 Scheduled time
- 2 Calling Instructions
- 3 Agenda
  - 3.1 News, Notes, & Miscellaneous Fun Stuff
  - 3.2 Featured Presentation
  - 3.3 Questions For Developers
  - 3.4 Janitorial Items
  - 3.5 Items Needing Documentation
  - 3.6 Items Needing Discussion
  - 3.7 Stuff started but needs some community input
- 4 Upcoming Presentations
  - 4.1 Suggestions For Future Meetings & Future To Dos
  - 4.2 Presenters Needed For These Topics
- 5 Need SIP Trunking or DIDs?
- 6 See Also

Scheduled time

Wednesday May 30th, 2012 at 1700 UTC (1200 CDT / 1300 EDT)

Calling Instructions

- SIP
  - sip:888@conference.freeswitch.org
  - Codecs supported: PCMU/PCMA, G.722, CELT, Speex, among many others
- PSTN

Country	Number	Contributor
USA	+1-919-386-9900	
Spain	+34-91-290-12-71	Thanks to SIPtize
UK	+44-330-320-0105	Thanks to ziron.net
UK	+44-1904-201-313	Thanks to ukddi.com (Routed by Steven Ayre)
UK	+44-203-298-5931	Thanks to ukddi.com (Routed by Avi Marcus)
Germany	+49-228-9293-9009	Thanks to Yiftach at ChooChee
Australia	+61-7-3188-7519	Thanks to Jay Binks - NetSIP.com.au
Israel	+972-2-372-0394	Thanks to Avi Marcus - BestFone.com
Canada	+1-438-800-0531	Thanks to NG Communications
France	+33-975-181-606	Thanks to NG Communications
Netherlands	+31-858-880-387	Thanks to NG Communications

South Africa	+27-87-8204656	Thanks to Othos Telecom

- Google Talk
  - conf+888@conference.freeswitch.org
- Skype
  - freeswitch_conf_call (max 20 concurrent users)
  - Use *Call->Show Dialpad* to unmute
- Flash (in-browser VoIP RTMP client)
  - conference.freeswitch.org

Note: When you join you will initially be **muted** and need to press 0 if you wish to speak (this reduces background noise on the conference).

At some times muting may be moderated (eg presentations). Pressing 0 will put you in a queue and you will be unmuted when it's your turn to speak.

Be sure to be logged into #freeswitch on irc.freenode.net! There are frequently related comments during the conference.

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

- Don't forget to sign up for ClueCon
- Note: If you asked about free Vestec starter kits you need to contact Kashif Kahn to get signed up. Only a few people have contacted him thus far...
- FYI, if you have a recording of last week's conf call please let me know. -MC
- Note: FSClient has been updated. If MitchCapper is around we'll hit him up for some info. :)
  - http://wiki.freeswitch.org/wiki/FSClient

Featured Presentation

- Community discussion (Bring your topic!)
  - MSC: Let's discuss sound prompts in various languages. I'd like to get a bead on what sounds are stored where, etc.
    - Also, TC and I have nailed down some language that makes our recordings "properly licensed in a FOSS-friendly way."
  - MSC: Do you own the FS 1.0.6 "Bridge Book"? If so, I'd like talk to you about edits, omissions, corrections, etc.

Questions For Developers

- Add your questions here

Janitorial Items

- Add your questions here

Items Needing Documentation

- Add your items here

Items Needing Discussion

- Add your items here

Stuff started but needs some community input

- Add your items here

Upcoming Presentations

- June 2013
    - June 19 - FreeSWITCH Dev Team
    - June 26 - Lorenzo/Alexandr (HOMER 3.5)

- "Coming Soon"
    - Plivo update
    - Other surprises

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"
    - For additional details, see mod_ha_cluster
- Jeff Lenk: developing for FS in Windows environment
- Echo
    - What/Why/Where

    - Troubleshooting
- Contributing
    - How non-C programmers can help out - it's hard to write docs when you don't really know what's going on and referred to the code.

Presenters Needed For These Topics

- Internals of modules like mod_callcenter and mod_conference
- mod_shout/shoutcast/mod_vlc, 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
- WebRTC using Freeswicth

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