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Calling Instructions

Wednesday June 29th, 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

- FSCB Update
  - We have decided to remove the "call center" and "carrier grade" chapters
    - We will keep the most useful of those recipes and put them elsewhere
    - We need more ideas on recipes!

        - Let us know what you think - what specific things would you like to see in recipe form?

- Fun stuff gleaned from doing the ChangeLog
    - fs_cli: added timeout options for API commands (Thanks Math)
        - -t or --timeout <ms>
    - Arrays available in chan vars!
        - see new dp apps: capture, push, unshift
        - For ESL, pushHeader and unshiftHeader wrappers available (needs docs please)
    - API 'conference' can get exact count of members:
        - conference xxxx list count
    - Minor xml config update: I added conf/lang/en/ivr/ so that there is an obvious place to put IVR-ish phrase files that don't fit under "demo" or "vm"
    - New chan var: deny_refer_requests
        - Allows you to deny REFER requests (needs docs please)
    - Added append flag to mod_shout - can now append to MP3 files
        - (I did not see where/how to set the flag - any shout gurus who can fill us in?)
    - Anthony added the ability transfer recording media bugs to the other leg when doing an attended transfer!
    - New chan vars:
        - flip_record_on_hold to make the recording flip to the other leg on hold
        - record_check_bridge to make recording the same file on the opposite leg of a bridge considered a duplicate attempt
        - record_toggle_on_repeat to make repeat recording the same file toggle the recording off
            - (needs docs please)
    - New sofia command "check_sync" - like flush_inbound_reg but doesn't actually unreg
        - sofia profile xxx check_sync <call_id> | <user@domain>
    - New conference member flag: nomoh (anyone wanna guess what it does? :)
    - Coolness: scoped channel variables (%[var=val,var2=val2] blocks valid in any app data field and will only last for that one app execution)
        - Please test this and give me your feedback - I have not gotten it to work just yet
    - New conf command: hup (kick w/o the kick sound)
        - conference <conf> hup <member_id>
    - New SIP compatibility items - makes FS deal with all those retarded implementations that don't follow the spec
        - Chan var **sip_liberal_dtmf**
        - Param profile **liberal-dtmf**
    - FreeSWITCH status and control updates, e.g. if fsctl pause has been issued
        - **status** shows "FreeSWITCH is ready/not ready"
        - **fsctl pause_check** - returns true if fsctl pause is in effect
        - **fsctl ready_check** - returns true if FreeSWITCH is ready
    - New reporting vars:
        - **last_hold_time** - last time call was held
        - **hold_accum** - total amount of time call was on hold

- After Venky, mitchcapper will update us on mod_portaudio changes

Featured Presentation

- Venky to discuss plivo.org
    - Download the PPT 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)

Questions For Developers

- Add your questions

Janitorial Items

- Community members organizing a GMV sounds order - let MSC know if you want to pitch in.

Items Needing Documentation

- 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
- 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

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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