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Calling Instructions

Wednesday August 11th at 1700 UTC (1200 CST / 1pm EST)
sip:888@conference.freeswitch.org or via the good old PSTN at +1-919-386-9900
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

- The book is here!!!
  - Please submit any error reports through the Packt website

Featured Presentation

- DRK: mod_managed for billing and CDRs

Upcoming Presentations

The dates for the following presentations are subject to change, so you need to check back.

- August 4, 2010 - CLUECON BABY!
- August 11, 2010 - DRK - mod_managed for billing and CDRs
- August 18, 2010 - bkw - NAT and VoIP: let's figure it out
- August 25, 2010 - Moc from MocTel will be talking about a new modules called mod_CallCenter

Questions For Developers

If you have questions for Tony, Mike, or Brian please add them here...

Items Needing Discussion

- Add items here

Janitorial Items

- Chan vars - let's crowdsource the descriptions of these vars
- Need wiki volunteer: create_wiki_page.pl script
    - auto create new pages for chan vars, APIs, modules, etc.
- Need volunteers to assist with the mail list moderation queue
    - CMF volunteered to assist during weekdays (GMT -7 aka Los Angeles)

Items Needing Documentation

We definitely need assistance with documenting stuff, especially new channels variables. A sample template for doing a channel variable can be found here: Variable_skeleton

I also added a skeleton API: API_skeleton template page

Stuff started but needs some community input

- Most important documentation task: T.38. Please see the email thread, especially bkw's comments.
    - Please add content to this page: T.38

Stuff yet to be documented

- After the email thread about eavesdropping it looks like we need to document some mod_shout stuff:
    - how to uuid_record to an icecast server (including basic configs needed in icecast.xml)
    - how to use the telecast API (bonus: add syntax info to mod_shout.c)
- New conference param: terminate-on-silence (commit msg)
- Verbose events:
    - global param verbose-channel-events, fsctl verbose_events [on|off]
    - commit msg
- New "disable hold" feature:
    - disable_hold channel variable
    - disable-hold profile param
    - (commit msg)
- Channels to check if you have been auth'd:
    - sip_acl_authed_by
    - sip_acl_token
    - commit msg
- New mod_conference chan var: conference_member_id (commit msg)
- New mod_sofia chan var: sip_force_audio_fmtp (commit msg)
- New mod_sofia chan var: sip_copy_multipart (commit msg)
- break and uuid_break

User Tips & Tricks

If you have something you'd like to share with the community then by all means add it here and we will give you a few minutes on the conference call to discuss it!

- Don't forget about the contrib folders! They are now in their own repository. See Git_Tips#Initial_Checkout for information on how to check it out.

Suggestions For Future Meetings & Future To Dos

- Add your thoughts here

Presenters Needed For These Topics

- SIP 101 (beginning SIP, how does it work, how to look at different SIP packets)

- NAT traversal debugging in FS environments (when to use autonat, how to diagnose your NAT routers, etc.)
- 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.

Need SIP Trunking or DIDs?

If you are looking for SIP Trunking, Origination or Termination please visit http://www.VoiceNetwork.ca

- Unlimited Incoming DIDs from $3.95
- Per Minute DIDs from $0.99 @ 0.01 per minute
- Outbound Canadian Termination from $0.008 and USA from $ 0.0125

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