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Calling Instructions

Wednesday July 28st 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

- FS book update - Release for August
  - New cover: File:9966 cover ForAuthor.pdf, no longer looks like a feminine hygiene commercial

    - MSC has proofread and submitted chaps 1-8; chap 9 is done except I need Pyite
    - Packt isn't talking to me (again)
    - Probably will see book in early Aug

Featured Presentation [[intra]lanman - mod_xml_curl]

- [intra]lanman talks about mod_xml_curl, get your questions ready !

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...

- For Tony: you've created what is possibly the most advanced telephony software on the planet, overcoming huge obstacles and meeting intense engineering challenges. All that being said, which text editor do you use?

Items Needing Discussion

- ClueCon Update
    - No more hotel, no more $599 discount, no more speaking slots. :)
- Is anyone using T.38?
    - Please let us know how it's working for you
    - Which carrier(s) - www.VoiceNetwork.ca

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

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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