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Calling Instructions

Wednesday September 15th at 1700 UTC (1200 CST / 1pm EST)
sip:888@conference.freeswitch.org or via the good old PSTN at +1-919-386-9900
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.

News, Notes, & Miscellaneous Fun Stuff

Featured Presentation

- DRK: Billing, CDR's, with Mod_managed

Upcoming Presentations

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

- September 15, 2010 - DRK
- September 22, 2010 - lanman mod_xml_curl
- September 29, 2010 - OPEN

Questions For Developers

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

Items Needing Discussion

- We are still looking for Cookbook recipe ideas. We will be asking the community frequently for ideas. If you have ideas please please please put them on the Cookbook page. If we get enough input from the community we probably will be able to do a new FS cookbook with Packt.
    - Today is the last day to submit recipe ideas before we submit a proposal to Packt.

From kimc

FreeSWITCH version: 1.0.head (git-70331e8 2010-08-30 17-33-05 -0500) FreeBSD 8.1-stable and 9-current Voicemail recorded by G722 phones is very distorted while 'greeting' recordings sound fine. The wav files are distorted when played on other software/os Voicemail recorded with the same hardware with the phone in G711 mode sounds fine.

A 'workaround' is to call the voicemail app without the use of loopback:

```
<action application="voicemail" data="default ${domain_name] $1"/>
```

instead of:

```
<action application="bridge" data="loopback/app=voicemail:default ${domain_name} ${dialed_extensi
```

- Difference between mod_fifo and mod_callcenter, to make a better call center environment which mod is better and in what aspects ?

- With the XML IVR engine how would you handle the case where there is no dtmf user input? One way is to let it timeout where it returns to the dialplan in the location where it was called. A problem then is a follow-me bridge doesn't work correctly when it follows a 'timed-out' ivr session like this.
- How to count ASR &ACD & PDD?
    - Not to mention other quality measures including CSR, ABR, NER
    - PDD = time until progress media (or answer if no progress) = (progress_media_uepoch>0?progress_media_uepoch:answer_uepoch)-start_uepoch_uepoch, also see progress_mediasec.progress_mediamsec etc.
    - Other measures require storing CDRs in a database and summarising the data (or updating counters)
        - ACD = total_duration / num_answered
        - ABR = (num_answered / num_calls) * 100% (num_calls being both successful and failed calls)
        - ASR = (num_answered / num_terminated) * 100% (num_terminated being successful calls only)
        - CSR = (num_terminated / num_calls) * 100%
        - NER = (Answers + User Busy + Ring No Answer + Terminal Rejects) / Total call attempts (seizures) [ITU E.411]

Janitorial Items

- The show command docs could use some love. They are really sparse. It would be nice to have an example of some of things you can do with actual examples like:

```
show channels as xml
show channels like foo as xml
show distinct_channels
show calls
```

Let me know if you think we can add anything else to the show page. Some of the show commands are quite obvious, like "show api" and "show application" so we don't need to do too much on those.

Items Needing Documentation

- Add items here

Stuff started but needs some community input

- Add items here

Stuff yet to be documented

- New mod_sofia chan var: sip_force_audio_fmtp (commit msg)
- New mod_sofia chan var: sip_copy_multipart (commit msg)
- New FIFO param: taking_calls (commit msg)

- New FIFO flag: no_media (commit msg)
- New FIFO chan vars: fifo_bridged, fifo_manual_bridged (commit msg)
- New chan var: last_bridge_to (commit msg)

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)
- 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.

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 crash the conference server. This tests the high Availability that is built into FreeSWITCH.

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

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