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Calling Instructions

Wednesday October 6th 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

- New sounds order going in - let me (MSC) know ASAP if FreeSWITCH is needing any English sound prompts recorded
- New additions to default dialplan (see below)
- New executable program in FS: tone2wav (commit msg)
    - Generates a wave file from TGML
- New handy sofia command (commit msg)
    - sofia profile <profile> gwlist up|down
    - Useful for feeding into mod distributor to exclude dead gateways

Featured Presentation

- Hopefully Darren Schreiber (pyite) will be calling in to give us an update from ITExpo as well as blue.box.

Upcoming Presentations

- All Open - Please let me (MSC) or NormT know if you have a topic or idea.

Questions For Developers

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

Items Needing Discussion

- Default dialplan updates in September
    - Resolve some SCA issues where host and IP are mixed (commit msg)

```
In Local_Extension, conf/dialplan/default.xml:

-    <action application="bridge" data="user/${dialed_extension}@${domain_name}"/>
+    <action application="bridge" data={sip_invite_domain=$${domain}}user/${dialed_extension}@${d
```

- - Add att_xfer example to dialplan (commit msg1 and commit msg2)

```
In Local_Extension, conf/dialplan/default.xml:

+    <action application="bind_meta_app" data="4 b s execute_extension::att_xfer XML features"/>

In conf/dialplan/features.xml:

+    <extension name="att_xfer">
+     <condition field="destination_number" expression="^att_xfer$">
+       <action application="read" data="3 4 'tone_stream://%(10000,0,350,440)' digits 30000 #"/>
+       <action application="set" data="origination_cancel_key=#"/>
+       <action application="att_xfer" data="user/${digits}@$${domain}"/>
```

```
+     </condition>
+    </extension>
```

- Internal SIP Profile Change
    - From BKW: improve defaults to cover strange behaviors (commit msg)

```
In conf/sip_profiles/internal.xml

 -    <!--<param name="presence-hosts" value="$${domain}"/>-->
 +    <param name="presence-hosts" value="$${domain},$${local_ip_v4}"/>
```

- ClueCon videos update

Janitorial Items

- Need some assistance with Jira: when MikeJ did Jira updates some didn't get all their attachments. We need Jira-capable people to assist with Jira cleanup - need to make sure all Jira's have their attachments. (more details forthcoming)

Items Needing Documentation

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)
- New stats (in CDRs) rtcp_packet_count and rtcp_octet_count (commit msg)
- Moc's new tod thing (commit msg)
- Moc's new date/time thing (commit msg)
- Moc's new 3-letter day thing (commit msg)
- New dp app: bridge_export (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.

New sofia sip tracing

```
sofia global siptrace on
```

Nik Martin - Just tooting my own horn. Click to dial from OS X Address Book:

/freeswitch-contrib/nmartin/osx_address_book_dial

Suggestions For Future Meetings & Future To Dos

- Moy: FreeTDM
- Math: Sofia internals

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)

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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