

# **FreeSWITCH in REAL LIFE on Centos or OS X**

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# Prerequisites for Centos and OS X

- **SVN**: for the access to the current code repository
- **GNUMAKE**: The GNU version of make.
- **AUTOCONF**: Version 2.50 or higher
- **AUTOMAKE**: Version 1.70 or higher
- **GLIBTOOL**: Version 1.5.14 or higher
- **GCC**: Version 3.95 or higher
- **WGET**: Almost any version
- **LIBNCURSES**: Almost any version
- **GNUTLS**: googletalk and TLS support

## Centos:

*“yum install subversion autoconf automake libtool gcc-c++ ncurses-devel make”*

## OS X

- Install Apple [XCode development software](#) (free registration required)
- Install package manager [Fink](#) (similar to yum of centos)
- *“fink install svn-client make automake1.9 autoconf2.5 libtool14 wget”*
- *No fink package for GNUTLS under OS X, has to be compiled from source*

# Prepare to compile Freeswitch for Stable Release

- `cd /usr/src`
- `wget http://files.freeswitch.org/freeswitch-1.0.1.tar.gz`
- `tar zxvf freeswitch-1.0.1.tar.gz`
- `cd freeswitch-1.0.1`
- `./configure` ( **Freeswitch default prefix= /usr/local/freeswitch**) or
- `./configure --prefix=/opt/local/freeswitch` ( **Install Freeswitch under /opt/local/freeswitch**)
- `cp build/modules.conf.in modules.conf`

**Edit modules.conf for the modules to build, default build without**

**Googletalk support** (`endpoints/mod_dingaling`)

**TTS support** (`asr_tts/mod_flite` or `asr_tts/mod_cepstral`)

**ASR support** (`asr_tts/mod_pocketsphinx`)

# Prepare to compile Freeswitch for SVN Trunk

- `cd /usr/src`
- `svn checkout http://svn.freeswitch.org/svn/freeswitch/trunk freeswitch`
- `cd freeswitch`
- `./bootstrap.sh`
- `./configure` ( **Freeswitch default prefix= /usr/local/freeswitch**) or
- `./configure --prefix=/opt/local/freeswitch` ( **Install Freeswitch under /opt/local/freeswitch**)
- `cp build/modules.conf.in modules.conf`
- **Edit modules.conf for the modules to build, default build without**
- **Googletalk support** (`endpoints/mod_dingaling`)
- **TTS support** (`asr_tts/mod_flite` or `asr_tts/mod_cepstral`)
- **ASR support** (`asr_tts/mod_pocketsphinx`)

# Compile and Install Freeswitch

## Freeswitch Binary Code

- ◆ *make*
- ◆ *make install*

## Freeswitch Sounds Files and Music On Hold Files

### 8K HZ

*make sounds-install*

- ◆ *make moh-install*

### 16K HZ

- ◆ *make hd-sounds-install*
- ◆ *make hd-moh-install*

### 32K HZ

- ◆ *make uhd-sounds-install*
- ◆ *make uhd-moh-install*

# Freeswitch directories

## **/usr/local/freeswitch**

**/bin** (binary codes)

**/conf** (configuration files)

**/autoload\_configs** ( acl/ivr/conference/asr/voicemail/switch configuration etc)

**/dialplan** ( dialplans for default/public contexts, similar to extensions.conf in \*)

**/directory** ( for user registration files)

**/default** ( all users registered under default contexts are here)

**/jingle\_profiles** ( for googletalk client mode and component mode settings)

**/lang** ( for multi-language supports including the IVR macro definitions)

**/sip\_profiles** ( different SIP User Agent Profiles, similar to sip.conf in \*)

**internal.xml and /internal** (profile for internal SIP UA on port 5060, all users need authentication)

**external.xml and /external** (profile for public SIP UA on port 5080, no authentication required, used for SIP trunks/gateways setup )

**nat.xml and /nat**(profile for NAT SIP UA on port 5070, no authentication required)

**/ssl** (TLS support for SIP)

**/CA** ( private certificate files created by Freeswitch for supporting TLS)

**/db** (runtime sqlite database files created by Freeswitch)

**/grammar** ( grammar files for ASR applications)

**/scripts** ( all Javascript, Lua, perl scripts going here)

**/sounds** (all sounds and music files going here)

**/log** (CDR and freeswitch logs going here)

**/storage** ( all voice mails local storage)

**/mod** ( modules binary codes all here)

# Freeswitch configuration cheatsheet

- conf/**vars.xml** (Global variables)
- conf/autoload\_configs/**modules.conf.xml** (modules to load)
- conf/autoload\_configs/conference.conf.xml (Conference setup)
- conf/autoload\_configs/ivr.conf.xml (IVR menus setup)
- conf/autoload\_configs/**switch.conf.xml** (switch wide setup including RTP ports range)
- conf/autoload\_configs/voicemail.conf.xml ( voicemail global setup)
- dialplan/**default.xml** and **public.xml** ( dialplans for default and public contexts)
- sip\_profiles/**internal.xml** ( SIP profile for internal users on port 5060)
- sip\_profiles/**external.xml** ( SIP profile for external users on port 5080)
- sip\_profiles/external/ vbuzzer.**xml**( SIP profile for external SIP trunk vbuzzer)
- directory/default/**1000.xml** (user configurations for the user/extension 1000)
- conf/jingle\_profiles/client.xml ( Googletalk setup for the client mode)
- conf/jingle\_profiles/server.xml ( Googletalk setup for the component mode)

# VAR.S.XML

- `<X-PRE-PROCESS cmd="set" data="hold_music=local_stream://moh" />`
- `<X-PRE-PROCESS cmd="set" data="global_codec_prefs=G722,PCMU,PCMA,H261,H263,H263-1998,H263-2000,H264" />`
- `<X-PRE-PROCESS cmd="set" data="xmpp_server_profile=jabber.yourcorp.com" />`
- `<X-PRE-PROCESS cmd="set" data="external_rtp_ip=aa.bb.cc.dd" />`
- `<X-PRE-PROCESS cmd="set" data="external_sip_ip=aa.bb.cc.dd" />`
- `<X-PRE-PROCESS cmd="set" data="outbound_caller_name=FreeSWITCH" />`
- `<X-PRE-PROCESS cmd="set" data="outbound_caller_id=0000000000" />`
-

# Modules.conf.xml

- <!-- Endpoints -->
- <load module="mod\_dingaling"/>
- <load module="mod\_sofia"/>
- <!-- Applications -->
- .....
- <load module="mod\_conference"/>
- <load module="mod\_fifo"/>
- <!-- Languages -->
- <load module="mod\_spidermonkey"/>
- <load module="mod\_lua"/>
- <!-- ASR /TTS -->
- <load module="mod\_flite"/>
- <load module="mod\_pocketsphinx"/>
- <!-- Codec Interfaces -->
- <load module="mod\_voipcodecs"/>
- <load module="mod\_h26x"/>
- <!-- Say -->
- <load module="mod\_say\_en"/>

# Switch.conf.xml

- <!--Colorize the Console -->
- <param name="colorize-console" value="true"/>
- <!--Most channels to allow at once -->
- <param name="max-sessions" value="1000"/>
- <!--Most channels to create per second -->
- <param name="sessions-per-second" value="30"/>
- ...
- <param name="mailer-app" value="sendmail"/>
- <param name="mailer-app-args" value="-t"/>
- <param name="dump-cores" value="yes"/>
- ...
- <!--RTP port range -->
- <param name="rtp-start-port" value="1650"/>
- <param name="rtp-end-port" value="4560"/>

## sip\_profiles/internal.xml

- <param name="user-agent-string" value="FreeSWITCH Rocks!"/>
- <param name="debug" value="0"/>
- <param name="sip-trace" value="no"/>
- **<param name="context" value="public"/>**
- <param name="rfc2833-pt" value="101"/>
- <param name="sip-port" value="5060"/>
- <param name="dialplan" value="XML"/>
- <param name="dtmf-duration" value="100"/>
- <param name="codec-prefs" value="\$\$\${global\_codec\_prefs}"/>
- <param name="use-rtp-timer" value="true"/>
- <param name="rtp-timer-name" value="soft"/>
- <param name="rtp-ip" value="\$\$\${local\_ip\_v4}"/>
- <param name="sip-ip" value="\$\$\${local\_ip\_v4}"/>
- <param name="hold-music" value="\$\$\${hold\_music}"/>
- <param name="dtmf-type" value="rfc2833"/>
- **<param name="manage-presence" value="true"/>**
- **<param name="multiple-registrations" value="true"/>**

## sip\_profiles/external/vbuzzer.xml

- <include>
- <gateway name="vbuzzer.com">
- <param name="username" value="yourvbuzzerusername"/>
- <param name="realm" value="vbuzzer.com"/>
- <param name="from-user" value="yourvbuzzerusername"/>
- <param name="from-domain" value="vbuzzer.com"/>
- <param name="password" value="yoursecret"/>
- <param name="proxy" value="vbuzzer.com:80"/>
- <param name="register" value="true"/>
- <param name="register-transport" value="udp"/>
- <param name="caller-id-in-from" value="false"/>
- <param name="ping" value="15"/>
- </gateway>
- </include>

## jingle\_profiles/client.xml

- <profile type="client">
- <param name="name" value="xmppc"/>
- <param name="login" value="yourgmail@gmail.com/talk"/>
- <param name="password" value="secret"/>
- <param name="server" value="talk.google.com"/>
- <param name="dialplan" value="XML"/>
- <param name="message" value="FreeSwitch (The Future of Telephony)"/>
- <param name="rtp-ip" value="192.168.1.x"/>
- <!-- if you are behind NAT, you should probably use stun -->
- <param name="ext-rtp-ip" value="your external IP here"/>
- <param name="auto-login" value="true"/>
- <param name="auto-reply" value="Press \*Call\* to join my conference"/>
- <param name="sasl" value="plain"/>
- <param name="tls" value="true"/>
- <param name="exten" value="1000"/>
- </profile>

# jingle\_profiles/server.xml

- <include>
- <profile type="component">
- <param name="name" value="jabber.yourcorps.com"/>
- <param name="password" value="secret"/>
- <param name="dialplan" value="XML"/>
- <param name="context" value="public"/>
- <param name="ext-rtp-ip" value="aa.bb.cc.dd"/>
- <param name="rtp-ip" value="192.168.1.xx"/>
- <param name="server" value="jabber.yourcorps.com:5347"/>
- <param name="use-rtp-timer" value="true"/>
- <param name="exten" value="\_auto\_"/>
- </profile>
- </include>

## Directory/default/1000.xml

- `<user id="1000" mailbox="1000">`
- `<params>`
- `<param name="password" value="1234"/>`
- `<param name="vm-password" value="1000"/>`
- `</params>`
- `<variables>`
- `<variable name="accountcode" value="1000"/>`
- `<variable name="user_context" value="default"/>`
- `<variable name="effective_caller_id_name" value="Extension 1000"/>`
- `<variable name="effective_caller_id_number" value="1000"/>`
- `</variables>`
-

## Dialplan/default.xml

- `<extension name="10digitlocal">`
- `<condition field="destination_number" expression="^((416|647)\d{7})$">`
- `<action application="set" data="hangup_after_bridge=true"/>`
- `<action application="bridge"`
- `data="sofia/gateway/vbuzzer.com/1$1|sofia/gateway/voipstunt.com/1$1|`
- `sofia/gateway/voipraider.com/1$1"/>`
- `</condition>`
- `</extension>`
  
- `<!-- a sample IVR -->`
- `<extension name="ivr_demo">`
- `<condition field="destination_number" expression="5000">`
- `<action application="answer"/>`
- `<action application="sleep" data="2000"/>`
- `<action application="ivr" data="demo_ivr"/>`
- `</condition>`
- `</extension>`

# Asterisk Interconnection

[http://wiki.freeswitch.org/wiki/Connecting\\_Freeswitch\\_And\\_Asterisk](http://wiki.freeswitch.org/wiki/Connecting_Freeswitch_And_Asterisk)

- Freeswitch
- autoload\_configs/acl.conf.xml
  - `<list name="asterisk_box" default="deny">`
  - `<node type="allow" cidr="192.168.1.1/32"/>`
  - `</list>`
- sip\_profiles/external.xml
  - `<param name="apply-inbound-acl" value="asterisk_box"/>`
- dialplan/default.xml
  - `<extension name="ast_extens">`
  - `<condition field="destination_number" expression="^(2\d{3})$">`
  - `<action application="bridge" data="sofia/external/$1@192.168.1.1"/>`
  - `</condition>`
  - `</extension>`
- Asterisk (sip.conf)
  - [192.168.1.100]
  - type=peer
  - host=192.168.1.100
  - port=5080
  - disallow=all
  - allow=ulaw
  - trustpid=yes
  - sendrpid=yes

# Exchange 2007 UM Integration

[http://wiki.freeswitch.org/wiki/Exchange\\_2007\\_UM](http://wiki.freeswitch.org/wiki/Exchange_2007_UM)

- dialplan/default.xml
- Under local extensions
- `<condition field="destination_number" expression="^{caller_id_number}$">`
- ...
- `<!--<action application="voicemail" data="check default $$domain ${dialed_ext}"/>-->`
- `<action application="export" data="sip_h_Diversion=${dialed_ext}"/>`
- `<action application="bridge" data="{absolute_codec_string=PCMU}sofia/internal/${dialed_ext}@192.168.1.250;transport=tcp"/>`
- ...
- `<!--<anti-action application="voicemail" data="default $$domain ${dialed_ext}"/>-->`
- `<anti-action application="export" data="sip_h_Diversion=${dialed_ext}"/>`
- `<anti-action application="bridge"`
- `data="{absolute_codec_string=PCMU}sofia/internal/`
- `{dialed_ext}@192.168.1.250;transport=tcp"/>`
- ...
- `</condition>`

## Freeswitch Features Live Demo & FAQ

- BLF
- Conference
- Intercom
- BLA
- MP3 Local Stream
- IVR
- Asterisk Interconection
- ASR(Directory)
- ASR (Pizza Demo)
- TTS in Conference
- Googletalk Conference
- Bridge into Conference
- Exchange 2007 UM integration
- Exchange 2007 UM TTS/ASR