mod_conference

About

mod_conference provides both inbound and outbound conference bridge service for FreeSWITCH™. It can process multiple bit rates, load various profiles that specify DTMF controls, play prompt sounds and tones, and many other functions. You can create as many conferences as you like, as long as there still are free system resources (i.e. memory, CPU power, network bandwidth) available.

Conferences created in the dialplan use profiles that are defined in conf/autoload_configs/conference.conf.xml, if you are using the standard configuration files.

Conference Configuration

Conference configuration settings are stored in `conf/autoload_configs/conference.conf.xml`

Community member Stanislav Sinyagin has posted a nice personal blog entry on how he configured his conference bridge.

The configuration has the following structure, each "`[... config here ...]`" should be replaced with specific configuration statements as detailed in the sections that follow.

conference.conf.xml

```
<configuration name="conference.conf" description="Standard Conference configuration">

  <advertise>
          [... config here ...]
  </advertise>

  <caller-controls>
    <group name="default">
          [... config here ...]
    </group>
  </caller-controls>

  <chat-permissions>
    <profile name="default">
          [... config here ...]
    </profile>
  </chat-permissions>

  <profiles>
    <profile name="default">
          [... config here ...]
    </profile>
  </profiles>


</configuration>
```

advertise

This block specifies whether empty conferences should be advertised in presence, allowing you to see status of empty conferences. Any endpoint that supports presence, such as mod_sofia, can subscribe to these presence notifications.

Example configuration

```
<advertise>
  <room name="888@$${subdomain}" status="FreeSWITCH"/>
</advertise>
```

The conference name (888 in this case) should be the profile name that you specified in <profiles> section, $${subdomain} will be replaced with the subdomain that you specified in freeswitch.xml.

"status" is advertised as whatever you pass to it (identifier) or "Available" if none is passed.

Example 'advertise' Event via mod_event_multicast

Advertise event

```
proto: conf
login: 888@example.com
from: 888@example.com
status: FreeSWITCH
rpid: idle
event_type: presence
Event-Name: PRESENCE_IN
Core-UUID: c76e2d7d-39d7-dc11-93bf-0090fb0792c6
FreeSWITCH-Hostname: example.com
FreeSWITCH-IPv4: 192.168.1.5
FreeSWITCH-IPv6: 127.0.0.1
Event-Date-Local: 2008-02-09 13:04:44
Event-Date-GMT: Sat, 09 Feb 2008 18:04:44 GMT
Event-Date-timestamp: 1202580284348009
Event-Calling-File: mod_conference.c
Event-Calling-Function: send_presence
Event-Calling-Line-Number: 5037
Multicast-Sender: example.com
```

caller-controls

Caller controls are used to modify the state of the conference, such as lowering the volume, mute a participant, and such. Below are the commands that can be assigned to digits and executed during a conference. The "moderator-controls" group provides additional controls for participants who enter the conference with the moderator flag set. See below.

Reserved Control Group Names

Name	Description
none	Use this name to prevent installing caller-controls for callers to a conference.
default	This group of settings will be assigned if no "caller-controls" is specified. You can also assign it explicitly. This group is defined in vanilla config, thus removing it from the configurations will make no caller controls at all.

Actions

Each of these actions can be configured in caller-controls to be invoked by a conference member who enters the conference with this caller-controls group. These are instances of the conference API.

Action	Description	Min. Version
mute	Toggle audio from this member into the conference	
mute on	Disable audio from this member into the conference	

mute off	Enable audio from this member into the conference	
deaf mute	Block audio from conference to this member as well as mute, in one action	
energy up	Increase minimum energy threshold by 1 unit above which sound will be admitted into conference (noise gate)	
energy equ	Reset minimum energy threshold to default	
energy dn	Decrease minimum energy threshold by 1 unit	
vol talk up	Increase member talk (mic) volume into conference by 1 unit	
vol talk zero	Reset talk volume to default setting	
vol talk dn	Decrease talk volume by 1 unit	
vol listen up	Increase member receive (earpiece) volume by 1 unit	
vol listen zero	Reset member receive volume to default setting	
vol listen dn	Decrease member receive volume by 1 unit	
hangup	Leave the conference	
event	Send the DTMF event via CUSTOM `conference::maintenance` subclass to the event system (even to event socket)	
lock	Toggle the conference lock state (no new members can enter when locked)	
transfer	Transfer member to a given extension in a dialplan context	
execute_application	Execute a dialplan application	
floor	Toggle yourself on and off of talking floor, as long as no one else has floor status.	
vid-floor	Video floor. If video floor is currently locked, it will revert to auto; if there is no current holder, you become video floor holder	1.6
vid-floor-force	Video floor. If video floor is currently locked, it will revert to auto, otherwise you become the locked video floor holder	1.6
vmute	Video mute. Toggle video from this member into the conference	1.6
vmute on	Disable video from this member into the conference	1.6
vmute off	Enable video from this member into the conference	1.6
vmute snap	Take a video snapshot for this user to be used when in vmute	1.6
vmute snapoff	Discard the vmute video snapshot	1.6

Default controls

```
<group name="default">
   <control action="vol talk dn"      digits="1"/>
   <control action="vol talk zero"    digits="2"/>
   <control action="vol talk up"      digits="3"/>
   <control action="vol listen dn"    digits="4"/>
   <control action="vol listen zero"  digits="5"/>
   <control action="vol listen up"    digits="6"/>
   <control action="energy dn"        digits="7"/>
   <control action="energy equ"       digits="8"/>
   <control action="energy up"        digits="9"/>
   <control action="mute"             digits="0"/>
   <control action="deaf mute"        digits="*"/>
   <control action="hangup"           digits="#"/>
</group>
```

Example Configuration

```
<caller-controls>
  <group name="somekeys">
    <control action="vol talk dn"         digits="1"/>
    <control action="vol talk zero"       digits="2"/>
    <control action="vol talk up"         digits="3"/>
    <control action="transfer"            digits="5"   data="100 XML default"/>
    <control action="execute_application" digits="0"   data="playback conf_help.wav"/>
    <control action="execute_application" digits="#"   data="execute_dialplan conference-menu"/>
  </group>
</caller-controls>
```

Limitations

Be aware that the caller-controls are applied across the entire conference session. You cannot apply one group of caller-controls to one member and then a second group of caller-controls to a second member in the same conference session.

chat-permissions

The FreeSWITCH conference also provides limited chat capabilities via SIP, XMPP, and Verto.

A conference can be controlled through chat. This profile exists as part of the conference entity so you can have multiple profiles to limit access. When the conference advertises its presence (above) to a chat server, anyone on a federated server can talk to it and issue commands by typing messages via their favorite chat client (even if that is another FreeSWITCH™ box). To send commands to ConferenceA, from your jabber client send a message to conf+ConferenceA@freeswitch.mydomain.com

As of revision 3789 all commands except "list" have been disabled "until there is security"

Another Note: After spending much time trying to get control of mod_conference via Openfire XMPP server, asked around, and "bougyman" suggested that XMPP muc functionality is very limited as of January 2012.

Here is an example configuration

chat-permissions example

```
<configuration name="conference.conf" description="Audio Conference">
 <profiles>
  <profile name="default">
   <param name="chat-permissions" value="default"/>
  </profile>
 </profiles>
 <chat-permissions>
  <profile name="default">
   <!-- both of these users have a functionally equivalent capability set -->
   <user name="bob@somewhere.com" commands="all"/>
   <!-- individually specified commands must be enclosed with the | (pipe) character -->
   <user name="harry@elsewhere.com" commands="
|deaf|dial|energy|kick|list|lock|mute|norecord|play|record|relate|say|saymember|stop|transfer|undeaf|unlock|unmu
te|volume_in|volume_out|"/>
  </profile>
 </chat-permissions>
</configuration>
```

profiles

You can specify a number of different profiles in the profiles section, these will let you easily apply a number of settings to a conference. Please note that the profiles are not conference rooms, but define settings that are later applied to conference rooms. The dialplan section in this document will describe how you create conference rooms and apply these profile settings.

profiles structure

```
<profiles>
  <profile name="default">
    <param name="paramName" value="paramValue"/>
  </profile>
</profiles>
```

You can have any number of <profile> tags, and each <profile> can have any number of <param> tags up to the entire set of parameters.

Conference Profile Parameters

You may specify the conference profile parameters listed below. Keep in mind that if TTS is enabled all audio-file params beginning with 'say:' will be considered text to say with TTS. A TTS module must be loaded by FreeSWITCH for this to work.

Name	Description	Example value	Allowed value	Default value	Played for	Min. Version
announce-count	Requires TTS. If the number of members equals or exceeds this value, the conference will speak the count to the conference after a new member joins	5	<integer>	0		
auto-gain-level	Enables Automatic Gain Control (AGC). If the parameter is set to 'true', then the default AGC value is used. If set to a number it will override the default value	900	<integer>|true	1100		
auto-record	Set a filename or stream value for this parameter in order to enable recording every conference call. Within mod_conference there is a special parameter named ${conference_name} that can be used to form the record filename. All channel variables are accessible as well for generating a unique filename. **NOTE: auto-record doesn't begin recording until the number of conferees specified in min-required-recording-participants have joined.**	**Example 1:** /var/myNFSshare/${conference_name}_${strftime(%Y-%m-%d-%H-%M-%S)}.wav **Example 2:** shout://user:pass@server.com/live.mp3 **Example 3:** /var/myNFSshare/${conference_name}_${strftime(%Y-%m-%d-%H-%M-%S)}.mp4 will give you audio and video recording.	<file-path>			
broadcast-chat-message	Message to send in response to chat messages.	Please go away.	<string>			
caller-controls	Name of the caller control group to use for this profile. It must be one of those specified in the <caller-controls> section	somekeys	<control-name>	default		
moderator-controls	Name of the moderator control group to use for this profile. It must be one of those specified in the <caller-controls> section	somekeys	<control-name>	default		
cdr-log-dir	Target directory for conference CDRs to be written. Use "auto" to store in $PREFIX/log/conference_cdr. An absolute path is acceptable as is a relative path. A relative path will yield $PREFIX/log/<value> for the conference CDR directory	auto	<path>|auto			
cdr-event-mode	Include full cdr or path to file in conference cdr event	content	content|file|none			
caller-id-name	Default Caller ID Name for outbound calls originated by mod_conference	FreeSWITCH	<string>	conference		
caller-id-number	Default Caller ID Number for outbound calls originated by mod_conference	8777423583	<string>	0000000000		
channels	The number of audio channels. Special value "auto" sets this based on the channels of the first member to enter	1	1|2|auto	1		1.4

comfort-noise	Sets the volume level of background white noise to add to the conference. Special value "true" sets to default value. 0 for no CN (total silence).	1400	0-10000|true	1400		
conference-flags	Can be any combination of allowed values separated by "|" (pipe character).	wait-mod|audio-always	**See Table Conference Flags below**			
description	Description of the conference that is included in some events	The main conference	<string>			
domain	The domain name used for presence events from conferences	$${domain}	<string>			
energy-level	Noise gate. Energy level required for audio to be sent to the other users. The energy level is a minimum threshold of 'voice energy' that must be present before audio is bridged into the conference. Useful if a participant is in a noisy environment, so their background noise is heard only when they speak. 0 disables the noise gate and will bridge all packets even if they are only background noise.	300	0-1800			
interval	Number of milliseconds per frame. Which may be different from ptime in SIP SDP, or driver with TDM. Higher numbers require less CPU but can degrade conversation quality, so experimentation with your setup is best. The default is good for conversation quality. Special "auto" value sets interval based on interval of the first member who enters.	20	10-120 (only if divisible by 10) or auto			
max-members	Sets a maximum number of participants in conferences with this setting in its profile.	20	Integer > 1			
member-flags	Can be any combination of allowed values separated by "|". See table below for descriptions.	nomoh|mute	**See Table Member Flags below**			
pin	Pin code that must be entered before user is allowed to enter the conference.	12345	<sritng>			
moderator-pin	Pin code that must be entered before moderator is allowed to enter the conference.	12345	<string>			
pin-retries	Max number of times the user or moderator will be asked for PIN	3	<integer> >= 0	3		
rate	Audio sample rate. Special value "auto" sets this based on the rate of the first member to enter.	8000	8000, 12000, 16000, 24000, 32000, 44100, 48000, auto			
sound-prefix	Set a default path here so you can use relative paths in the other sound params.	/soundfiles	<path-string>			
suppress-events	For use with the event socket. This parameter is a comma delimited string that specifies which events will NOT be sent to the socket when getting CUSTOM conference::maintenance events.	del-member,start-talking,stop-talking	**See Table Conference Events below**			
terminate-on-silence	Specifies the number of contiguous seconds of silence after which the conference will automatically terminate and disconnect all members.	300	<integer> > 0			
timer-name	Name of the timer interface in freeswitch to use for timing the conference	soft	<timer-name>	soft		
tts-engine	Text-To-Speech (TTS) Engine to use	cepstral	<tts-name>			
tts-voice	TTS Voice to use	david	<tts-voice>			
verbose-events	Events related to the conference will send more data. Specifically the events related to members will include all the channel variables on each event	true	true|false	false		
alone-sound	File to play if you are alone in the conference	yactopitc.wav	<sound-path>		User	
bad-pin-sound	File to play to when the pin is invalid	invalid-pin.wav	<sound-path>		User	
enter-sound	File to play when you join the conference	welcome.wav	<sound-path>		All	
exit-sound	File to play when you leave the conference	exit.wav	<sound-path>		All	
is-locked-sound	File to play when the conference is locked during the call to the members in the conference	is-locked.wav	<sound-path>		All	

is-unlocked-sound	File to play when the conference is unlocked during the call	is-unlocked.wav	<sound-path>		All	
join-only-sound	File to play when member with **join-only** flag tries to create the conference (join as the first).	no_reources_try_later.wav	<sound-path>		User	
kicked-sound	File to play when you are kicked from the conference	kicked.wav	<sound-path>		User	
locked-sound	File to play when the conference is locked and someone goes to join	locked.wav	<sound-path>		All	
max-members-sound	If **max-members** has been reached, this sound plays instead of allowing new users to the conference	max-members.wav	<sound-path>		Caller	
moh-sound	the given sound file/resource will be played only when the conference size is 1 member. It will loop over and over until the member count is 2 or more. When the conference goes back to 1 member it will play again	idlemusic.wav	<sound-path>		All	
muted-sound	File to play when member is muted	muted.wav	<sound-path>		User	
mute-detect-sound	If the **mute-detect** member-flag has been set, this sound plays when the user talks while muted	mute-detect.wav	<sound-path>		User	
perpetual-sound	The given sound file/resource will be played on a loop forever. This can be used to broadcast sales or emergency messages to callers.	announcement.wav	<sound-path>		All	
pin-sound	File to prompt the user to enter the conference pin code.	pin.wav	<sound-path>		User	
unmuted-sound	File to play when member is unmuted.	unmuted.wav	<sound-path>		User	
ivr-dtmf-timeout	Inter-digit timeout between DTMF digits in milliseconds	500	<integer> >= 500	500		
ivr-input-timeout	Time to wait for the first DTMF in milliseconds, zero = forever	0	<integer> >= 5000 or 0	0		
endconf-grace-time	Defines how much time all members have before the conference is terminated when the last member with **endconf** leaves in seconds	600	<integer> >= 0	0		
min-required-recording-participants	Minimum number of conference participants required for their audio to be heard in a recording and for auto recording to start. This can be either 1 (the default) or 2.	1	1-2	1		
outcall-templ	Template to use for outcall URL		<string>			
video-mode	The mode to run video conferencing in. passthrough is non transcoded video follow audio. transcode allows for better switching and multiple codecs. mux allows for multiple parties on the video canvas at the same time	mux	mux transcode passthrough	passthrough		1.6
video-layout-name	The layout name from conference_layouts.conf.xml or group prefixed by "group:". Setting this setting will enable the video mux. Not setting this will switch video presentation based on floor.	group:grid 2x2	<layout-name> or group:<group-name>			1.6
video-canvas-bgcolor	Overall Canvas color for video mux canvas as an HTML HEX color code	#333333	<color-spec>	#333333		1.6
video-letterbox-bgcolor	Color to use for bars on the caller video if their aspect ratio doesn't fit the member square in the layout as an HTML HEX color code	#FFFFFF	<color-spec>	#000000		1.6
video-canvas-size	Pixel dimensions of the video mux canvas	800x600	<integer>x<integer>			1.6
video-fps	Frames per second to run the video mux at	15	<integer>	15		1.6
video-codec-bandwidth	The bandwidth maximum for the conference output video stream, This setting can be auto, our defined in KB (kilobytes), mb (megabits) or MB (megabytes)	1MB	auto|<integer>KB|<integer>mb|<integer>MB			1.6
video-no-video-avatar	Path to PNG file for member without video to display.					1.6
video-canvas-count	Number of canvases to use in mux mode	1	1-5	1		1.6

video-quality	Motion factor of the video, this is used to adjust the video bitrate. From 1 to 4, low motion to high motion video. (ref. Kush Gauge formula for determining bitrate).	1	1-4	1		1.6
video-border-color	Border color around each video feed	#FFFFFF	<color-spec>	#000000		1.6
video-border-size	Border size (in pixels) around each video feed	5	<integer> 0-50	0		1.6
video-mute-banner	Sets the video mute banner text, font, font size, font color, and background color for the member. font_scale valid values 5-50, fg/bg hex color code, all settings besides text are optional.	{font_face=/path/to/font.ttf, font_scale=5, bg=#000000, fg=#FFFFFF} VIDEO MUTED" />				1.6
video-super-canvas-bgcolor	Background color of the supercanvas	#FFFFFF	<color-spec>	#068DF3		1.6
video-super-canvas-label-layers	Label canvases in supercanvas	true	Boolean	false		1.6
video-super-canvas-show-all-layers	Display all canvases, even empty ones	true	Boolean	false		1.6
video-auto-floor-msec	Milliseconds of speaking before a speaker gets the floor	800	Integer >= 0	0		1.6
video-kps-debounce	Milliseconds between sending packets to the client to request a client bitrate adjustment (video only). Note that FreeSWITCH may decide to force sending more frequently under certain circumstances.	30000	Integer >= 0	30000		1.6
auto-record-canvas-id	Which canvas on the supercanvas to auto record, (range is the)	2	Integer 1 - # of canvases	1		1.6
video-layout-conf	Specify an alternate conference layout config file.	custom_conference_layouts.conf		conference_layouts.conf		1.8

Table: Member Flags

These flags can be applied to individual conferees when entering the conference to control what they can do and how their media is processed.

Member Flag	Description	Min. Version
mute	Enter conference muted	
deaf	Enter conference deafed (can not hear conference); will also mute the mic	
mute-detect	Play the mute_detect_sound when talking detected by this conferee while muted	
dist-dtmf	Send any DTMF from this member to all participants	
moderator	Flag member as a moderator	
nomoh	Disable music on hold when this member is the only member in the conference	
endconf	Ends conference when all members with this flag leave the conference after profile param endconf-grace-time has expired	
mintwo	End conference when it drops below 2 participants after a member enters with this flag	
ghost	Do not count member in conference tally	
join-only	Only allow joining a conference that already exists	
positional	Process this member for positional audio on stereo outputs	1.4
no-positional	Do not process this member for positional audio on stereo outputs	1.4
join-vid-floor	Locks member as the video floor holder	1.6
no-minimize-encoding	Bypass the video transcode minimizer and encode the video individually for this member	1.6
vmute	Enter conference video muted	1.6
second-screen	Open a 'view only' connection to the conference, without impacting the conference count or data.	1.6

Table: Conference Flags

These flags control the operation of the conference session and apply to all members.

Conference Flag	Description	Min. Version
wait-mod	Members will wait (with music) until a member with the 'moderator' flag set enters the conference	1.0
audio-always	Do not use energy detection to choose which participants to mix; instead always mix audio from all members	1.2
restart-auto-record	If the auto-record conference param is set, and recording is stopped, auto recording will continue in a new file	1.2.16
rfc-4579	Enable support for RFC-4579 SIP–based call control for conferences	1.4
livearray-sync	Add support for livearray advertisement for the conference status	1.4
auto-3d-position	Enable 3d positioned audio support	1.4
json-events	Send event-channel JSON events when members enter and leave	1.4
video-floor-only	Only video members can get floor. REMOVED in 1.6	1.4
minimize-video-encoding	Use a single video encoder per output codec	1.6
livearray-json-status	Machine parseable version of display for conference live array. Example: {"audio":{"muted":false,"onHold":false,"talking":true,"floor":true,"energyScore":639}, "video":{"avatarPresented":false,"mediaFlow":"sendRecv","muted":false,"floor":true,"reservationID":null," videoLayerID":0}}	1.6
video-bridge-first-two	In mux mode, If there are only 2 people in conference, you will see only the other member	1.6
video-muxing-personal-canvas	In mux mode, each member will get their own canvas and they will not see themselves	1.6
video-required-for-canvas	Only video participants will be shown on the canvas (no avatars)	1.6
video-mute-exit-canvas	Don't display video muted users on the canvas	1.6
manage-inbound-video-bitrate	If calling client supports TMMBR, on each change of layout position FreeSWITCH will instruct the client to increase/reduce the video bit rate appropriately.	1.6

Table: Conference Events

These events are available to event consumers.

Event Name	Description
add-member	Member added to a conference
del-member	Member removed from a conference
energy-level	Conference default energy level changed
volume-level	Conference default volume level changed
gain-level	Conference default gain level changed
dtmf	Key bound to transfer, event, or execute_application is hit by member
stop-talking	Member stopped talking (as detected by energy level)
start-talking	Member started talking (as detected by energy level)
mute-detect	Detected member speaking (as detected by energy level) while muted
mute-member	Member became muted
unmute-member	Member became unmuted
kick-member	Member kicked
dtmf-member	Key bound to DTMF is hit by member
energy-level-member	Member energy level changed
volume-in-member	Member gain level changed
volume-out-member	Member volume level changed
play-file	Conference–level play file started
play-file-done	Conference–level play file ended
play-file-member	Member–level play file started
lock	Conference locked, no one else can enter
unlock	Conference unlocked
transfer	Member transferred to another conference
bgdial-result	Result from bgdial API command
floor-change	Conference floor changed
record	Conference recording started or stopped

Conference Dialplan Application

The conference dialplan application is used to create conferences and to bind a profile to them.

Syntax

```
<action application="conference" data="confname[@profile][+[pin][+flags{mute|deaf|...}]">
<action application="conference" data="bridge:confname[@profile]:none|endpoint[+flags{mute|deaf|...}]">
```

The first time a conference name (confname) is used, it will be created on demand, and the pin will be set to what ever is specified at that time: the pin in the data string if specified, or if not, the "pin" setting in the conference profile, and if that is also unspecified, then there is no pin protection. Any later attempt to join the conference must specify the same pin number, if one existed when it was created.

"profile" should be one of those specified in the conference configuration, or "default".

If the data value starts with "`bridge:`" then it is a bridging conference. The conference name should not be already in use. You can specify the special literal value of "_uuid_" for the conference name, and a a session-specific unique id will be generated for you.

Conferences stay alive until the number of members falls below the minimum. The minimum for bridging conferences is 2, and for other dynamically created conferences is 1.

Dialplan Examples

Action data	Description
confname	profile is "default", no flags or pin
confname+1234	profile is "default", pin is 1234
confname@profilename+1234	profile is "profilename", pin=1234, no flags
confname+1234+flags{mute}	profile is "default", pin=1234, one flag
confname++flags{endconf|moderator}	profile is "default", no p.i.n., multiple flags
bridge:confname:1000@${domain_name}	a "bridging" conference, you must provide another endpoint, or 'none'.
bridge:_uuid_:none	a "bridging" conference with UUID assigned as conference name
*Note that while some parameters are optional, their order is very important	

Simple Dialplan Example

```
<action application="conference" data="meeting@mykeys">
```

Bridging conference example that plays ringback while other party is bridged in

```
<extension name="test_bridging_conference">
   <condition field="destination_number" expression="^(3000)$">
      <action application="answer"/>
      <action application="playback" data="connecting_your_call.wav"/>
      <action application="set" data="ringback=${us-ring}"/>
      <action application="conference" data="bridge:$1-${domain_name}@default:user/1000@${domain_name}"/>
   </condition>
</extension>
```

Adding Callers From Within The Conference With DTMF

By combining several elements — the dialplan, API calls, bind_digit_action — you can create a simple system for a caller to add another user to the conference. See Conference_Add_Call_Example for the code and explanation.

Announcing Caller Count While In Conference

See Conference Announce Count Inline for a simple example of how to allow a caller to hear an announcement of how many members are in the conference.

Propagate Out-of-Band DTMF to Conference Members

By default, out-of-band DTMF signals (RFC 2833?) are absorbed by the conference. However, there are two ways to accomplish this:

- Set the dist-dtmf member flag in the conference configuration XML, eg: <param name="member-flags" value="dist-dtmf"/> With this parameter set, all of the caller controls such as modifying volume will be disabled and DTMF will simply pass through to all other members of the conference.
- There is also an API call that will allow your application to send DTMF to a single conference member or all members:

```
<confname> dtmf-string <[member_id|all|last]> <digits>
```

Prompt for Name

Dialplan to prompt callers to speak their names before joining the conference and announce their name to other participants.

Speak name

```
  <extension name="Record Name and schedule conf announce"
>
   <condition field="destination_number" expression="^55
(3\d\d\d)$">
     <action application="answer"
/>

     <action application="set" data="namefile=/tmp/${uuid}-name.wav" inline="true"
/>
     <action application="sleep" data="1000"
/>>

     <action application="playback" data="voicemail/vm-record_name1.wav"
/>
     <action application="playback" data="tone_stream://%(1000,0,500)"
/>
     <action application="record" data="${namefile} 1"
/>
     <action application="playback" data="ivr/ivr-call_being_transferred.wav"
/>
     <action application="set" data="res=${sched_api +1 none conference $1-${domain} play
file_string://${namefile}!conference/conf-has_joined.wav}"/>
     <action application="transfer" data="$1 XML default"
/>
   <
/condition>

 </extension>
```

API Reference

Command	Description	Syntax	Examples	Min. Version
agc	Adjust conference automatic gain control.	conference <confname> agc [on [<level>] | off]	Query level: conference testconf agc Disable: conference testconf agc off Enable: conference testconf agc on Enable w/ level: conference testconf agc on 1120	
bgdial	Background dial a destination via a specific endpoint	conference <confname> bgdial <dial-string> [<callerid_number> [<callerid_name>]]		
chkrecord	Query record status of conference	conference <confname> chkrecord		
deaf	Make a conference member deaf	conference <confname> deaf <member_id>|all|last|non_moderator		
dial	Dial a destination	conference <confname> dial <dial-string> [<callerid_number> [<callerid_name>]]		
dtmf	Send DTMF to any member of the conference	conference <confname> dtmf <member_id>|all|last|non_moderator <digits>	conference testconf dtmf all 134	
energy	Adjusts the conference energy level for a specific member	conference <confname> energy <member_id>|all|last|non_moderator <newval>		
enter_sound	Changes the sound played while entering the conference	conference <confname> enter_sound on|off|none|file <filename>		
exit_sound	Changes the sound played while leaving the conference	conference <confname> exit_sound on|off|none|file <filename>		

<table>
<tr><td>floor</td><td>Toggle floor status of the member.</td><td>conference <confname> <member_id>|all|last|non_moderator</td><td></td><td></td></tr>
<tr><td>file_seek</td><td>Seek to a point in the currently playing file, in milliseconds</td><td>conference <confname> file_seek [+|-]<val></td><td>Move forward 1 second: conference testconf file_seek +1000

Move backwards 1 second: conference testconf file_seek -1000

Move to 10 seconds from the start: conference testconf file_seek 10000</td><td></td></tr>
<tr><td>file-vol</td><td>Changes the volume of the currently playing sound file</td><td>conference <confname> file-vol <val> [async]</td><td></td><td></td></tr>
<tr><td>get</td><td>Get runtime parameter of the given conference

(table of allowed parameters below)</td><td>conference <confname> get <parameter_name></td><td></td><td></td></tr>
<tr><td>hup</td><td>Kick without the kick sound</td><td>conference <confname> hup <member_id>|all|last|non_moderator</td><td></td><td></td></tr>
<tr><td>kick</td><td>Kicks the member or all members from the conference</td><td>conference <confname> kick <member_id>|all|last|non_moderator</td><td></td><td></td></tr>
<tr><td>list</td><td>Lists conferences</td><td>conference list [pretty|summary|count|delim <string>]</td><td></td><td></td></tr>
<tr><td>lock</td><td>Lock a conference so no new members will be allowed to enter</td><td>conference <confname> lock</td><td></td><td></td></tr>
<tr><td>mute</td><td>Mutes a conference member</td><td>conference <confname> mute <member_id>|all|last|non_moderator [quiet]</td><td></td><td></td></tr>
<tr><td>nopin</td><td>Removes a pin from a conference</td><td>conference <confname> nopin</td><td></td><td></td></tr>
<tr><td>norecord</td><td>Removes a recording or all recordings from a conference</td><td>conference <confname> norecord <file-path>|all</td><td>conference testconf norecord all

conference testconf norecord /tmp/foo.wav</td><td></td></tr>
<tr><td>pause</td><td>Pause a conference recording</td><td>conference <confname> pause <file-path></td><td></td><td></td></tr>
<tr><td>pause_play</td><td>Pause playback of file to a conference</td><td>conference <confname> pause_play</td><td></td><td></td></tr>
<tr><td>pin</td><td>Set or change a pin number for a conference</td><td>conference <confname> pin <pin-number></td><td>Note: Members joined from conference dial command

do not get prompted for a pin</td><td></td></tr>
<tr><td>play</td><td>Play an audio file to a conference or a specific member</td><td>conference <confname> play [{vol=<volume>,full-screen=true,png_ms=100}]<file-path> [async|<member_id> [nomux]]</td><td>full-screen param will play the video in full screen mode in the conference,
png_ms depends on mod_png to be loaded, Specify a PNG file to play and how many milliseconds, -1 for indefinite</td><td></td></tr>
<tr><td>record</td><td>Record a conference to a file or stream</td><td>conference <confname> record <file-path></td><td></td><td></td></tr>
<tr><td>recording</td><td>Start or control a conference recording</td><td>conference <confname> recording start <file-path>

conference <confname> recording check

conference <confname> recording stop <file-path>|all

conference <confname> recording pause <file-path>

conference <confname> recording resume <file-path></td><td></td><td></td></tr>
<tr><td>relate</td><td>Mute or Deaf a specific member to another member

Control who sees who's video when not using video mux</td><td>conference <confname> relate <member_id>[,<member_id>] <other_member_id>[,<other_member_id>] [nospeak|nohear|clear|sendvideo]</td><td>See expanded entry below.</td><td></td></tr>
<tr><td>resume</td><td>Resume recording</td><td>conference <confname> resume <file-path></td><td></td><td></td></tr>
<tr><td>say</td><td>Speak text into conference using TTS</td><td>conference <confname> say <text></td><td></td><td></td></tr>
<tr><td>saymember</td><td>Speak text to a member of a conference using TTS</td><td>conference <confname> saymember <member_id> <text></td><td></td><td></td></tr>
<tr><td>set</td><td>Set runtime parameter of a conference</td><td>conference <confname> set <parameter_name> <value></td><td></td><td></td></tr>
<tr><td>stop</td><td>Stops any queued audio from playing into conference</td><td>conference <confname> [current|all] [<member_id>]</td><td></td><td></td></tr>
</table>

<table>
<tr><td>tmute</td><td>Toggle mute on/off for a member of a conference</td><td>conference <confname> tmute <member_id>|all|last|non_moderator [quiet]</td><td></td><td></td></tr>
<tr><td>transfer</td><td>Transfer a member from one conference to another conference</td><td>conference <confname> transfer <conference_name> <member_id></td><td></td><td></td></tr>
<tr><td>undeaf</td><td>Allow conference member to hear again</td><td>conference <confname> undeaf <member_id>|all|last|non_moderator</td><td></td><td></td></tr>
<tr><td>unlock</td><td>Unlock a conference so that new members can enter</td><td>conference <confname> unlock</td><td></td><td></td></tr>
<tr><td>unmute</td><td>Unmute a conference member</td><td>conference <confname> unmute <member_id>|all|last|non_moderator [quiet]</td><td></td><td></td></tr>
<tr><td>volume_in</td><td>Adjust the gain on the audio coming from a member</td><td>conference <confname> volume_in <member_id>|all|last|non_moderator [<newval>]</td><td></td><td></td></tr>
<tr><td>volume_out</td><td>Adjust the volume of audio going to a member</td><td>conference <confname> volume_out <member_id>|all|last|non_moderator [<newval>]</td><td></td><td></td></tr>
<tr><td>xml_list</td><td>List all or specific conferences in XML format</td><td>conference xml_list

conference <confname> xml_list</td><td></td><td></td></tr>
<tr><td>vid-floor</td><td>Set or force the video floor for a member</td><td>conference <confname> vid-floor <member_id>|last [force]</td><td></td><td>1.6</td></tr>
<tr><td>vmute</td><td>Enable video mute for a member</td><td>conference <confname> vmute <member_id>|all|last|non_moderator [quiet]</td><td></td><td>1.6</td></tr>
<tr><td>tvmute</td><td>Toggle video mute on/off for a member</td><td>conference <confname> tvmute <member_id>|all|last|non_moderator [quiet]</td><td></td><td>1.6</td></tr>
<tr><td>vmute-snap</td><td>Take or clear a snapshot image of the member to be used during vmute</td><td>conference <confname> vmute-snap <member_id>|all|last|non_moderator [clear]</td><td></td><td>1.6</td></tr>
<tr><td>unvmute</td><td>Unmute video for a member</td><td>conference <confname> unvmute <member_id>|all|last|non_moderator [quiet]</td><td></td><td>1.6</td></tr>
<tr><td>vid-banner</td><td>Set a text banner on the members video</td><td>conference <confname> vid-banner <member_id>|all|last|non_moderator <text></td><td></td><td>1.6</td></tr>
<tr><td>vid-mute-img</td><td>Check, set or clear the file to use as the members video mute image</td><td>conference <confname> vid-mute-img <member_id>|all|last|non_moderator [<path>|clear]</td><td></td><td>1.6</td></tr>
<tr><td>vid-logo-img</td><td>Check, set or clear the file to use as the members video logo image</td><td>conference <confname> vid-logo-img <member_id>|all|last|non_moderator [<path>|clear]</td><td>conference foo vid-logo-img 1 {options}/path/to/png

conference foo vid-logo-img 1 clear

*see settable channel var video_logo_path below for all the options you can use with this vid-logo-img api</td><td>1.6</td></tr>
<tr><td>vid-res-id</td><td>Set or clear the video reservation id for this member</td><td>conference <confname> vid-res-id <member_id>|all <val>|clear [force]</td><td>conference foo vid-res-id 1 res-bar

conference foo vid-res-id 2 res-baz force

conference foo vid-res-id all clear</td><td>1.6</td></tr>
<tr><td>clear-vid-floor</td><td>Clear when the video floor is locked to a user</td><td>conference <confname> clear-vid-floor</td><td></td><td>1.6</td></tr>
<tr><td>vid-layout</td><td>List available video mux layouts or set the layout or group to use in the video mux</td><td>conference <confname> vid-layout list|<layout-name>|group <group_name> [<canvas_id>]</td><td></td><td>1.6</td></tr>
<tr><td>vid-fps</td><td>Set the frames per second rate of the video mux output for the conference</td><td>conference <confname> vid-fps <fps></td><td></td><td>1.6</td></tr>
<tr><td>vid-bandwidth</td><td>Set the bandwidth used by the output video from the conference</td><td>conference <confname> vid-bandwidth <BW></td><td></td><td>1.6</td></tr>
<tr><td>vid-write-png</td><td>Take a snapshot of the current video mux output and save it to a PNG file</td><td>conference <confname> vid-write-png <path></td><td></td><td>1.6</td></tr>
<tr><td>vid-canvas</td><td>Set the canvas to display this member on</td><td>conference <confname> vid-canvas <member_id>|all|last|non_moderator [<canvas_id>]</td><td></td><td>1.6</td></tr>
<tr><td>vid-watching-canvas</td><td>Set the canvas displayed to this member</td><td>conference <confname> vid-watching-canvas <member_id>|all|last|non_moderator [<canvas_id>]</td><td></td><td>1.6</td></tr>
</table>

API Additional Notes

dial and bgdial

If the caller id values are not set, the variables in conference.conf.xml will be used. Specifically, the value for caller-id-number will be used for the number and the value for caller-id-name will be used for the name.

If the conference will be dynamically created as a result of this api call (ie this will be the first participant in the conference) - and the caller id name and number is not provided in the api call - the number and name will be "00000000" and "FreeSWITCH". This appears to be unaffected by the variables in conference.conf.xml.

get and set

Parameter	Description	Units	Writable
run_time		Seconds	
count		Number	
max_members		Number	Yes
rate		Number	
profile_name		String	
sound_prefix		String	Yes
caller_id_name		String	Yes
caller_id_number		String	Yes
is_locked		"locked" or blank	
endconference_grace_time		Seconds	Yes
uuid		String	
wait_mod		"true" or blank	

Dialplan usage example

```
<action application="set" data="conf_runtime=${conference(${conference_name} get run_time)}"/>
<action application="set" data="conf_sounddir=${conference(${conference_name} get sound_prefix)}"/>
<action application="set" data="conf_oldsound=${conference(${conference_name} set sound_prefix
${sound_prefix_pl})}"/>
<action application="set" data="void_result=${conference(${conference_name} set endconference_grace_time 300)}"
/>
```

list

Command Output

First line:

- Conference <conference name> (<member_count> member[s][locked])
    - “locked” - The lock/unlock status of the conference.

Each following line is a comma-separated list for each conference leg with the following items:

- id of participant(starts at 1 after FS restart)
- Register string of participant
- UUID of participant's call leg
- Caller id name
- Caller id number
- Status (hear|speak|talking|video|floor)
    - “hear” - The mute/unmute status of the member.
    - “speak” - The “deaf /undeaf” status of the member.
    - “talking” - The input channel is currently providing some amount of sound energy into conference.
    - “video” - Providing video?
    - “floor” - This member currently owns the floor.
- Volume In setting
- Detected energy
- Volume Out setting
- Energy Level setting

A handy way to test in an XML dialplan if a conference is active and allow a late caller to join

```
<extension name="late entry">
   <condition field="destination_number" expression="^(300\d)$" />
   <condition field="${conference $1 list count}" expression="^\d+">
      <action application="conference" data="$1@default++flags{nomoh}"/>
   </condition>
</extension>
```

If the conference is not active, the second "condition" test will fail and bypass this dialplan extension entry. If the conference is active the caller will join it with the 'nomoh' flag set.

relate

```
conference <conf-name> relate <datum-member1>[,<datum-member2>] <related-member1>[,<related-member2>]
<relationship>
```

<datum> is the conference member that is the subject of the relation; in other words, the relationship will be established with respect to <datum>

<related-member> is the conference member that will be affected by the relate command

<relationship> can be one of

- nohear — the datum member will no longer hear the related-member; i.e. datum's ear will be muted to related-member's mouth
- nospeak — the datum member will no longer be able to speak to the related-member
- clear — reset the relationship between the two conference members

Note that "relate 1 2 nohear" gives the same result as "relate 2 1 nospeak", it is simply a matter of perspective.

Examples

```
conference 3000 relate 1 2 nospeak
```

Member 1 may now no longer speak to member 2, i.e. member 2 now cannot hear member 1.

```
conference 3000 relate 1 2 clear
```

Member 1 may now speak to member 2 again

```
conference 3100 relate 1 2 nohear
```

Member 1 now cannot hear member 2

```
conference 3100 relate 1 2 clear
```

Member 1 can now hear member 2 again

Both conference-member arguments can take multiple values, separated by commas (but no spaces!)

```
conference 3300 relate 1,2 3 nospeak
```

Member 1 and member 2 may now no longer speak to member 3.

```
conference 3001 relate 1 2,3 nohear
```

Member 1 now cannot hear member 2 nor member 3

Event Socket Use

You can subscribe to the following to receive conference events:

```
conference::maintenance
```

The "suppress-events" parameter can be added to the conference profile to prevent events from firing. e.g. if you're not interested in start or stop talking events:

```
<profile name="default">
  ...other options...
  <param name="suppress-events" value="start-talking,stop-talking"/>
</profile>
```

The events that can be suppressed are:

del-member, energy-level, volume-level, gain-level, dtmf, stop-talking, start-talking, mute-member, unmute-member, kick-member, dtmf-member, energy-level-member, volume-in-member, volume-out-member, play-file, play-file-member, speak-text, speak-text-member, lock, unlock, transfer, bgdial-result and floor-change.

Outbound Conference

Use **conference_set_auto_outcall** to have **mod_conference** call endpoints and join them to a conference bridge. To have it call more than one participant, just repeat the **conference_set_auto_outcall** action in the dialplan for each destination number or address.

Syntax

```
<action application="conference_set_auto_outcall" data="dialstring"/>
```

Example

Here is an example of using **conference_set_auto_outcall** with some of the other conference_auto_outcall_* parameters to start a conference when someone dials **12345**. Extensions 1000 and 1001 will be dialed when the conference starts and will enter the conference muted.

Outbound example

```
<extension name="Demonstrate conference_set_auto_outcall">
  <condition field="destination_number" expression="^(12345)$">

    <action application="answer"/>

    <action application="set" data="conference_auto_outcall_timeout=5"/>
    <action application="set" data="conference_utils_auto_outcall_flags=mute"/>
    <action application="set" data="conference_auto_outcall_caller_id_name=$${effective_caller_id_name}"/>
    <action application="set" data="conference_auto_outcall_caller_id_number=$${effective_caller_id_number}"/>
    <action application="set" data="conference_auto_outcall_profile=default"/>
        <action application="set" data="conference_auto_outcall_prefix={sip_auto_answer=true,
execute_on_answer='bind_meta_app 2 a s1 transfer::intercept:${uuid} inline'}"/>
        <action application="set" data="conference_auto_outcall_timeout=60"/>

    <action application="conference_set_auto_outcall" data="user/1000@$${domain}"/>
    <action application="conference_set_auto_outcall" data="user/1001@$${domain}"/>
    <action application="conference_set_auto_outcall" data="sofia/internal/gateway/signalwire/12024561000"/>

    <action application="conference" data="$1@default"/>

  </condition>
</extension>
```

Note that the internal extensions 1000 and 1001 will provide no ringback to avoid polluting the conference bridge with excessive noise, but the gateway is not under control of FreeSWITCH so it will provide early media.

Alternatively, you can set multiple destinations in one line, just remember to escape your variables if you have more than one or any non-escaped chars in it.

```
<action application="conference_set_auto_outcall" data="['var1=a,var2=b']user/1001@$${domain},['var1=c,var2=d']
user/1002@$${domain}"/>
```

Variable	Description
conference_auto_outcall_announce	File name of audio message to play to conference member joining conference via the conference_set_auto_outcall application.

conference_auto_outcall_caller_id_name	Caller ID name to use when dialing endpoints using conference_set_auto_outcall application
conference_auto_outcall_caller_id_number	Caller ID number to use when dialing endpoints using conference_set_auto_outcall application
conference_auto_outcall_delimiter	Delimiter between values held in cpstr, default = "," (the comma character)
conference_utils_auto_outcall_flags	Conference member flags to set for members joining via conference_set_auto_outcall application (use | as separator for multiple flags)
conference_auto_outcall_maxwait	Maximum time in seconds that the channel that initiated the conference_set_auto_outcall will wait for members to join the conference
conference_auto_outcall_prefix	String to prepend to each of the dial-string values set from the conference_auto_outcall application
conference_auto_outcall_profile	Conference profile to use for members joining the conference via the conference_set_auto_outcall application
conference_auto_outcall_timeout	Originate timeout to use when joining a member to a conference via conference_set_auto_outcall application
conference_auto_outcall_skip_member_beep	When true, do not play a beep to moderator when initiating an outbound conference call. Tone is 640Hz for 500msec

Channel Variables

Read-Only

These channel variables are set by mod_conference.

Variable	Description
conference_last_matching_digits	Contains the last matching digits that the member sent into the conference
conference_member_id	Contains the member_id of the member for the conference they are in
conference_moderator	Contains "true" if the channel is connected to a conference as a moderator
conference_name	The name of the last conference joined by this channel
conference_recording	Contains the file name of the conference recording for the conference the channel is connected to
conference_uuid	Unique id of the most recent conference in which the channel was a member
last_transferred_conference	The name of the last conference this channel was connected to

Settable Channel Variables

Set these channel variables to control the behavior of mod_conference for the current session. conference_auto_outcall_ variables are described above in the Outbound Conference section

Variable	Description	Example	Min. Version
`conference_controls`	Specify which conference control set to use when transferring a caller into a conference	`<action application="set" data="conference_controls=moderator"/>`	

`conference_enforce_security`	Allows the conference security to be overridden. This applies in two different scenarios, one for inbound and one for outbound.
	
	By default, conference security is always applied to inbound calls and is always skipped for outbound calls.
		Inbound
		
		`<action application="set" data="conference_enforce_security=false"/>`
		`<action application="conference" data="3000"/>`
		
		Outbound
		
		`originate {conference_enforce_security=true}sofia/internal/1001 &conference(3000)`	

`conference_enter_sound`	Overrides the conference parameter for enter sound	`<action application="set" data="conference_enter_sound=silence_stream://10"/>`	
`conference_exit_sound`	Overrides the conference parameter for exit sound	`<action application="set" data="conference_exit_sound=silence_stream://10"/>`	
`conference_flags`	A comma separated list of conference flags, which will be applied when the conference is created. This allows to dynamically set the conference flags from the dial plan as opposed to setting them in the conference profile.	`<action application="set" data="conference_flags=wait-mod,audio-always"/>`	
`conference_member_flags`	A comma separated list of member flags to be set on the channel, which is joining the conference.	`<action application="set" data="conference_member_flags=mute"/>`	
`conference_permanent_wait_mod_moh`	This variable controls how the enter and exit sounds interact with the MOH when the `wait_mod` flag is set. When this variable is set to true, the MOH keeps playing and the enter and exit sounds are mixed with the MOH. When the variable is false or not set, then any playing MOH is first stopped, then the enter or exit sound is played and the MOH is started again. This functionality is useful in case the enter and exit sounds are used to announce the name of the caller, who is joining or leaving a conference.		
`video_banner_text`	Sets the video banner text, font, font size, font color, background color, min/max font size for the member (must be set before entering conference). font_scale valid values float between 5-50, fg/bg hex color code, min /max font size 5-24, all settings besides text are optional.	`<action application="set" data="video_banner_text={font_face=/path/to/font.ttf, font_scale=5,bg=#000000,fg=#FFFFFF,min_font_size=8,max_font_size=14}Banner Text"/>`	1.6
video_mute_banner	Sets the video mute banner text, font, font size, font color, background color, min /max font size for the member (must be set before entering conference). font_scale valid values float between 5-50, fg/bg hex color code, min/max font size 5-24, all settings besides text are optional.	`<action application="set" data="video_mute_banner={font_face=/path/to/font.ttf, font_scale=5,bg=#000000,fg=#FFFFFF,min_font_size=8,max_font_size=14}VIDEO MUTED"/>`	1.6
`video_avatar_png`	Path to PNG file to use when an avatar image is needed.	`<action application="set" data="video_avatar_png=/path/to/file.png"/>`	1.6
`video_mute_png`	Path to PNG file to use when you video mute, by default it will use your last video frame (overrides `video_avatar_png`).	`<action application="set" data="video_mute_png=/path/to/file.png"/>`	1.6
`video_no_video_avatar_png`	Path to PNG file for member without video to display (overrides `video_avatar_png`).	`<action application="set" data="video_no_video_avatar_png=/path/to/file.png"/>`	1.6
`video_avatar_png`	set video replacement avatar images	`<action application="set" data="video_avatar_png=/path/to/file.png"/>`	1.6

<table>
<tr><td>video_initial_canvas</td><td>Set to the initial canvas member should be displayed on</td><td><action application="set" data="video_initial_canvas=2"/></td><td>1.6</td></tr>
<tr><td>video_initial_watching_canvas</td><td>Set to the initial canvas member should see</td><td><action application="set" data="video_initial_watching_canvas=2"/></td><td>1.6</td></tr>
<tr><td>video_use_dedicated_encoder</td><td>If set to true, it allows the channel to have a dedicated encoder with its own bandwidth settings</td><td><action application="set" data="video_use_dedicated_encoder=true"/></td><td>1.6</td></tr>
<tr><td>conference_auto_record</td><td>Enable recording for this conference.</td><td><action application="set" data="conference_auto_record=/tmp/${strftime(%Y-%m-%d %H:%M)}.wav"/></td><td>1.6</td></tr>
<tr><td>video_logo_path</td><td>set an image logo, and optional text overlay</td><td><action application="set" data="video_logo_path={position=left-bot,text_x=center,center_offset=190, text=#000000:transparent:/var/ccw/font/AEH.ttf:50:'${toupper ${caller_id_name}}',alt_text_x=center, alt_center_offset=190,alt_text_y=88,alt_text=#ffffff:transparent:/path/to/font/AEH.ttf:40:'${caller_id_number}'} /var/png/freeswitch.png"/>

Parameters separated by a comma (,) are as follows:

position=left-top | left-mid | left-bot | center-top | center-mid | center-bot | right-top | right-mid | right-bot
fit=fit-size | fit-scale | fit-size-and-scale | fit-necessary
center_offset=[0 or higher]
text=foreground:background:font_face:font_size:txt
text_x=[0 or higher | center]
text_y=[0 or higher]
alt_text=foreground:background:font_face:font_size:txt
alt_center_offset=[0 or higher]
alt_text_x=[0 or higher | center]
alt_text_y=[0 or higher]

text= and alt_text= arguments separated by colon (:) are as follows:

foreground a hex notation color code i.e. #FF0000 for red
background a hex notation color code or can be the word transparent
font_face a path to your font face i.e. /usr/share/font/monospace.ttf
font_size a decimal value, or percentage value i.e. 10 or 10%
txt the literal text to display over the image logo

You could potentially use your name in text=x:x:x:x:${caller_id_name} and email, phone, or whatever in <b>alt_text=x:x:x:x:${caller_id_number}
and give alt_text_y slightly different y value so they're not on top of each other...

Note:

These options prepend the path of the file within { } curly braces, also notice video_logo_path and vid-logo-img achieve the same thing

when using these options in dialplan, utilize video_logo_path like this

```
<action application="set" data="video_logo_path={OPTIONS}/path/to/my.png"/>
```

when using these options in fs_cli or esl, utilize vid-logo-img like this

```
conference CONF vid-logo-img 1 {OPTIONS}/path/to/my.png
```
</td><td></td></tr>
</table>

Sound files

Just about any format is supported, but currently it must be at the sample rate of the conference (no resampling is done). Since disk is cheaper than CPU, use a wav.

Custom Conference Layouts

Layout grid is square, its directly proportional to the canvas resolution and aspect ratio. These layouts will work on 4:3 or 16:9 aspect ratio. The values x, y, and scale are all of a range of 0-360, proportional to the actual final resolution.

Layout Example

```
<configuration name="conference_layouts.conf" description="Audio Conference">
        <layout-
settings>


<layouts>

                        <layout name="presenter-overlap-small-top-right">
                                <image x="0" y="0" scale="360" floor-only="true"/>
                                <image x="300" y="0" scale="60" overlap="true" reservation_id="presenter"/>
                        </layout>

                        <layout name="1up_top_left+5">
                                <image x="0" y="0" scale="240" floor="true" audio-position="-.5:0:0"/>
                                <image x="240" y="0" scale="120" audio-position=".25:0:0"/>
                                <image x="240" y="120" scale="120" audio-position=".25:0:-.25"/>
                                <image x="0" y="240" scale="120" audio-position="-.5:0:-1"/>
                                <image x="120" y="240" scale="120" audio-position="0:0:-1"/>
                                <image x="240" y="240" scale="120" audio-position=".25:0:-1"/>
                        </layout>
                        <layout name="1x1">
                                <image x="0" y="0" scale="360" floor="true"/>
                        </layout>
                        <layout name="2x2">
                                <image x="0" y="0" scale="180"/>
                                <image x="180" y="0" scale="180"/>
                                <image x="0" y="180" scale="180"/>
                                <image x="180" y="180" scale="180"/>
                        </layout>
                </layouts>
                <groups>
                        <group name="grid">
                                <layout>1x1</layout>
                                <layout>2x2</layout>
                        </group>
                </groups>
        </layout-settings>
<configuration>
```

Layout Param	Description
border	Border size around each video feed (in pixels), values from 0 - 50. Overrides video-border-size conference parameter
bgimg	Full path to PNG file to use as a background image for the entire canvas

Layout Image Param	Description
x	Value from 0-360, proportional to the canvas size, amount out of 360 for this images top left corner x position
y	Value from 0-360, proportional to the canvas size amount out of 360 for this images top left corner y position
scale	Value from 0-360, proportional to the canvas size amount out of 360 for this images size
floor	with floor="true", this box will prefer the video floor holder and will switch as floor holder changes

reservation_id	arbitrary named setting used to specify which location to put someone using the vid-res-id api command
overlap	with overlap="true", this box will be redrawn on every frame. You must set this if box overlaps others to avoid flicker
floor-only	with floor-only="true" this box will only show the video floor holder, and will be blank if there is none
file-only	with file-only="true" this box will only show video files that are played via the conference play api.
audio-position	Uses OpenAL to position the audio into the location of the square the speaker is in. Values are x:y:z where x, y and z are values on 3d coordinate system. These can be positive or negative floating point numbers. The larger the number, the further the "distance". X=left/right, Y=forward/backard, X=up/down. These are the same values used in the conference position command.
auto-3d-position	Automatically position the conference member based on their location in the current layout.
border	Border size around each video feed (in pixels), values from 0 - 50. Overrides <layout> tag

Check this page for layout examples: FreeSWITCH 1.6 Video#Examples

See Also

Conference Add Call Example