

ConfBridge 1.10

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Overview

Asterisk, since its early days, has offered a conferencing application called MeetMe (app_meetme.so). MeetMe provides DAHDI-mixed software-based bridges for multi-party audio conferencing. MeetMe is used by nearly all Asterisk implementations - small office, call center, large office, feature-server, third-party application, etc. It has been extremely successful as an audio bridge.

Over time, several significant limitations of MeetMe have been encountered by its users. Among these are two of distinction: MeetMe requires DAHDI for mixing, and is thus limited to 8kHz (PSTN) audio sampling rates; and MeetMe is delivered in a fairly static form, it does not provide extensive configuration options.

To address these limitations, a new conferencing application, based upon the ConfBridge application introduced in Asterisk 1.6.0, is now available with Asterisk 1.10. This new ConfBridge application replaces the older ConfBridge application. It is not intended to be a direct replacement for MeetMe, it will not provide feature parity with the MeetMe application. Instead, the new ConfBridge application delivers a completely redesigned set of functionality that most users will find more than sufficient, and in many ways better, for their conferencing needs.

ConfBridge Concepts

ConfBridge provides four internal concepts:

1. Conference Number
2. Bridge Profile
3. User Profile
4. Conference Menu

A **Conference Number** is a numerical representation for an instance of the bridge. Callers joined to the same conference number will be in the same conference bridge; they're connected. Callers joined to different conference numbers are not in the same conference bridge; they're separated. Conference Numbers are assigned in the dialplan. Unlike MeetMe, they're not pre-reserved.

A **Bridge Profile** is a named set of options that control the behavior of a particular conference

bridge. Each bridge must have its own profile. A single bridge cannot have more than one Bridge Profile.

A **User Profile** is a named set of options that control the user's experience as a member of a particular bridge. Each user participating in a bridge can have their own individual User Profile.

A **Conference Menu** is a named set of options that are provided to a user when they present DTMF keys while connected to the bridge. Each user participating in a bridge can have their own individual Conference Menu.

ConfBridge Configuration

ConfBridge Profiles and Menus are configured in the `confbridge.conf` configuration file - normally located at `/etc/asterisk/confbridge.conf`. The file contains three reserved sections:

- `[general]`
- `[default_bridge]`
- `[default_user]`

The **[general]** section is currently unused, but is reserved for future use.

The **[default_bridge]** section contains all options invoked when ConfBridge is instantiated from the dialplan without a bridge profile argument.

The **[default_user]** section contains all options invoked when ConfBridge is instantiated from the dialplan with a user profile argument.

Each section contains a **type** definition. The type definition determines the function of the section. The three **types** are:

- `bridge`
- `user`
- `menu`

bridge is used to denote Bridge Profile section definitions.

user is used to denote User Profile section definitions.

menu is used to denote Conference Menu section definitions.

All other sections, defined by a section identifier encapsulated in square brackets, are user-definable.

Example

This is an example, using non-valid options and functions, of a `confbridge.conf` configuration file, displaying the organizational layout. The various options and functions are described later in this page.

```

[general]
; comments are preceded by a comma
;
; the general section is blank
;
[default_bridge]
type=bridge
; Bridge Profile options go here
myoption=value
myoption2=othervalue
;
[default_user]
type=user
; User Profile options go here
myoption=value
myoption2=othervalue
;
[sample_menu]
type=menu
; Conferece Menu options go here
DTMF=function
otherDTMF=otherFunction
;

```

Bridge Profile Configuration Options

A Bridge Profile provides the following configuration options:

Option	Values	Description
type	bridge	Set this to bridge to configure a bridge profile
max_members	integer; e.g. 50	Limits the number of participants for a single conference to a specific number. By default, conferences have no participant limit. After the limit is reached, the conference will be locked until someone leaves. Admin-level users are exempt from this limit and will still be able to join otherwise-locked, because of limit, conferences.
record_conference	yes/no	Records the conference call starting when the first user enters the room, and ending when the last user exits the room. The default recorded filename is 'confbridge-<name of conference bridge>-<start time>.wav and the default format is 8kHz signed linear. By default, this option is disabled. This file will be located in the configured monitoring directory as set in asterisk.conf
record_file	path, e.g. /tmp/myfiles	When record_conference is set to yes, the specific name of the recorded file can be set using this option. Note that since multiple conferences may use the same Bridge profile, this can cause issues, depending on the configuration. It is recommended to only use this option dynamically with the CONFBRIDGE() dialplan function. This allows the recorded name to be specified and a unique name to be chosen. By default, the recorded file is stored in Asterisk's spool/monitory directory, with a unique filename starting with the 'confbridge' prefix.

internal_sample_rate	auto, 8000, 12000, 16000, 24000, 32000, 44100, 48000, 96000, 192000	Sets the internal native sample rate at which to mix the conference. The "auto" option allows Asterisk to adjust the sample rate to the best quality / performance based on the participant makeup. Numbered values lock the rate to the specified numerical rate. If a defined number does not match an internal sampling rate supported by Asterisk, the nearest sampling rate will be used instead.
mixing_interval	10, 20, 40, 80	Sets, in milliseconds, the internal mixing interval. By default, the mixing interval of a bridge is 20ms. This setting reflects how "tight" or "loose" the mixing will be for the conference. Lower intervals provide a "tighter" sound with less delay in the bridge and consume more system resources. Higher intervals provide a "looser" sound with more delay in the bridge and consume less resources
sound_join	filename	The sound played to the bridge when a user joins, typically some kind of beep sound
sound_leave	filename	The sound played to the bridge when a user leaves, also typically some kind of beep sound
sound_has_joined	filename	The sound played as a user intro, e.g. "xxxx has joined the conference."
sound_has_left	filename	The sound played as a user parts the conference, e.g. "xxxx has left the conference."
sound_kicked	filename	The sound played to a user who has been kicked from the conference.
sound_muted	filename	The sound played to a user when the mute option is toggled on.
sound_unmuted	filename	The sound played to a user when the mute option is toggled off.
sound_only_person	filename	The sound played when a user is the only person in the conference.
sound_only_one	filename	The sound played to a user when there is only one other person in the conference.
sound_there_are	filename	The sound played when announcing how many users there are in a conference.
sound_other_in_party	filename	Used in conjunction with the sound_there_are option, used like "sound_there_are" <number of participants> "sound_other_in_party"
sound_place_into_conference	filename	The sound played when someone is placed into a conference, after waiting for a marked user.
sound_wait_for_leader	filename	The sound played when a user is placed into a conference that cannot start until a marked user enters.
sound_leader_has_left	filename	The sound played when the last marked user leaves the conference.
sound_get_pin	filename	The sound played when prompting for a conference PIN
sound_invalid_pin	filename	The sound played when an invalid PIN is entered too many (3) times
sound_locked	filename	The sound played to a user trying to join a locked conference.
sound_locked_now	filename	The sound played to an Admin-level user after toggling the conference to locked mode.
sound_unlocked_now	filename	The sound played to an Admin-level user after toggling the conference to unlocked mode.
sound_error_menu	filename	The sound played when an invalid menu option is entered.

Example

In this example, a Bridge Profile called "fancybridge" will be created. It will be configured to allow up to 20 callers, and will be set to mix at 10ms (tight mixing) at an automatic sampling rate. Additionally, it will be recorded.

```
[fancybridge]
type=bridge
max_members=20
mixing_interval=10
internal_sample_rate=auto
record_conference=yes
```

User Profile Configuration Options

A User Profile provides the following configuration options:

Option	Values	Description
type	user	Set this to user to configure a user profile
admin	yes/no	Sets if the user is an Admin or not. By default, no.
marked	yes/no	Sets if the user is Marked or not. By default, no.
startmuted	yes/no	sets if the user should start out muted. By default, no.
music_on_hold_when_empty	yes/no	Sets whether music on hold should be played when only one person is in the conference or when the user is waiting on a marked user to enter the conference. By default, off.
quiet	yes/no	When set to "yes," enter/leave prompts and user introductions are not played. By default, no.
announce_user_count	yes/no	Sets if the number of users in the conference should be announced to the caller. By default, no.
announce_user_count_all	yes/no; or an integer	Sets if the number of users should be announced to all other users in the conference when someone joins. When set to a number, the announcement will only occur once the user count is above the specified number
announce_only_user	yes/no	Sets if the only user announcement should be played when someone enters an empty conference. By default, yes.
wait_marked	yes/no	Sets if the user must wait for another marked user to enter before joining the conference. By default, no.
end_marked	yes/no	If enabled, every user with this option in their profile will be removed from the conference when the last marked user exists the conference.
dsp_drop_silence	yes/no	Drops what Asterisk detects as silence from entering into the bridge. Enabling this option will drastically improve performance and help remove the buildup of background noise from the conference. This option is highly recommended for large conferences, due to its performance improvements.
dsp_talking_threshold	integer in milliseconds	<p>The time, in milliseconds, by default 160, of sound above what the DSP has established as base-line silence for a user, before that user is considered to be talking. This value affects several options:</p> <ol style="list-style-type: none">1. Audio is only mixed out of a user's incoming audio stream if talking is detected. If this value is set too loose, the user will hear themselves briefly each time they begin talking until the DSP has time to establish that they are in fact talking.2. When talker detection AMI events are enabled, this value determines when talking has begun, which causes AMI events to fire. If this value is set too tight, AMI events may be falsely triggered by variants in the background noise of the caller.3. The drop_silence option depends on this value to determine when the user's audio should be mixed into the bridge after periods of silence. If this value is too loose, the beginning of a user's speech will get cut off as they transition from silence to talking.

dsp_silence_threshold	integer in milliseconds	The time, in milliseconds, by default 2500, of sound falling within what the DSP has established as the baseline silence, before a user is considered to be silent. The best way to approach this option is to set it slightly above the maximum amount of milliseconds of silence a user may generate during natural speech. This value affects several operations: <ol style="list-style-type: none"> 1. When talker detection AMI events are enabled, this value determines when the user has stopped talking after a period of talking. If this value is set too low, AMI events indicating that the user has stopped talking may get falsely sent out when the user briefly pauses during mid sentence. 2. The drop_silence option depends on this value to determine when the user's audio should begin to be dropped from the bridge, after the user stops talking. If this value is set too low, the user's audio stream may sound choppy to other participants.
talk_detection_events	yes/no	Sets whether or not notifications of when a user begins and ends talking should be sent out as events over AMI. By default, no.
denoise	yes/no	Whether or not a noise reduction filter should be applied to the audio before mixing. By default, off. This requires codec_speex to be built and installed. Do not confuse this option with drop_silence. denoise is useful if there is a lot of background noise for a user, as it attempts to remove the noise while still preserving the speech. This option does not remove silence from being mixed into the conference and does come at the cost of a slight performance hit.
jitterbuffer	yes/no	Whether or not to place a jitter buffer on the caller's audio stream before any audio mixing is performed. This option is highly recommended, but will add a slight delay to the audio and will incur a slight performance penalty. This option makes use of the JITTERBUFFER dialplan function's default adaptive jitter buffer. For a more fine-tuned jitter buffer, disable this option and use the JITTERBUFFER dialplan function on the calling channel, before it enters the ConfBridge application.
pin	integer	Sets if the user must enter a PIN before joining the conference. The user will be prompted for the PIN.
announce_join_leave	yes/no	When enabled, this option prompts the user for their name when entering the conference. After the name is recorded, it will be played as the user enters and exists the conference. By default, no.
dtmf_passthrough	yes/no	Whether or not DTMF received from users should pass through the conference to other users. By default, no.

Example

In this example, we will create a user profile called "fancyuser" that includes music on hold, user count announcements, join/leave announcements, silence detection, noise reduction and requires a PIN of 456.

```
[fancyuser]
type=user
music_on_hold_when_empty=yes
announce_user_count_all=yes
announce_join_leave=yes
dsp_drop_silence=yes
denoise=yes
pin=456
```

Conference Menu Configuration Options

A Conference Menu provides the following configuration options:

Option	Values	Description
type	menu	Set this to menu to configure a conference menu

playback	(<name of audio file1>&<name of audio file2>&...)	Plays back an audio file, or a string of audio files chained together using the & character, to the user and then immediately returns them to the conference.
playback_and_continue	(<name of audio file 1>&<name of audio file 2>&...)	Plays back an audio file, or a series of audio files chained together using the & character, to the user while continuing the collect the DTMF sequence. This is useful when using a menu prompt that describes all of the menu options. Note that any DTMF during this action will terminate the prompt's playback.
toggle_mute		Toggles mute on and off. When a user is muted, they will not be able to speak to other conference users, but they can still listen to other users. While muted, DTMF keys from the caller will continue to be collected.
no_op		This action does nothing. Its only real purpose exists for being able to reserve a sequence in the configuration as a menu exit sequence.
decrease_listening_volume		Decreases the caller's listening volume. Everything they hear will sound quieter.
increase_listening_volume		Increases the caller's listening volume. Everything they hear will sound louder.
reset_listening_volume		Resets the caller's listening volume to the default level.
decrease_talking_volume		Decreases the caller's talking volume. Everything they say will sound quieter to other callers.
increase_talking_volume		Increases the caller's talking volume. Everything they say will sound louder to other callers.
reset_talking_volume		Resets the caller's talking volume to the default level.
dialplan_exec	(context,exten,priority)	Allows one to escape from the conference and execute commands in the dialplan. Once the dialplan exits, the user will be put back into the conference.
leave_conference		Allows a user to exit the conference and continue execution in the dialplan.
admin_kick_last		Allows an Admin to remove the last participant from the conference. This action only works for users whose User Profiles set them as conference Admins.
admin_toggle_conference_lock		Allows an Admin to toggle locking and unlocking the conference. When the conference is locked, only other Admin users can join. When the conference is unlocked, any user may join up to the limit defined by the max_members Bridge Profile option. This action only works for users whose User Profiles set them as conference Admins.

Example

In this example, we'll create a menu called "fancymenu." This menu will utilize many of the options listed above. We will construct a features menu that plays when the user enters the * character. Since we will do this using the playback_and_continue option, we will define other menu items as being a "subset" of the * command, e.g. *4, so that once the user presses *, they can listen to the menu options and then press the specific "after-star" option, e.g. 4, to affect the option. Additionally, we will duplicate those same sub-features as non-* features, so that the user does not need to have entered the * menu structure in order to affect the options, they can just press the key, e.g. "4" at any time, regardless of whether or not they're in the *-tree.

```
[fancymenu]
type=menu
*=playback_and_continue(conf-togglemute&press&digits/1&silence/1&conf-
```

Of particular note in this example, we're calling the dialplan_exec option. Here, we're specifying "addcaller,1,1." This means that when someone dials 3, Asterisk will escape them out of the bridge momentarily to go execute priority 1 of extension 1 in the addcaller context of the dialplan (extensions.conf). Our dialplan, including the addcaller context, in this case, might look like:

```
[addcaller]
exten => 1,1,Originate(SIP/otherpeer,exten,conferences,100,1)

[conferences]
exten => 100,1,ConfBridge(1234)
```

Thus, when someone dials "3" while in the bridge, they'll Originate a call from the dialplan that puts SIP/otherpeer into the conference. Once the dial has completed, the person that dialed "3" will find themselves back in the bridge, with the other participants.

ConfBridge Dialplan Syntax

The syntax for the new ConfBridge application is as follows:

```
ConfBridge([confno][,bridge_profile[,user_profile[,menu]]])
```

The ConfBridge application takes the following arguments

- confno - The conference number
- bridge_profile - The Bridge Profile name from confbridge.conf. When left blank, a dynamically built Bridge Profile created by the CONFBRIDGE dialplan function is searched for on the channel and, if available, used. If no dynamic profile is found, the "default_bridge" profile found in confbridge.conf is used.



It is important to note that while User Profiles are unique for each participant, Bridge Profiles are unique to the bridge, **not** the user. So you can only create one Bridge Profile per conference

- user_profile - The User Profile name from confbridge.conf. When left blank, a dynamically built User Profile created by the CONFBRIDGE dialplan function is searched for on the channel and, if available, used. If no dynamic profile is present, the "default_user" profile found in confbridge.conf is used.
- menu - The Conference Menu name from confbridge.conf. No menu is applied by default if this option is left blank.

Example 1

In this example, callers will be joined to conference number 1234, using the default Bridge Profile, the default User Profile, and no Conference Menu.

```
exten => 1,1,Answer( )
exten => 1,n,ConfBridge(1234)
```

Example 2

In this example, callers will be joined to conference number 1234, with the default Bridge Profile, a User Profile called "1234_participants" and a Conference Menu called "1234_menu."

```
exten => 1,1,Answer( )
exten => 1,n,ConfBridge(1234,,1234_participants,1234_menu)
```

ConfBridge Dialplan Functions

The CONFBRIDGE dialplan function is used to set customized Bridge and/or User Profiles on a channel for the ConfBridge application. It uses the same options defined in confbridge.conf and allows the creation of dynamic, dialplan-driven conferences.

Syntax

```
CONFBRIDGE(type,option)
```

- type - Refers to which type of profile the option belongs to. Type can be either "bridge" or "user."
- option - Refers to the confbridge.conf option that is to be set dynamically on the channel. This can also refer to an existing Bridge or User Profile by using the keyword "template." In this case, an existing Bridge or User Profile can be appended or modified on-the-fly.

Examples

Example 1

In this example, the custom set User Profile on this channel enables announce_join_leave (so users will be announced as they come and go), sets users to join muted (so that they're not able to speak), and pushes them into bridge "1."

```
exten => 1,1,Answer()  
exten => 1,n,Set(CONFBRIDGE(user,announce_join_leave)=yes)  
exten => 1,n,Set(CONFBRIDGE(user,startmuted)=yes)  
exten => 1,n,ConfBridge(1)
```

Example 2

In this example, we will include an existing User Profile, the default_user User Profile as defined in confbridge.conf, and we will set additional parameters (admin and marked) that aren't already defined in the default_user User Profile.

```
exten => 1,1,Answer()  
exten => 1,n,Set(CONFBRIDGE(user,template)=default_user)  
exten => 1,n,Set(CONFBRIDGE(user,admin)=yes)  
exten => 1,n,Set(CONFBRIDGE(user,marked)=yes)  
exten => 1,n,ConfBridge(1)
```

ConfBridge CLI Options

ConfBridge offers several options that may be invoked from the Asterisk CLI.

confbridge kick <conference> <channel>

Removes the specified channel from the conference, e.g.:

```
*CLI> confbridge kick 1111 SIP/mypeer-00000000
Kicking SIP/mypeer-00000000 from confbridge 1111
```

confbridge list

Shows a summary listing of all bridges, e.g.:

```
*CLI> confbridge list
Conference Bridge Name          Users   Marked Locked?
=====
1111                            1       0 unlocked
```

confbridge list <conference>

Shows a detailed listing of participants in a specified conference, e.g.:

```
*CLI> confbridge list 1111
Channel                        User Profile      Bridge Profile    Menu
=====
=====
SIP/mypeer-00000001          default_user      1111
sample_user_menu
```

confbridge lock <conference>

Locks a specified conference so that only Admin users can join, e.g.:

```
*CLI> confbridge lock 1111
Conference 1111 is locked.
```

confbridge unlock <conference>

Unlocks a specified conference so that only Admin users can join, e.g.:

```
*CLI> confbridge unlock 1111
Conference 1111 is unlocked.
```

confbridge mute <conference> <channel>

Mutes a specified user in a specified conference, e.g.:

```
*CLI> confbridge mute 1111 SIP/mypeer-00000001
Muting SIP/mypeer-00000001 from confbridge 1111
```

confbridge unmute <conference> <channel>

Unmutes a specified user in a specified conference, e.g.:

```
*CLI> confbridge unmute 1111 SIP/mypeer-00000001
Unmuting SIP/mypeer-00000001 from confbridge 1111
```

confbridge record start <conference> <file>

Begins recording a conference. If "file" is specified, it will be used, otherwise, the Bridge Profile record_file will be used. If the Bridge Profile does not specify a record_file, one will be automatically generated in Asterisk's monitor directory. Usage:

```
*CLI> confbridge record start 1111
Recording started
*CLI> == Begin MixMonitor Recording
ConfBridgeRecorder/conf-1111-uid-618880445
```

confbridge record stop <conference>

Stops recording the specified conference, e.g.:

```
*CLI> confbridge record stop 1111
Recording stopped.
*CLI> == MixMonitor close filestream (mixed)
== End MixMonitor Recording
ConfBridgeRecorder/conf-1111-uid-618880445
```

confbridge show menus

Shows a listing of Conference Menus as defined in confbridge.conf, e.g.:

```
*CLI> confbridge show menus
----- Menu -----
sample_admin_menu
sample_user_menu
```

confbridge show menu <menu name>

Shows a detailed listing of a named Conference Menu, e.g.:

```
*CLI> confbridge show menu sample_admin_menu
Name: sample_admin_menu
*9=increase_talking_volume
*8=no_op
*7=decrease_talking_volume
*6=increase_listening_volume
*4=decrease_listening_volume
*3=admin_kick_last
*2=admin_toggle_conference_lock
*1=toggle_mute
*=playback_and_continue(conf-adminmenu)
```

confbridge show profile bridges

Shows a listing of Bridge Profiles as defined in confbridge.conf, e.g.:

```
*CLI> confbridge show profile bridges
----- Bridge Profiles -----
1111
default_bridge
```

confbridge show profile bridge <bridge>

Shows a detailed listing of a named Bridge Profile, e.g.:

```
*CLI> confbridge show profile bridge 1111
-----
Name: 1111
Internal Sample Rate: 16000
Mixing Interval: 10
Record Conference: no
Record File: Auto Generated
Max Members: No Limit
sound_only_person: conf-onlyperson
sound_has_joined: conf-hasjoin
sound_has_left: conf-hasleft
sound_kicked: conf-kicked
sound_muted: conf-muted
sound_unmuted: conf-unmuted
sound_there_are: conf-thereare
sound_other_in_party: conf-otherinparty
sound_place_into_conference: conf-placeintoconf
sound_wait_for_leader: conf-waitforleader
sound_get_pin: conf-getpin
sound_invalid_pin: conf-invalidpin
sound_locked: conf-locked
sound_unlocked_now: conf-unlockednow
sound_lockednow: conf-lockednow
sound_error_menu: conf-errormenu
```

confbridge show profile users

Shows a listing of User Profiles as defined in confbridge.conf, e.g.:

```
*CLI> confbridge show profile users
----- User Profiles -----
awesomeusers
default_user
```

confbirdge show profile user <user>

Shows a detailed listing of a named Bridge Profile, e.g.:

```
*CLI> confbridge show profile user default_user
```

```
-----
```

Name:	default_user
Admin:	false
Marked User:	false
Start Muted:	false
MOH When Empty:	enabled
MOH Class:	default
Quiet:	disabled
Wait Marked:	disabled
END Marked:	disabled
Drop_silence:	enabled
Silence Threshold:	2500ms
Talking Threshold:	160ms
Denoise:	disabled
Talk Detect Events:	disabled
DTMF Pass Through:	disabled
PIN:	None
Announce User Count:	enabled
Announce join/leave:	enabled
Announce User Count all:	enabled

ConfBridge Asterisk Manager Interface (AMI) Actions

ConfbridgeList

Lists all users in a particular ConfBridge conference. ConfbridgeList will follow as separate events, followed by a final event called ConfbridgeListComplete

Example

```
Action: ConfbridgeList
Conference: 1111

Response: Success
EventList: start
Message: Confbridge user list will follow

Event: ConfbridgeList
Conference: 1111
CallerIDNum: malcolm
CallerIDName: malcolm
Channel: SIP/malcolm-00000000
Admin: No
MarkedUser: No

Event: ConfbridgeListComplete
EventList: Complete
ListItems: 1
```

ConfbridgeListRooms

Lists data about all active conferences. ConfbridgeListRooms will follow as separate events, followed by a final event called ConfbridgeListRoomsComplete.

Example

```
Action: ConfbridgeListRooms

Response: Success
EventList: start
Message: Confbridge conferences will follow

Event: ConfbridgeListRooms
Conference: 1111
Parties: 1
Marked: 0
Locked: No

Event: ConfbridgeListRoomsComplete
EventList: Complete
ListItems: 1
```

ConfbridgeMute

Mutes a specified user in a specified conference.

Example

```
Action: ConfbridgeMute
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: User muted
```

ConfbridgeUnmute

Unmutes a specified user in a specified conference.

Example

```
Action: ConfbridgeUnmute
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: User unmuted
```

ConfbridgeKick

Removes a specified user from a specified conference.

Example

```
Action: ConfbridgeKick
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: User kicked
```

ConfbridgeLock

Locks a specified conference.

Example


```
Action: ConfbridgeLock
Conference: 1111

Response: Success
Message: Conference locked
```

ConfbridgeUnlock

Unlocks a specified conference.

Example

```
Action: ConfbridgeUnlock
Conference: 1111

Response: Success
Message: Conference unlocked
```

ConfbridgeStartRecord

Starts recording a specified conference, with an optional filename. If recording is already in progress, an error will be returned. If RecordFile is not provided, the default record_file as specified in the conferences Bridge Profile will be used. If record_file is not specified, a file will automatically be generated in Asterisk's monitor directory.

Example

```
Action: ConfbridgeStartRecord
Conference: 1111

Response: Success
Message: Conference Recording Started.

Event: VarSet
Privilege: dialplan,all
Channel: ConfBridgeRecorder/conf-1111-uid-1653801660
Variable: MIXMONITOR_FILENAME
Value: /var/spool/asterisk/monitor/confbridge-1111-1303309869.wav
Uniqueid: 1303309869.6
```

ConfbridgeStopRecord

Stops recording a specified conference.

Example

```
Action: ConfbridgeStopRecord
Conference: 1111

Response: Success
Message: Conference Recording Stopped.

Event: Hangup
Privilege: call,all
Channel: ConfBridgeRecorder/conf-1111-uid-1653801660
Uniqueid: 1303309869.6
CallerIDNum: <unknown>
CallerIDName: <unknown>
Cause: 0
Cause-txt: Unknown
```

ConfBridge Asterisk Manager Interface (AMI) Events

ConfbridgeStart

This event is sent when the first user requests a conference and it is instantiated

Example

```
Event: ConfbridgeStart
Privilege: call,all
Conference: 1111
```

ConfbridgeJoin

This event is sent when a user joins a conference - either one already in progress or as the first user to join a newly instantiated bridge.

Example

```
Event: ConfbridgeJoin
Privilege: call,all
Channel: SIP/mypeer-00000001
Uniqueid: 1303309562.3
Conference: 1111
CallerIDnum: 1234
CallerIDname: mypeer
```

ConfbridgeLeave

This event is sent when a user leaves a conference.

Example

```
Event: ConfbridgeLeave
Privilege: call,all
Channel: SIP/mypeer-00000001
Uniqueid: 1303308745.0
Conference: 1111
CallerIDnum: 1234
CallerIDname: mypeer
```

ConfbridgeEnd

This event is sent when the last user leaves a conference and it is torn down.

Example

```
Event: ConfbridgeEnd
Privilege: call,all
Conference: 1111
```

ConfBridgeTalking

This event is sent when the conference detects that a user has either begin or stopped talking.

Start talking Example

```
Event: ConfbridgeTalking
Privilege: call, all
Channel: SIP/mypeer-00000001
Uniqueid: 1303308745.0
Conference: 1111
TalkingStatus: on
```

Stop talking Example

```
Event: ConfbridgeTalking
Privilege: call, all
Channel: SIP/mypeer-00000001
Uniqueid: 1303308745.0
Conference: 1111
TalkingStatus: off
```

Usage Notes, FAQ and Other

There are many points to consider when using the new ConfBridge application. Some will be examined here.

Mixing Interval

The mixing interval for a conference is defined in its Bridge Profile. The allowable options are 10, 20, 40, and 80, all in milliseconds. Usage of 80ms mixing intervals is only supported for conferences that are sampled at 8, 12, 16, 24, 32, and 48kHz. Usage of 40ms intervals includes all of the aforementioned sampling rates as well as 96kHz. 192kHz sampled conferences are only supported at 10 and 20ms mixing intervals. These limitations are imposed because higher mixing intervals at the higher sampling rates causes large increases in memory consumption. Adventurous users may, through changing of the MAX_DATALEN define in bridge_softmix.c allow 96kHz and 192kHz sampled conferences to operate at longer intervals - set to 16192 for 96kHz at 80ms or 32384 for 192kHz at 80ms, recompile, and restart.

Maximizing Performance

In order to maximize the performance of a given machine for ConfBridge purposes, there are several steps one should take.

- Enable `dsp_drop_silence` is enabled in the User Profile.
 - This is the **single most** important step one can take when trying to increase the number of bridge participants that a single machine can handle. Enabling this means that the audio of users that aren't speaking isn't mixed in with the bridge.
- Lengthen `mixing_interval` in the Bridge Profile.
 - The default interval is 20ms. Other options are 10, 40, and 80ms. Lower values provide a "tighter" sound, but require substantially more CPU. Higher values provide a "looser" sound, and consume substantially less CPU. Setting the value to 80 provides the highest number of possible participants.
- Connect clients at the same sampling rate.
 - Requiring the bridge to resample between clients that use codecs with different sampling rates is an expensive operation. If all clients are dialed in to the bridge at the same sampling rate, and the bridge operates at that same rate, e.g. 16kHz, then the number of possible clients will be maximized.
- Run Asterisk with a higher priority.
 - By default, Asterisk operates at a relatively normal priority, as compared to other processes on the system. To maximize the number of possible clients, Asterisk should be started using the `-p` (realtime) flag. If the load becomes too large, this can negatively impact the performance of other processes, including the console itself - making it difficult to remotely administer a fully loaded system.

As the number of clients approaches the maximum possible on the given machine, given its processing capabilities, audio quality will suffer. Following the above guidelines will increase the number of connected clients before audio quality suffers.

Sound Prompts

The following Conference Menu and Bridge Profile options should have sound files associated with them. At the present time, prior to the actual release of Asterisk 1.10, many of these sound files do not yet exist; though some already do. Comments or suggestions on phrasings are welcome.

- `decrease_listening_volume` - "...to decrease the audio volume from other participants" and "To decrease the audio volume from other participants..."
- `increase_listening_volume` - "...to increase the audio volume from other participants" and "To increase the audio volume from other participants..."
- `reset_listening_volume` - "...to reset the audio volume of the conference to the default level" and "To reset the audio volume of the conference to the default level..."
- `decrease_talking_volume` - "...to decrease your speaking volume to other participants" and "To decrease your speaking volume to other participants..."

participants..."

- increase_talking_volume - "...to increase your speaking volume to other participants" and "To increase your speaking volume to other participants..."
- reset_talking_volume - "...to reset your speaking speaking volume to the default level" and "To reset your speaking volume to the default level..."
- admin_kick_last - "...to remove the participant who most recently joined the conference" and "To remove the participant who most recently joined the conference..."
- sound_muted - "You are now muted"
- sound_wait_for_leader - "The conference will begin when the leader arrives." Also add "The conference will begin when our **glorious leader** arrives"
- sound_locked_now - "The conference is now locked."
- toggle_mute - "...to mute or unmute yourself" and "To mute or unmute yourself..." and an extended prompt that says "...to mute or unmute yourself. When you are muted, you cannot send audio to other participants; however, you will still hear audio from other unmuted participants."
- leave_conference - "...to leave the conference" and "To leave the conference..."
- admin_toggle_conference_lock - "...to lock or unlock the conference" and "To lock or unlock the conference..." and an extended prompt that says "...to lock or unlock the conference. When a conference is locked, only conference administrators can join."
- sound_has_joined - "...has joined the conference."
- sound_has_left - "...has left the conference."
- sound_kicked - "You have been removed from the conference."
- sound_unmuted - "You are no longer muted."
- sound_only_person - "You are currently the only participant in the conference."
- sound_there_are - "There are currently..."
- sound_other_in_party - "...participants in the conference."
- sound_place_into_conference - "There conference will now begin."
- sound_leader_has_left - "The conference has ended."
- sound_get_pin - "Please enter your personal identification number, followed by the pound or hash key."
- sound_invalid_pin - "You have entered too many invalid personal identification numbers."
- sound_locked - "The conference is currently locked and cannot be joined."
- sound_unlocked_now - "The conference is now unlocked."
- sound_error_menu - "You have entered an invalid option."
- exit_menu - "...to exit the menu." and "To exit the menu..."