Freeswitch Conference Commands

About conference services

The Conference Web Services APIs allow the administrator to send commands to the FreeSwitch platform.

Base URL

The base URL for the Conference web services is:
https://\{user id}:{password}@{admin config host name}/sipxconfig/rest/my/conference/{conference-name}
For the above URL you can use the */{command}&{arg 1}&{arg 2}...* URL with the **PUT** HTTP method to send the desired commands and arguments to FreeSwitch.

Available FreeSwitch commands and arguments

Command name	Command details	Usage
bgdial		<endpoint_module_name> /<destination> <callerid number> <callerid name>
deaf	Make a conference member deaf.	<[member_id|all]|last>
dial	Dial a destination via a specific endpoint. For more information, click here.	<endpoint_module_name> /<destination> <callerid number> <callerid name>
dtmf	Send DTMF to any member of the conference.	<[member_id|all|last]> <digits>
energy	Adjusts the conference energy level for a specific member.	<member_id|all|last> [<newval>]
hup	Kick without the kick sound.	conference <confname> hup < [member_id|all|last]>
kick	Kicks a specific member form a conference.	<[member_id|all|last]>
list	Lists all or a specific conference members. For more information, click here.	conference list [delim <string>]
		conference <confname> list [delim <string>]
lock	Lock a conference so no new members will be allowed to enter.	lock
mute	Mutes a specific member in a conference.	<[member_id|all]|last>

norecord		<[filename|all]>
nopin	Removes a pin for a specific conference.	nopin
pin	Sets or changes a pin number for a specific conference. **Note**: if you set a conference pin and then issue a command like conference <confname> dial sofia/default/123456@softswitch 123456 will not be challenged with a pin, but he will just joins the conference named <confname>.	<pin#>
play	Play an audio file in a conference to all members or to a specific member. You can stop that same audio with the Stop command below.	<file_path> [async|<member_id>]
record		<filename>
relate	Mute or Deaf a specific member to another member. For more information, click here.	<member_id> <other_member_id> [nospeak|nohear|clear]
say	Write a message to all members in the conference.	<text>
saymember	Write a messaget to a specific member in a conference.	<member_id> <text>
stop	Stops any queued audio from playing.	<[current|all|async|last]> [<member_id>]
transfer	Transfer a member from one conference to another conference. To transfer a member to another extension use the api transfer command with the uuid of their session.	<conference_name> <member id> [... <member id>]
unmute	Unmute a specific member of a conference.	<[member_id|all]|last>
undeaf	Allow a specific member to hear the conference..	<[member_id|all]|last>
unlock	Unlock a conference so that new members can enter.	unlock
volume_in	Adjusts the input volume for a specific conference member.	<member_id|all|last> [<newval>]
volume_out	Adjusts the output volume for a specific conference member.	<member_id|all|last> [<newval>]
xml_list		

Specific response codes:

Error 404 - when {confName} not found
Error 403 - when authenticated user is not owner of {confName}
Error 400 - when no {command} is specified or the command is incorect

Dial *Additional Information*

If the caller ID values are not set, the variables set in the *conference.conf.xml* are used. Specifically, the value for *caller-id-number* is used for the number and the value for c*aller-id-name* is used for the name.
If the conference is dynamically created as a result of this API and the *caller-id-number* and *caller-id-number* is not provided in the API call then the number and name will be "*00000000*" and respectively "*FreeSWITCH*".
Example:
conference testconf dial {originate_timeout=30}sofia/default/1000@softswitch 1234567890 FreeSWITCH_Conference
The above API call will dial out of a conference named "testconf" to the user located at the specified endpoint with a 30 second timeout. The endpoint will see the call as coming from "FreeSWITCH_Conference" with a caller id of 1234567890.
Note: The values provided in the dial string overwrite the *caller-id-number* and c*aller-id-name* variables provided at the end of the API call.
Click here for the list of dial string options available.

List *Additional Information*

The output generated by the system is named by default with the following format:

<conference name> (<member_count> member[s][locked]),

where *locked* can represent either the locked or unlockes status of the conference.
The following items are a separated list in CSV format for each conference leg:

Item	Description
ID of participant	
Register string of participants	
UUID of participants call leg	
Caller ID number	
Caller ID name	
Status	Options are: • *Hear* - The mute/unmute status of the member. • *Speak* - The "deaf /undeaf" status of the member. • *Talking* - The input channel is providing some amount of sound energy. • *Video* - Providing video? • *Floor* - This member currently owns the floor.
Volume In	
Volume Out	
Energy Level	

Relate *Additional Information*

Examples:

- *conference my_conf relate 1 2 nospeak:* Member 1 may now no longer speak to member 2, i.e. member 2 now cannot hear member 1.
- *conference my_conf relate 1 2 clear:* Member 1 may now speak to member 2 again
- *conference my_conf relate 1 2 nohear:* Member 1 now cannot hear member 2
- *confernce my_conf relate 1 2 clear:* Member 1 can now hear member 2 again

Examples

Lock a conference with name "WeeklyTeamConf":
curl --digest -k -X PUT https://200:123@localhost/sipxconfig/rest/my/conference/WeeklyTeamConf/lock
Invite user in conference given username:
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/invite\&401
Invite user in conference given instant message ID:
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/inviteim\&401im
Other examples:
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/xml_list
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/kick\&all
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/record
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/record\&stop
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/record\&status
curl --digest -k https://400:123@gerula-dev.buc.ro/sipxconfig/rest/my/conference/Conf400/record\&duration
Sample PHP click to call code:

```php
<?php
$to="101";//Number to dial
$from="5001";//userid in sipx
$pass="1234";//sipx pin (NOT SIP password)
//replace sipx.gcgov.local with your sipx server
$url = "http://sipx.gcgov.local:6667/callcontroller/".$from."/".$to."?isForwardingAllowed=true";
$ch = curl_init();
curl_setopt($ch, CURLOPT_URL, $url);
curl_setopt($ch, CURLOPT_HTTPAUTH, CURLAUTH_DIGEST);
curl_setopt($ch, CURLOPT_POST, 1);
curl_setopt($ch, CURLOPT_USERPWD, $from.":".$pass);
$result = curl_exec($ch);
curl_close($ch);
?>
```

Sample contact information:

```
<contact-information>
<jobTitle>Data Entry Assistant</jobTitle>
<jobDept>Data Management Services</jobDept>
<companyName>Museum of Science</companyName>
<homeAddress>
<city>NY</city>
</homeAddress>
<officeAddress>
<street>1 Science Park</street>
<city>Boston</city>
<country>US</country>
<state>MA</state>
<zip>02114</zip>
</officeAddress>
<imId>myId</imId>
<emailAddress>john.doe@example.com</emailAddress>
<useBranchAddress>false</useBranchAddress>
<avatar>https://secure.gravatar.com/avatar/8eb1b522f60d11fa897de1dc6351b7e8?s=80&amp;d=G</avatar>
<firstName>John</firstName>
<lastName>Doe</lastName>
</contact-information>
```