

Conferencing

Introduction

With sipXecs release 4.0 (development release 3.11) sipXecs introduces a conferencing server. This conferencing service allows voice conferencing (meet-me). The conferencing solution is easy to use as it is integrated into the GUI management application of sipXecs called sipXconfig. Conference servers can be created and administered to either run on the same host as the rest of the sipXecs solution or on a separate host dedicated to conferencing services. Dynamic conference controls are integrated into the user portal so that every user gets a personal conference server that can easily be administered.

The sipXecs conferencing solution is based on a cooperation with the [FreeSWITCH project](#). In our view FreeSWITCH is now mature enough to be used in a production environment as a conferencing server. FreeSWITCH is [superior to Asterisk in many respects](#). While conferencing is the first media application we have based on FreeSWITCH, it is likely that more will follow in the future.

Performance

We are still in the process of completing all the load and performance tests. Our objective is to provide a conferencing server that offers a minimum of 500 ports on standard hardware. Test results done by FreeSWITCH indicate that the final number could be significantly higher than that. We will publish more detailed results as we finish the testing.

The conferencing server also supports several codecs and is able to mix audio that arrives using different codecs from clients.

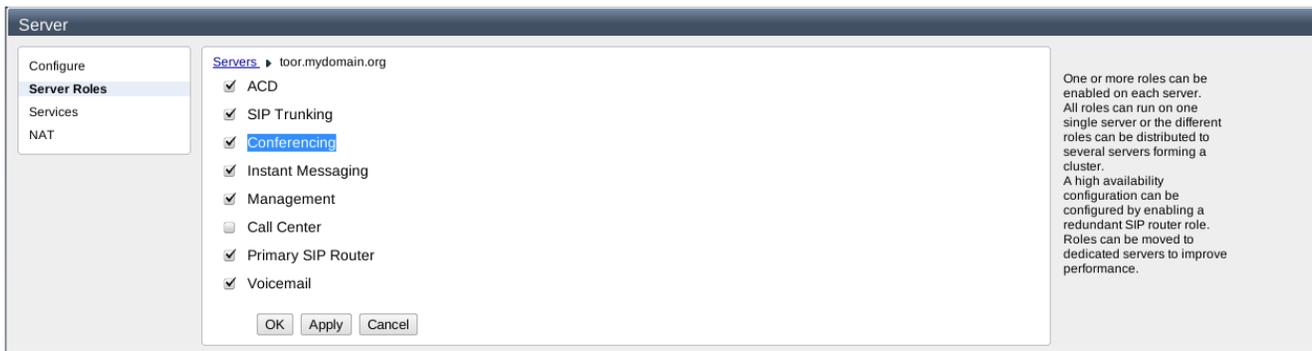
sipXecs & FreeSWITCH

sipXecs & FreeSWITCH integration is designed to create a plug & play distributed conferencing solution managed by the sipXecs Configuration Server. Ease of use has been one of the major design goals. In addition, FreeSWITCH has been fully integrated into the sipXecs build and installation process. FreeSWITCH installs like any other distributed component of the sipXecs solution. It is integrated into the sipXecs process management. Its configuration is auto-generated by sipXconfig.

The conference server is an independent application (FreeSWITCH) that runs on the same server as the rest of the sipXecs system. The conference server communicates with conference participants using the SIP and RTP protocols. It is configured through the sipXecs Configuration Server (sipXconfig) that populates configuration files and asks FreeSWITCH to reload those settings over XML RPC dynamically.

Enabling Conferencing role

For enabling *Conferencing* role navigate to System > Servers page and choose the location you want Conference Server to run on. On the Server Role tab enable *Conferencing* option. After adding *Conferencing* role you will be warned that FreeSWITCH configuration should be reloaded so the changes to take effect (by a message saying *One or more services need to be reloaded. For details click: here*)



Conference Server Configuration

The conference server and individual conferences are configured by selecting the *Conferencing* entry under the *Features* menu (see below). Several conferencing servers can be configured where each conferencing server can run on dedicated hardware. For each conferencing server the user interface shows how many conferences are currently active on that server.

SIPX ECS Fri, 11 Feb 2011 1:57 PM | Home | Help | Logout | Search

Users | Devices | Features | System | Diagnostics

Conference Servers

	Name	De	
<input type="checkbox"/>	toor.mydomain.org	Prim	<div style="float: right; text-align: right;"> <input checked="" type="checkbox"/> Refresh every 30 seconds </div> <div style="clear: both;"></div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Conferencing </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Conferences </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Auto Attendants </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Intercom </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Paging Groups </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Hunt Groups </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Call Park </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Music on Hold </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Phonebooks </div> <div style="border: 1px solid #ccc; padding: 2px; margin-bottom: 2px;"> Instant Messaging </div>

Quick Links

[Servers](#)
[User Groups](#)

Conference Servers are created and administered under System / Servers. A single conference server can host a large number of conferences. For every user it is possible to automatically assign a personal conference. Go to User Groups / Conference Assignments to configure this feature before creating the users.

The conference server can run on dedicated hardware or be collocated with other services. Several conference servers can be created per system.

This page will refresh automatically. You can switch automatic refreshing off by clearing the Refresh checkbox.

You can also modify the refresh interval by clicking on the current interval and then enter a new value.

Navigate to *Conference Server > Configuration* tab to modify conference server settings. These settings are applied to all conferences on this conference server.

Conference Server

Configuration

Conferences

[Conferencing](#) > toor.mydomain.org

Host **toor.mydomain.org**

IP address **192.168.160.105**

Server description **Primary server**

DTMF Commands for a Conference Participant

Mute Audio

Toggle in Audio (Default: 0)
Toggle input audio to conference from participant's own call leg.

Toggle all Audio (Default: *)
Toggle audio both to and from the conference.

Voice Activity Detection

Energy Threshold UP (Default: 9)
Increase minimum volume required to send audio to the conference.

Energy Threshold RESET (Default: #)
Reset minimum volume required to send audio to the conference to default.

Energy Threshold DOWN (Default: 7)
Decrease minimum volume required to send audio to the conference.

Conference Audio

Volume UP (Default: 6)
Increase the audio volume coming out from the conference.

Volume RESET (Default: 5)
Reset the audio volume coming out from the conference to default.

Volume DOWN (Default: 4)
Decrease the audio volume coming out from the conference.

Microphone Gain UP (Default: 3)
Increase the volume sent to the conference.

Microphone Gain RESET (Default: 2)

Conference Settings

For adding a new conference navigate to *Conference Server > Conferences* tab and access *Add new Conference* link. Every conference has an extension number and can be assigned a DID Number (used for external accessibility). In addition every conference can have an owner and a participant access code.

Conference

Configuration

Conferencing > toor.mydomain.org > New Conference

Enabled

Name

Extension

DID Number

Another alias for the conference

Description

Conference owner John (200) [Change owner...](#) [Unassign](#)

The user that should have permission to administer and control this conference. Unassigned conferences may only be controlled by administrators.

Auto-record

If checked then conference calls will be recorded to the conference owner's mailbox

Participant PIN (Default: 6432)

Participant access code for both audio conference and Dindim web conference. Can be empty.

Maximum legs (Default: 0)

The maximum number of call legs to be allowed by this bridge. 0 means unlimited.

Music On Hold source (Default: System Music Directory)

Selects the source of the on hold music for this conference. System Music Directory option will play all the music files in the Music On Hold directory on a continuous rotating basis. Sound Card option will stream audio from the local sound card.

[OK](#) [Apply](#) [Cancel](#)

If *Auto-record* is enabled for a conference bridge then the owner of that conference receives the recording of that conference call in a new "conference" mailbox. The conference bridge must have an owner or recording cannot be configured. Conference call recording must be enabled before the call starts. The call is recorded from the time the first user joins the call until the time the last user leaves the call. Once the recording is complete the conference owner may log into their user portal and listen to the conference recording. They may also save the conference recording onto their desktop machine.

Note: Recorded conference calls do not light the user's MWI lamp since they are not messages for the user.

Dynamic Conference Controls

Conference

Configuration

Participants

Web Conference

IM Chat Room

Conferencing > toor.mydomain.org > myConf

Refresh every 30 seconds

Status	Name	Nickname
	AudioCodes (601@toor.mydomain.org)	Add a nickname
	401 (401@toor.mydomain.org)	Add a nickname

[Isolate](#) [Include](#) [Mute](#) [Un-Mute](#) [Disconnect](#) [Refresh](#)

[Invite Participant](#)

This page will refresh automatically. You can switch automatic refreshing off by clearing the *Refresh* checkbox. You can also modify the refresh interval by clicking on the current interval and then enter a new value.

Conference Chat Room

If Instant Messaging role is enabled every voice conference will have an associated chat room. The name of the chat room is the same as the name of the conference.

Conference

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Moderated (Default: unchecked)
If checked, only the owner of the chat room will be able to send messages; external users can be given chat privileges from the owner. The owner can grant other users chat privileges by issuing the following command on the client "role participant nickname".

Public (Default: checked)
A public room can be found by any user through normal means such as searching and service discovery.

OK Apply Cancel

If Instant Messaging role is enabled every voice conference will have an associated chat room. The name of the chat room is the same as the name of the conference.

Checking Status

Once configuration settings are applied, sipXconfig publishes this information to FreeSWITCH. Select "Job Status" from the "Diagnostics" menu to view status:

Job Status

Refresh every 30 seconds

Job	Start Time	Stop Time	Status	Error / Warning
Data replication: alias	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: validusers.xml	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: sipX_context.xml	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: conference.conf.xml	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: autoattendants.xml	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: conferencebridge.xml	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: contact-information.xml	2/11/11 4:53 PM	2/11/11 4:53 PM	Completed	
File replication: xmpp-account-info.xml	2/11/11 4:54 PM	2/11/11 4:54 PM	Completed	
File replication: presenceroouting-prefs.xml	2/11/11 4:54 PM	2/11/11 4:54 PM	Completed	
File replication: sipxopenfire.xml	2/11/11 4:54 PM	2/11/11 4:54 PM	Completed	

Clear Completed Clear All Refresh

The Job Status page is for diagnostics purposes only and provides information about system management and configuration activity to trained technicians. A failed status typically indicates a serious system problem that requires immediate attention. In a distributed (clustered) system a failure status can indicate a network or hardware problem. To troubleshoot try restarting the servers, check your network or contact support. This page will refresh automatically. You can switch automatic refreshing off by clearing the *Refresh* checkbox. You can also modify the refresh interval by clicking on the current interval and then enter a new value.

FreeSWITCH reload configuration

An XML-RPC call is made to FreeSWITCH, where FreeSWITCH is asked to reload the conference admission and configuration data.