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.

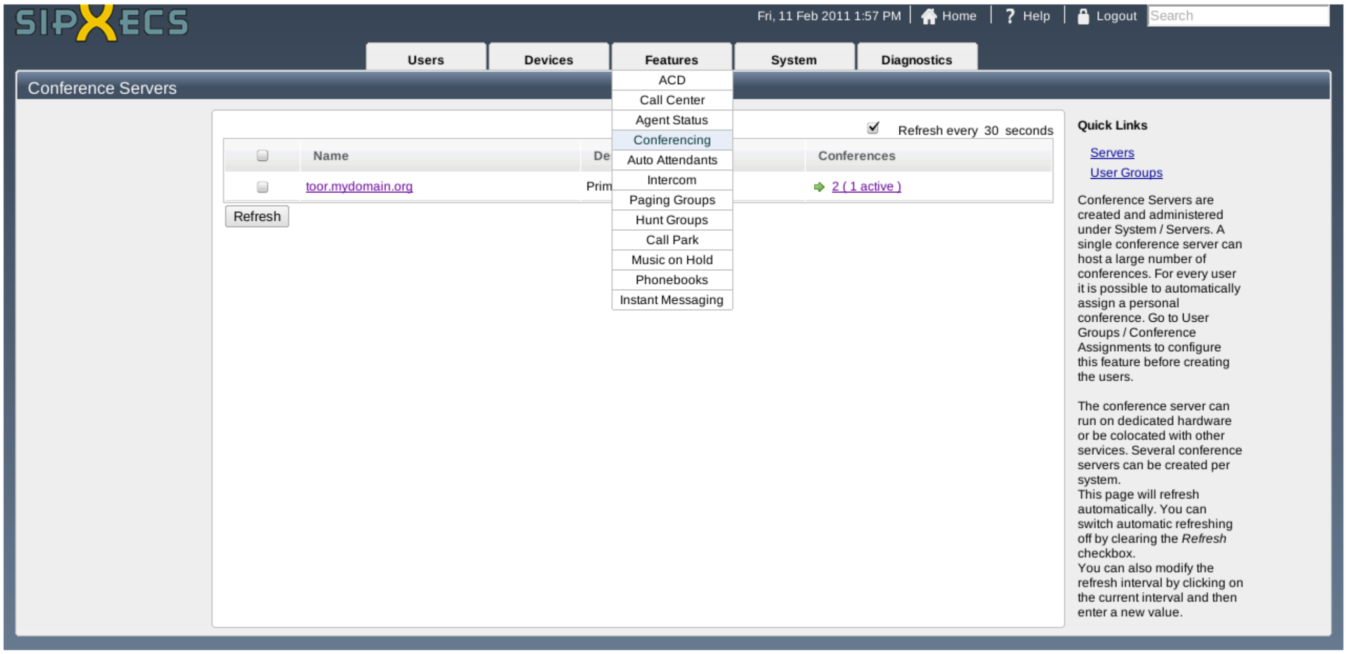
The conference server is an independent application (FreeSWITCH) that runs on the same server as the rest of the sipXecs system. The conference server communicates with conference participants using the SIP and RTP protocols. It is configured through the sipXecs Configuration Server (sipXconfig) that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

Enabling Conferencing role

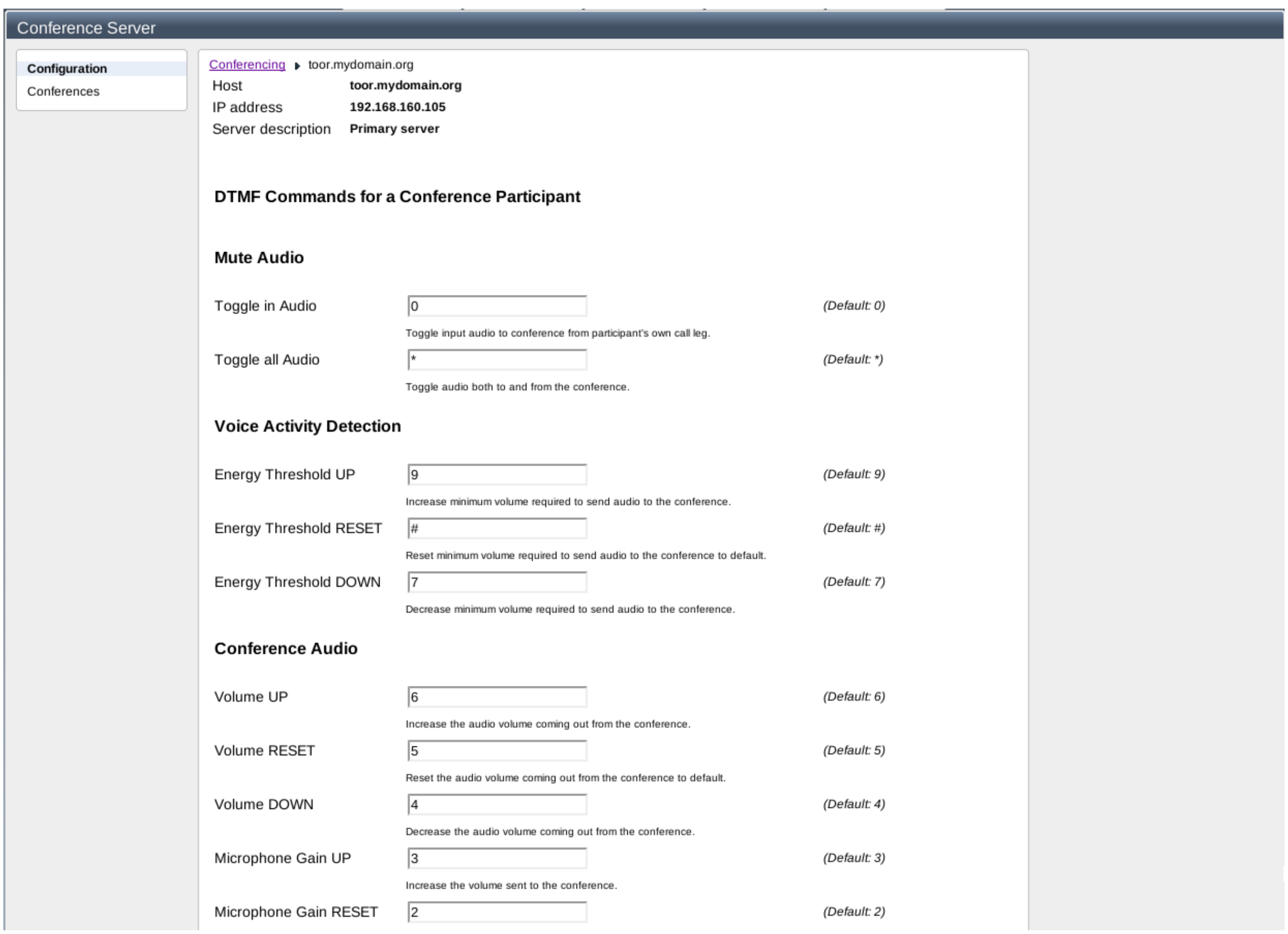
For enabling *Conferencing* role navigate to System > Servers page and choose the location you want Conference Server to run on. On the Server Role tab enable *Conferencing* option. After adding *Conferencing* role you will be warned that FreeSWITCH configuration should be reloaded so the changes to 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.

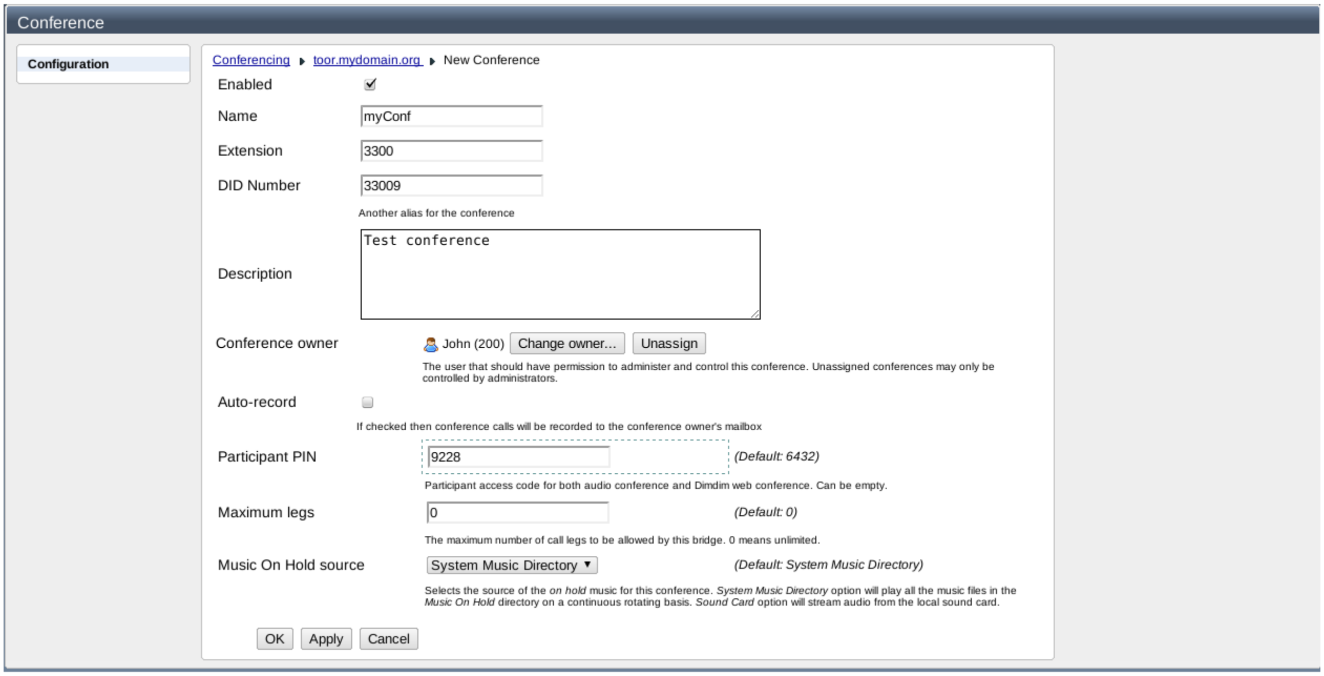


Navigate to *Conference Server* > *Configuration* tab to modify conference server settings. These settings are applied to all conferences on this conference server.



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.



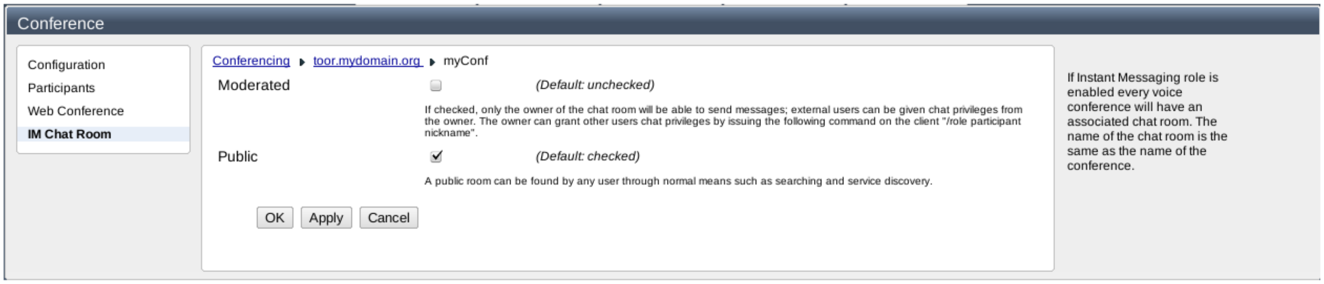
If *Auto-record* is enabled for a conference bridge then the owner of that conference receives the recording of that conference call in a new "conference" mailbox. The conference bridge must have an owner or recording cannot be configured.
Conference call recording must be enabled before the call starts. 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

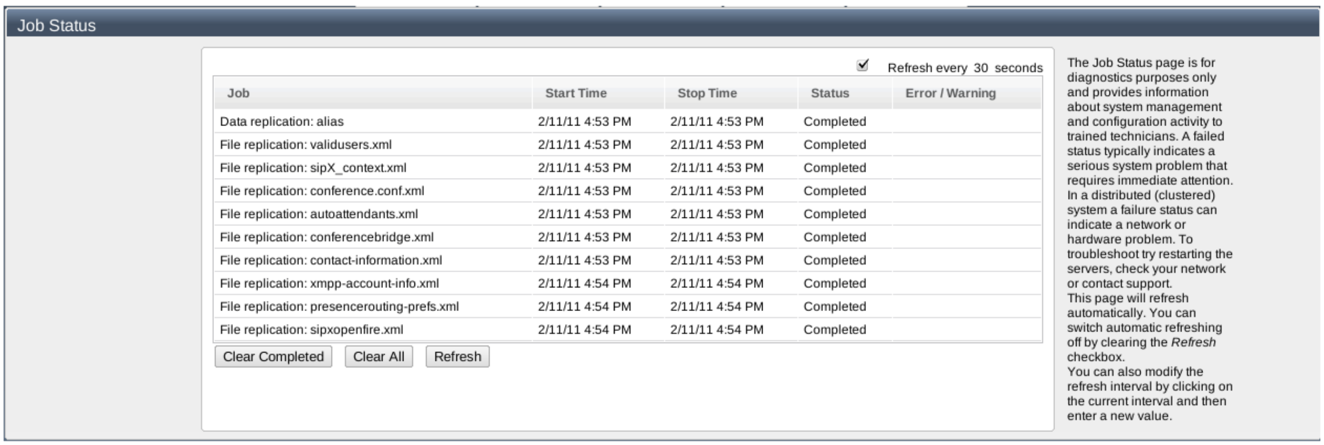
Conference Chat Room

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.



Checking Status

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



FreeSWITCH reload configuration

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