sipXecs 3.6

LDAP Support

LDAP support has made it into the 3.6 release. We allow backend synchronization with an LDAP capable directory and upload the relevant information into Config Server. For performance reasons session authentication will still be done internal to sipX. The implementation should be compatible also with Microsoft AD but it has not yet been tested.

Intercom / Paging

Release 3.6 adds an Intercom capability that in a first phase will support point-to-point intercom using Polycom phone's auto-answer capability.

Dial Plan Templating

In support of international deployments it will be possible to create country or region specific dial plans that can be selected within config server. As an example and in addition to the U.S. dial plan there is now a Swiss dial plan as well as a Polish dial plan. Additional dial plans are easily defined in XML and can be added to sipX as part of a language pack.

Domain Alias

Release 3.6 will allow domain aliasing, which improves deployability of sipX in environments based on Microsoft Windows Server, as well as in cases where sipX needs to be responsible for calls from different domains or IP addresses.

Park Server Enhancements

The sipX Park Server now includes several additional configurable features. There is a time-out value configurable so that after a defined period of time the parked called is transferred back to the person that parked the call. Also, there is a configurable escape key from park. When pressed the call is transferred back to the person that parked the call. Also, it is now possible to configure whether the system allows several calls to be parked on one park orbit (FILA) or not.

Gateway Configuration improvements

The PSTN gateway configuration now allows adding a gateway specific prefix before the number is dialed. This allows the accommodation of different number conversion requirements in case several gateways are associated with a single dial plan rule.

yum based Install & Updates

We plan to eliminate the single file install script and allow installs simply using yum. This is already possible for the Debian build since release 3.0.1.

Support for new Polycom phones

This release adds support for the new Polycom phones such as the SoundPoint IP430. We also updated the plug & play functionality to including the Polycom 2.0 firmware. The sipX Config Server now supports mixed deployment with Polycom phones on the 1.x firmware release and phones already on the new 2.0 firmware release.

FTP Server support for phone management

In addition to the TFTP server sipX now comes standard with a configured FTP server. The FTP server provides access to the same configuration directory used by the TFTP server for phones capable of using FTP instead of TFTP. This is especially convenient for Polycom phones that come factory configured to use FTP.

Support for Hitachi Cable WiFi Phones

This release will support the Hitachi Cable IP 5000 and IP 3000 WiFi phone in Config Server.

SIP Trunk support

We added an option to add a SIP destination as a trunking gateway. This SIP trunking gateway is selectable from the dial plan in the same way a PSTN gateway is selected. A route header field allows the definition of a Session Border Controller (SBC) used to route the call across NAT / Firewall.

Custom Permissions

sipX already offered a flexible mechanism to use different permissions in dial plan rules. We now added the ability to define additional custom permissions that are administrered by the admin and used in the same way built in permissions are.

Caller ID manipulation (CLID / CLIR)

We greatly extended the ability to manipulate and define Caller ID for outbound calls. This allows a much more flexible mapping of User ID to CLID on a per user, per user group as well as on a per gateway basis. In addition we added Caller ID Restriction (CLIR) on a per user, per user group and per gateway basis. The User definition now includes, in addition to User ID and Aliases, a line to define outgoing caller ID. This makes it possible to have e.g. an alpha-numeric or 4 digit local extension as your User ID while still send the full DID number of the users as caller ID to the PSTN.

Updated SNOM phone support

This release updated support for the Snom phones to firmware release 6.2.

Music On Hold (MOH) for Snom phones

Snom phones are the first to support the new IETF standard for Music On Hold (MoH). Release 3.6 provides an IETF standard compliant music source to which the Snom phones can transfer a call when hold is pressed. This provides for a scalable implementation of MoH. A music file can be uploaded from Config Server.

Performance improvement of the media server

We were able to work on some performance improvements for the sipX media server (voice mail subsystem). With that improvement we should now be able to support more virtual media server ports.

Notes

- We missed on the BLF feature in 3.6. We will try and make good on that in release 3.8.