sipXcom 17.04

April 5, 2017

Summary

eZuce is pleased to announce the General Availability Release of sipXcom 17.04.

As with the previous two releases, we're continuing to focus more on fixes and minor improvements in sipXcom as work continues on the next generation of code (see 17.04.docker and 17.08.docker branches). Unite Lite (the new User Portal) gains the ability to disable certain user functionality (this was also included in 16.12.1). The Admin portal also gets links to the sipXcom.org blog as well as links to the Wiki and Forums.

Also as always, thanks to the Dev & QA team at eZuce for their excellent work on this release. Also thanks to IANT for a large number of Yealink and a couple other fixes.

In all 35 issues (enhancements / fixes) are addressed for sipXcom in this beta release.

The next sipXcom release will be 17.08. We're hoping to have at least an Alpha release of sipXcom running with some services 'dockerized'.

Highlights

sipXcom New Features:

- Unite Lite (new user Portal) Admin Control over User Features
- Unite Lite user control over Conference Bridge Entry / Exit tones
- Unite Lite user control over Conference Bridge Voice Announce of Entry / Exit
- Statistics collection for sipXproxy service (phase 1 of a multi-part project)

sipXcom Improvements:

- Improvements to Yealink phone configurations (Thanks IANT!)
- REST API to create/modify a user/user group and set properties
- Improved backup speed by breaking apart backup and compression
- Improved CDR display in Unite Lite for users. Added duration and ability to select Time Zones.

Notes

1. Full Release Notes with installation information are located here: http://wiki.sipxcom.org/display/sipXcom/sipXcom+17.04

Who Should Install?

This release is recommended for all 4.6 and later installations.

Questions

Please post to the sipXcom-users google group if you have questions.

https://groups.google.com/forum/#!forum/sipxcom-users

New Installs

A new ISO is available for 17.04 at: http://download.sipxcom.org/pub/sipXecs/ISO/

Update

To update please edit your /etc/yum.repos.d/sipxecs.repo file and reference the new download server (download.sipxcom.org). The repo should look as follows:

```
[sipXcom]
name=sipXecs software for CentOS $releasever - $basearch
baseurl=http://download.sipxcom.org/pub/sipXecs/17.04/CentOS_$releasever/$basearch
gpgcheck=0
```

To edit this file, login to your sipX server as root and then use either vi or nano (easier).

vi /etc/yum.repos.d/sipxecs.repo

or

nano /etc/yum.repos.d/sipxecs.repo

Once the repo file is modified, run:

yum clean all

yum update

Issues Resolved

	JIRA name	RN Content	Enhancement/Fix/Known Issue	Key words
SIPX-526	Missing DNS record for Proxy-Forwarding to Registrar	Fix for DNS records when a server did not have Registrar enabled. To reproduce issue: Create a cluster with 3 Servers in the following configuration: PBX01 with Proxy and Registrar PBX02 with Proxy and Registrar PBX03 with Proxy With this setup every message, sent to the PBX03-Proxy won't get to one of the existing Registrars. Results in lost calls Config on PBX03-Proxy routes to rr.pbx03.voip.domain.de This name is not generated in DNS Current DNS config is: _sip._tcp.rr IN SRV 30 10 5070 pbx01 _sip._tcp.rr.pbx01 IN SRV 10 10 5070 pbx01 _sip._tcp.rr.pbx01 IN SRV 30 10 5070 pbx02 _sip._tcp.rr IN SRV 30 10 5070 pbx02 _sip._tcp.rr.pbx02 IN SRV 30 10 5070 pbx01 _sip._tcp.rr.pbx02 IN SRV 10 10 5070 pbx02 DNS should be configured like this to make it work: _sip._tcp.rr IN SRV 30 10 5070 pbx01 _sip._tcp.rr.pbx01 IN SRV 10 10 5070 pbx01 _sip._tcp.rr.pbx01 IN SRV 30 10 5070 pbx02 _sip._tcp.rr IN SRV 30 10 5070 pbx02 _sip._tcp.rr.pbx02 IN SRV 30 10 5070 pbx01 _sip._tcp.rr.pbx02 IN SRV 10 10 5070 pbx02 _sip._tcp.rr.pbx03 IN SRV 30 10 5070 pbx01 _sip._tcp.rr.pbx03 IN SRV 30 10 5070 pbx02	Fix	dns
SIPX-539	Yealink Emergency DND Feature	Enhancement to allow provisioning support for Yealink's Emergency DND Feature. From Yealink Provisioning Guide: Specify the authorized numbers when DND is enabled. Parameters: features.dnd.emergency_enable features.dnd.emergency_authorized_number	Enhancement	yealink
SIPX-540	Yealink Call Number Filter	Enhancement to allow provisioning support for Yealink's Call Number Filter. From Yealink Provisioning Guide: Configure the characters the IP phone filters when dialing. Parameters: features.call_num_filter	Enhancement	yealink
SIPX-560	Alert Info External	This is an improvement of Proxy Plugin to set Alert-Info-Header. Some phones (e.g. Yealink) bypass the Proxy if the From-Header do not end with @<sipdomain>. For SIP-Devices that have no ability to add custom headers to a SIP Message (e.g. Patton) it is necessary to scan the from header for a tag (x-sipx-alert-info=external) to set the Alert-Info Header for From-Uris with the SIP Domain inside.	Enhancement	yealink
SIPX-565	Yealink Provisioning of Voice Quality Monitoring	Enhancement to the Yealink provisioning to enable Yealink configuration of RTCP-XR parameters. Yealink can send a Report to a data collector to get information about the quality of the last call. Event is: vq-rtcpxr On activation the Phone sends a Publish Message with the Report to a service you can configure	Enhancement	yealink
SIPX-569	Syslog only receiving on UDP 514	In older sipXcom releases, the default Syslog transport setting for Polycom phone groups was UDP. In release 16.02, the default setting syslog transport setting is TCP, which triggers the phones to send log file information to Sipxcom using TCP transport port 1468. This is a fix to set syslog transport back to UDP which is preferred over TCP to preserve TCP sockets.	Fix	sipxconfig
SIPX-572	Jitsi Provisioning DND	An enhancement to Jitsi provisioning with the following parameter: net.java.sip.communicator.impl.protocol.RejectIncomingCallsWhenDnD For sipXcom/Uniteme it is necessary to set this parameter has to be set to "true", because the Proxy/Registrar could not handle the DND itself.	Enhancement	jitsi

SIPX-577	DNS NAPTR prefer TCP	Enhancement to adjust the DNS NAPTR records to have client prefer TCP vs. UDP since TCP is the prefered protocol for SipXcom/UniteMe Current DNS NAPTR configures them to be equal voip.domain.de. IN NAPTR 2 0 "s" "SIP+D2U" "" _sip._udp voip.domain.de. IN NAPTR 2 0 "s" "SIP+D2T" "" _sip._tcp This should be changed to the following so if some devices use auto configuration, TCP will be chosen. voip.domain.de. IN NAPTR 2 0 "s" "SIP+D2U" "" _sip._udp voip.domain.de. IN NAPTR 1 0 "s" "SIP+D2T" "" _sip._tcp	Fix	dns
SIPX-578	Stats collecting submodule for Proxy	Enhancement to sipXproxy that is required to implement internal metrics collecting submodule for proxy and output collected metrics into file with pre-configured interval. In the first approach Proxy shall to have statistics file with name=value content (like "ProxyQueueSize=123"). Single line for single metric. File should be updated every StatsUpdateInterval seconds configuration parameter. Default value will be 15 seconds. In future if we will need we can add this to webui configuration.	Enhancement	sipxproxy
SIPX-581	Update Admin GUI w/Blog Posts and Links	This is an enhancement to the Admin GUI to receive blog post updates from sipxcom.org web site in the Admin GUI as well as link to important sipxcom.org URLs. Add RSS feed capabilities to Admin GUI. The feed should come from http://sipxcom.org/feed/ Add the important links on the right as shown in Mockup: Downloads (ISO's & RPM's): sipXcom ISO Images and RPM Repositories (link the text after the ':' to http://wiki.sipxcom.org/display/sipXcom/sipXcom+ISO+Images+and+RPM+Repositories) JIRA Issue Tracker: http://jira.sipxcom.org sipXcom on Github: https://github.com/sipXcom sipXcom User's Mailing List: https://groups.google.com/d/forum/sipxcom-users sipXcom Developers Mailing List: https://groups.google.com/d/forum/sipxcom-dev Paid Version & Optional Features: https://www.ezuce.com At top of the page darken icons to 50% grey. Add Paid Support, change 'Help' to 'Docs'. Paid Support should link to: http://ezuce.com/products-solutions/procare-sipxcom-support	Enhancement	sipxconfig
SIPX-585	Yealink Resource List Subscription	Fix for Yealink Provisioning to check if lines on the phone have BLFs. Current Issue: If user has no BLFs, Yealink start to spam to proxy with resource list subscriptions and if you have enough Yealink to proxy stops working.	Fix	yealink
SIPX-587	Rotate proxy_stats.json file - proxy stats	This is an enhancement in support of the proxy statistics enhancement. The file where proxy stats are collected should be include in the Logrotate mechanism.	Enhancement	sipxconfig sipxproxy
SIPX-600	Custom settings enhancement for Yealink provisioning	Enhancement to add the ability to add custom settings to Yealink configuration. This is a good feature with polycom provisioning which is now available for Yealink provisioning. Settings which are currently missing in GUI could be set via this custom config.	Enhancement	yealink
UC-3683	REST API to create/modify a user/user group and set properties	Enhancement to allow for user management through a rest API. Currently we have SOAP support for user creation, but there is no support on user settings (like user called id for example) Use the new REST Api engine based on Apache CXF to create such api. The settings management is generally handled by current existing REST support	Enhancement	sipxconfig
UC-4290	Enhancement request for Unite Web CDR duration	An end user would like to see a call Duration column (or, alternatively, the End Date/Time) in the end user Call Detail History in Unite Web.	Enhancement	Unite Web
UC-4296	Web UI Search searches Voicemail PIN Token	Fixed an issue where Voicemail PIN was included as a searchable field when searching for an extension in the system. To reproduce the issue: Example: You search for "96" The search will show you every user starting with 96 and user with a voicemail pin token that starts with 96	Fix	sipxconfig
UC-4297	Increase elasticsearch user limits	Fix for the Elasticsearch user file limits which are set too low for larger systems. We should have an entry in /etc/security/limits.d/ for user elasticsearch to increase currently number of open files and proccesses that user can spawn per this description: https://www.elastic.co/guide/en/elasticsearch/guide/current/_file_descriptors_and_mmap.html	Fix	sipxconfig

UC-4307	Add new settings for User Portal configuration	This is an enhancement in support of Unite Web work to allow an Administrator to be able to control the enabling and disabling of features of the User Portal by User and by User Group. This work is in support of the 4 new features. Feature 1 - Disable Dial Pad and Search icons These options would be configurable in the Uniteme Administration GUI in Users -> Users -> “username” or Users -> User Groups -> “user group name”. In the left side menu there would be a new menu item called User Portal. In the User Portal configuration page there would be the following configuration options for this feature: Enable Dial Pad Icon Type: Checkbox Default: Enabled Enable Search Icon Type: Checkbox Default: Enabled Feature 2 - Disable Contact Click to Call and Chat In the User Portal configuration page (in Users -> Users -> “username” and Users -> User Group -> “user group name”) there would be the following configuration options for this feature: Enable Contact Click to Call Type: Checkbox Default: Enabled Enable Contact Click to Chat Type: Checkbox Default: Enabled Feature 3 - Disable Click to Call from Conf Bridge In the User Portal configuration page (in Users -> Users -> “username” and Users -> User Group -> “user group name”) there would be the following configuration options for this feature: Enable Conference Bridge Click to Call Type: Checkbox Default: Enabled Feature 4 - Disable Unite Web menu items In the Unite Web configuration page (in Users -> Users -> “username” and Users -> User Group -> “user group name”) there would be the following configuration options for this feature: Enable Activity List Type: Checkbox Default: Enabled Enable Contacts Type: Checkbox Default: Enabled Enable Group Chats Type: Checkbox Default: Enabled Enable Conference Bridge Type: Checkbox Default: Enabled Enable Voicemails Type: Checkbox Default: Enabled Enable My Profile Type: Checkbox Default: Enabled Enable Call History Type: Checkbox Default: Enabled Enable Settings Type: Checkbox Default: Enabled Enable Settings Personal Attendant Type: Checkbox Default: Enabled Enable Settings Call Forwarding Type: Checkbox Default: Enabled Enable Settings Speed Dials Type: Checkbox Default: Enabled Enable Settings User Settings Type: Checkbox Default: Enabled Enable Settings User Settings Change Password Type: Checkbox Default: Enabled Enable Settings User Settings Voicemail PIN Type: Checkbox Default: Enabled Enable Settings User Settings Announcement Type: Checkbox Default: Enabled Enable Settings User Settings eMail Type: Checkbox Default: Enabled Enable Settings User Settings Attach audio Type: Checkbox Default: Enabled Enable Settings User Settings Alternate eMail Type: Checkbox Default: Enabled Enable Settings User Settings Alternate Attach audio Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Room Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Enabled Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Name Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Moderator PIN Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Participant PIN Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Max. members Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Quickstart Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Auto-record Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Moderated Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Public Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Entry Tone Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Exit Tone Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Voice Announce Entry Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Voice Announce Exit Type: Checkbox Default: Enabled Enable Settings User Settings MoH Audio source Type: Checkbox Default: Enabled Enable Settings User Settings MoH Personal MoH Type: Checkbox Default: Enabled Enable Settings User Settings MoH Files Type: Checkbox Default: Enabled Enable Settings User Settings MoH Audio file Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Conference enter Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Conference exit Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Voicemail begin Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Voicemail end Type: Checkbox Default: Enabled Enable Settings Sound Notifications Type: Checkbox Default: Enabled	Enhancement	Unite Web sipxconfig
UC-4311	Polycom SoundPoint IP 650 and 560 background images are not working	Fixed an issue with background images on Polycom SPIP 650 and 560 phones.	Fix	Polycom sipxconfig
UC-4313	Conference settings REST API to include new parameters	This is a Rest API enhancement to support Unite Web and Unite Lite enhancements. We need to add new parameters: Play Entry Tone Type: Checkbox Play Exit Tone Type: Checkbox Play Voice Announce Entry Type: Checkbox Default: Enabled Play Voice Announce Exit Type: Checkbox Default: Enabled to existing REST API method: 'GET' and 'PUT' /my/conferences/"conference name"	Enhancement	Unite Web conferencing sipxconfig
UC-4324	Unite Web call history improvements	Enhanced the Unite Web Call History. These are some of the changes that need to be implemented: 1.show timezone drop down as in old style portal , but defaulting to show the timezone of the user's PC (old style portal relies on what is set under User->Time Zone , but we don't need that) 2.Show call history entries by default, based on the default selection. As soon as you go on the call history page, without hitting Apply. 3.Reverse Apply button location with To/from box 4.When using To/From box, results should show up if you both hit Enter or click Apply 5.Fix time format to 00:00:00 in Start/Stop columns instead of 00:0:00 6.Sorting of columns in results, if it's easy	Enhancement	Unite Web
UC-4328	REST API to manage user properties for user portal	An Enhancement to support the work for Unite Web work. Provide set of rest apis that a regular user (USER_ROLE) can access, in order to update logged in user properties/settings	Enhancement	Unite Web sipxconfig
UC-4349	Separate backup tar & compression	An administrator would like to speed up the backup process. For backup scripts, separately specify the tar and the gzip and set the compression level to 1. first: tar -cvf then: gzip -1 This will increase speed but also increase the backup size (but probably not significantly. More concerned with backup window on larger systems than backup size on smaller systems.	Enhancement	backup

UC-4363	Jitsi provisioning automatic display name	An administrator would like such that if nothing is entered for the Display name, that the provisioning fills this with name and surname stored in the user profile by itself (same behavior as Polycom Plugin). The current Version of Jitsi Provisioning can configure the line display name (Lines > SIP > Displayname), it's just not utilizing the name and surname from the user profile.	Enhancement	jitsi
UC-4376	Disable MWI subscription if Voicemail permissions are disabled	Fixed an issue where Polycom phones are trying aggressively to subscribe for Event: message-summary. To those Subscribes proxy returns 403 but server is flooded by SUBSCRIBES by the users that have Voicemail permissions are disabled.	Fix	Polycom
UW-355	On Safari and IOS, the scrolling is not working properly	Fixed an issue under the Profile tab that was reproduced after the user tried to edit some fields and then scroll down and save it. After saving, when the user would scroll up to the beginning the user would notice that the scroll is working very hard and sometimes is not working at all (because the user can accidentally scroll from the margin and as result entire web page will be scrolled). If the user scrolls up or down from the middle of the screen then it will work ok without problems.	Fix	Unite Web
UW-367	Mute microphone and mute speaker (conference controls) require multiple taps/clicks	Fixed an issue with Conference room controls that required the user to click multiple times to mute microphone or mute speaker. Reproduce Issue: 1. Using a conference room owner, login into unite web 2. Join the conference with the owner, via his phone by dialing the conf room number (you can't drag and drop contacts into conference in android) - linked jira, and even if you could, you would not be able to answer the incoming invite call - linked jira 3. From unite web, switch to conference bridge The conf participants will show up here. You have the ability to control microphone, speaker, end call. 4. Tap the microphone icon to enable mute, and then tap it again to unmute. Issue: while you can tap to mute, it takes 2-3-4 taps to unmute, and then 2-3 more to mute again if you want to. The same is valid for the mute speaker button. End call button works fine. Workaround is to wait 5-10 seconds between each mute/unmute - if that's a workaround Reproduces on all platforms	Fix	Unite Web
UW-379	Users can't see defined group speed dials	Fixed an issue where even though the checkbox is set, the user will not see which speed dials are defined for this group. Steps to reproduce: 1. Define an user group and add a speed dial to this user group 2. Using an user which is part of this group, login into UW and go under Settings->Speed dials 3. Make sure the Only use group speed dials is checked If the user goes to old style portal they will see them.	Fix	Unite Web
UW-383	Call history entries time difference	Fixed an issue with the UW call history. Issue Description: System has current time 14:00. Users phone has current time 14:00. User calls some other user and checks the call history in unite web. Issue: Call entry shows time: 12:00 - 2 hour difference. Reviewing the System CDR entries , the call entry shows time 14:00 Reviewing the old user portal the call entry shows time 14:00	Fix	Unite Web
UW-384	Disable Dial Pad and Search icons	An administrator would like to make the Dial Pad and Search icons unavailable to a user or a group of users. These options would be configurable in the Uniteme Administration GUI in Users -> Users -> “username” or Users -> User Groups -> “user group name”. In the left side menu there would be a new menu item called Unite Web. In the Unite Web configuration page there would be the following configuration options for this feature: Enable Dial Pad Icon Type: Checkbox Default: Enabled Enable Search Icon Type: Checkbox Default: Enabled	Enhancement	Unite Web
UW-385	Disable Contact Click to Call and Chat	An administrator would like to disable the ability of a user to use click to call on a contact and also disable the ability of a user to use click to chat. In the Contacts menu, if a user clicks on a contact’s avatar, information is displayed about that user. After clicking on the Avatar, information about the contact is displayed. The first part of this feature request is to be able to disable the click to call capability. The button for this is highlighted. The information should remain (username and extension) but should not allow for click to call. The second part of this feature request is to be able to disable the click to chat functionality. Click to chat works when a user clicks on the user name in the contacts list. Clicking on the name would normally display a chat area on the right frame. In the Unite Web configuration page (in Users -> Users -> “username” and Users -> User Group -> “user group name”) there would be the following configuration options for this feature: Enable Contact Click to Call Type: Checkbox Default: Enabled Enable Contact Click to Chat Type: Checkbox Default: Enabled	Enhancement	Unite Web
UW-386	Disable Click to Call from Conf Bridge	An Administrator would like to be able to disable the click to call feature from the user’s conference bridge management screen. Clicking on the highlighted button would normally bring up a dial box where a user can enter an extension or phone number to dial. In the Unite Web configuration page (in Users -> Users -> “username” and Users -> User Group -> “user group name”) there would be the following configuration options for this feature: Enable Conference Bridge Click to Call Type: Checkbox Default: Enabled	Enhancement	Unite Web

UW-387	Disable Unite Web menu items	An Administrator would like to be able to disable specific menu items in Unite Web and Unite Web Lite. The menu list is accessed by clicking on the menu button in the upper left. The administrator would like to be able to control which menu items a user or user group has access to. The administrator would also like to control what settings a user can change. In the Unite Web configuration page (in Users -> Users -> “username” and Users -> User Group -> “user group name”) there would be the following configuration options for this feature: Enable Activity List Type: Checkbox Default: Enabled Enable Contacts Type: Checkbox Default: Enabled Enable Group Chats Type: Checkbox Default: Enabled Enable Conference Bridge Type: Checkbox Default: Enabled Enable Voicemails Type: Checkbox Default: Enabled Enable My Profile Type: Checkbox Default: Enabled Enable Call History Type: Checkbox Default: Enabled Enable Settings Type: Checkbox Default: Enabled Enable Settings Personal Attendant Type: Checkbox Default: Enabled Enable Settings Call Forwarding Type: Checkbox Default: Enabled Enable Settings Speed Dials Type: Checkbox Default: Enabled Enable Settings User Settings Type: Checkbox Default: Enabled Enable Settings User Settings Change Password Type: Checkbox Default: Enabled Enable Settings User Settings Voicemail PIN Type: Checkbox Default: Enabled Enable Settings User Settings Announcement Type: Checkbox Default: Enabled Enable Settings User Settings eMail Type: Checkbox Default: Enabled Enable Settings User Settings Attach audio Type: Checkbox Default: Enabled Enable Settings User Settings Alternate eMail Type: Checkbox Default: Enabled Enable Settings User Settings Alternate Attach audio Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Room Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Enabled Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Name Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Moderator PIN Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Participant PIN Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Max. members Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Quickstart Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Auto-record Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Moderated Type: Checkbox Default: Enabled Enable Settings User Settings Conference Bridge Public Type: Checkbox Default: Enabled Enable Settings User Settings MoH Audio source Type: Checkbox Default: Enabled Enable Settings User Settings MoH Personal MoH Type: Checkbox Default: Enabled Enable Settings User Settings MoH Files Type: Checkbox Default: Enabled Enable Settings User Settings MoH Audio file Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Conference enter Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Conference exit Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Voicemail begin Type: Checkbox Default: Enabled Enable Settings User Settings MyBuddy Voicemail end Type: Checkbox Default: Enabled Enable Settings Sound Notifications Type: Checkbox Default: Enabled See Feature 4 in https://docs.google.com/document/d/1wMp1RyFTJiKRNyWWIse1eP_216VaSjwVpBJHso1TPXI/edit# Is this document something that users can see? If not, we shouldn't reference it here but rather somehow port the content of it to the release notes.	Enhancement	Unite Web
UW-388	Add conf bridge welcome tones options + ability to disable /enable them as user	The ability to enable or disable entry / exit tones and also for voice announce was added in 16.12. Add ability for user to enable / disable conf bridge welcome tones from Unite Lite and Web. Add ability for user to enable / disable user announce on entry / exit.	Enhancement	Unite Web
UW-394	There are two download icons in latest Chrome	With the latest version of Chrome, Unite Web users have two download icons for each Voicemail on the Voicemail page	Fix	Unite Web