RTS

SOFTWARE REVISION HISTORY

for

VLink Intercom System

VVM/VSA v5.6.5 - 05/11/2020 – Public Release

SUMMARY

- Corrected various System Administration issues

- Corrected various WebRTC Control Panel issues

- Corrected intermittent issue with server hang on shutdown.

- Added support for interfacing SIP Client to AWS Chime.

- Added support for specifying the SIP Transport Protocol as either TCP or UDP.

- Corrected various potential server crash issues.

- Corrected WebRTC Control Panel for iOS to resolve login issues due to changes in iOS 13.4.1

- Corrected VLink Trunking when used with TM10Ks to not prevent sending corrupted Alpha based to the TM is System Number 32 is used.

- Corrected VLink Trunking issue with potential server hang with extremely large trunked system deployments

- Added VLink Trunking support for Fixed Groups to be shared as RTS Special Lists

CHANGES/FIXES

- VVM: Added preliminary support to VLink Trunking to allow for sharing of VLink Fixed Groups as RTS Special Lists. Not yet implemented is the Call Notification tally when the Fixed Group is activated from an RTS KP.

- VVM: Corrected the WebRTC VCP to restore the client configuration settings to the base template upon login if templated.

- VVM: Changed VLink Trunking, in the case of unassigned RTS Master Controller IP or a general failure to communicate, to shorten the sleep time from 5s to 1s as this would adversely affect the system shutdown time.

- VVM: Changed VLink Trunking when not licensed, to add a sleep time of 1s to the thread to reduce CPU loading.

- VVM: Corrected a server crash that occurred when uploading or downloading SSL Certificate files greater then 4K.

- VVM: Added activity log messages for licensing to show the number of Party Line and Fixed Group licenses.

- VVM: Correct the handling of SIP Trunks to not start the threads for any disabled SIP clients.

- VSA: Corrected a problem with the System Administration login where, when attempting to login with client credentials that had no administrative privelges, the login was rejected with "suspension of login privileges" instead of "no administrative priveleges".

- VVM: Corrected a server crash when using the browser and entering a login name or login password greater than the maximum alloted number of characters of 50 and 20 characters respectively.

- VVM: Changed the default two user license to include one PL and one FG.

- VVM: Updated the VLink Trunking default System Configuration file to turn off AGC and Silence Suppression for the RTS Trunks to support Trunk Testing by default.

- VVM: Updated the VLink Trunking default System Configuration file to enable all "Scroll Inhibits" for the RTS Trunks so that they can never be programmed to any panel either local or remote.

- VVM: Corrected the messages for the 'System restart' and 'SIP Server restart'.

- VVM: Corrected a VLink Trunking problem where adding RTS Alphas might result in an server hang if exceeding the maximum number of Alphas supported.

- VVM: Corrected a potential but unlikely server crash that could occur on a user selected WebRTC Client logout.

- VVM: Corrected a problem where the auto-generated internal SIP clients were unable to send SIP Invites to their external SIP client counter parts.

- VVM: Changed the debug message that indicates packet loss from an Error message to an Information message so that it does not appear in the debug log without specifically enabling the Audio Rx Information messages.

- VVM: Changed the Virtual Matrix installer to install the Windows Error Reporting registry keys necessary to enable the crash dumps for both the Virtual Matrix application and server.

- VVM/VCP/VDI: Corrected the installers to not reboot the PC without any prompting as installing Microsoft compoments.

- VVM: Corrected a problem where the WebSocket thread would intermittently but frequently hang on shutdown.

- VVM: Corrected a potential server crash issue introduced in v5.5 when changing "the Activity Log messages displayed on the web client login".

- VVM: Increased the maximum supported SIP User Name and SIP Host Name from 40 characters to 60 characters to accommodate services like AWS Chime that require such.

- VVM: Added support for a SIP Transport Protocol which can configured for each SIP client to be either UDP (default) or TCP.

- VVM: Corrected a server crashing problem when making outbound calls using very long SIP Host Names.

- VVM: Changed the dial sequence validation to not remove the '+' character as required by AWS Chime.

- VVM: Corrected a problem with the Control Panel for WebRTC where the client was unable to log in using an iOS device after updating to iOS v13.4.1. In this iOS update, Apple discontinued support for TLS v1.1 The problem was resolved by instantiating support for TLSv1.2 on the WebSocket.

- VVM: Corrected a critical problem where the Virtual Matrix could crash either when restarting or terminating due to a memory corruption issue which occurred during initial parsing of the configuration file.

- VVM: Changed the System Settings API to check for changes to any of the SIP configuration values in order to initiate a restart of the SIP server if changed.

- VVM: Added a temporary bypass of Domain Authentication for the WebRTC Control Panel which currently cannot be used for clients configured for Domain Authentication.

- VVM: Corrected the 2 client default demo license which is activated on initial installation to include the Control Panel for WebRTC for both RTS and Intracom builds and to include the WebRTC Control Panel Video for the Intracom build.

- VVM: Corrected a problem with VLink Trunking where it was assumed that the local matrix index for the VLink system could internally always be assigned to matrix index 31. While this erroneous assumption did not cause issues with TM2Ks, it did cause issues with the TM10Ks because it was possible for an RTS Adam frame to be assigned to matrix index 31. In this situation, the VLink Virtual Matrix would correctly share its alphas with the TM10K on initial power up, however, when the TM10K subsequently shared its alphas for all RTS Adam frames, the alphas for the RTS Adam frame assigned to matrix index 31 would overwrite the staged VLink alphas. Since the VLink alphas has already been shared with the TM10K, this problem was relatively benign in that no operational issues were immediately apparent. However, if the VLink Virtual Matrix subsequently lost communication with the TM10K due to a network disruption, then upon restoring the communication, VLink would again share it staged alphas with the TM10K, but the alphas at this point, due to the previous noted overwrite issue, would in fact be the alphas associated with the RTS Adam frame at matrix index 31.

- VVM: Changed the handling of the VCP for WebRTC clients such that the audio sampling rate for all clients is now based on the system audio sampling rate.

- VVM: Corrected a potential server crash due to a string handling method throwing an exception when the source string exceeded the buffer size of the target string.

- VVM: Changed the VLink Trunking error message for "Incoming Call Tally not supported" from an error message to a warning message.

- VVM: Added licensing flag for the WebRTC Control Panel for WebRTC.

- VVM: Added licensing flag for the WebRTC Control Panel Video.

- VVM: Temporarily excluded the SIP G722 codecs pending investigation of several customer related audio issues that occurred when the G722 codec was used.

- VVM: Changed the handling of the RTS Alphas to ensure that the Function Index for any Label Type does not exceed the maximum value supported by the Trunk Master.

- VVM: Corrected a problem with the WebRTC Control Panel implementation where, if a connected client was disabled, that client would not be able to log in again until after a server restart.

VVM/VSA v5.5.2 - 03/25/2020 – Public Release

SUMMARY

- Corrected potential server crash issues.

- Corrected various System Administration issues

- Changed the SSL handling to install self-signed certificate automatically if no certificate exists as a certificate is required for the WebRTC Control Panel.

- Corrected SIP call transfer issues.

- Added turnkey support for WebRTC Control Panel video.

- Correct potential VLink TIF communication issues.

CHANGES/ISSUE

- VVM: Corrected the URL used by the installer to automatically launch the VSA after initial installation to remove the '#/ which did not work in conjunctions with the new HTTP to HTTPS redirection mechanism.

- VVM: Corrected a problem where the Latch Disable was only being output for Intracom Clients, Fixed Groups and Party Lines. As a result, it was not possible to configure an RTS Alpha to be latch disabled.

- VVM: Corrected a problem with the TIF Dial Input Timeout where the value for the timeout would never be saved if no Dial Plans were configured.

- VVM: Corrected HTTP to HTTPS redirect to account for the "Host" header string containing a specific HTTP port number.

- VVM: Corrected a problem where after creating a new client with no password, the client might in fact be created with an existing password of previous client.

- VVM: Added support for server control of the Video Application server which runs as a separate process running

- VVM: Corrected the long-standing browser file caching issue by changing all HTML files to include a "no-caching" header. Additionally the HTTP->HTTPS redirection which was previously done in the browser code is now correctly handled by the server by returning a '301 Moved Permanently' response, along with the 'Location' header with the redirect URL.

- VVM: Changed the handling of SSL certificates such that upon startup, the server will check for the presence of an SSL Certificate and generate a self-signed SSL Certificate if it does not exist.

- VVM: Added configuration items to the System Settings for the Video Media Server hostname and ports as well as the Vida Application Server port, to be used by the WebRTC Control Panel as well as the Video Application Server allowing dynamic user configuration.

- VVM: Corrected an intermittent problem with the TIF implementation, primarily with call transfers where, with a newly connected call, there would be no audio heard from the caller. The problem was caused by an error in the handling of dynamic "in-dialog" changes to the caller's media server. The problem occurred when receiving and correctly processing the request to change to the caller's media server at the same time that an RTP packet was received from the prior media server. By design, when changing media servers, the old media server address is blacklisted and all not yet processed RTP packets are flushed. Erroneously, the RTP packets were in fact not flushed and a such, the RTP packet that was received from the prior media server was subsequently processed which resulted in reversing the change to the media server address causing all RTP packets from the new media server to never be received.

- VVM: Changed the Configuration File to allow duplicate login names which can be useful in some SIP implementations.

- VVM: Corrected a problem where a label was not being properly deleted.

- VVM: Corrected a problem when deleting labels where the Activity Log and Debug Log messages would show an invalid name for the label that was deleted.

- VVM: Change the VVM installer to install the NodeJS package as part of the installation as it is required for the Video Application server.

- VVM: Corrected a problem when using the native VCP/VDI clients where valid SIP credentials would not be rejected by the server as an incorrect client type. Instead the login would be accepted leading to a possible server crash.

- VVM: Change the maximum login name length supported on the server to 50 characters to support a customer required SIP User Name + Authentication ID that was longer than the previous maximum of 20 characters.

- VVM: Corrected the IP Web Server WS and WSS port assignments to not be hardcoded to one greater than the HTTP and HTTPS ports, respectively.

VVM/VSA v5.5.0 - 02/16/2020 – Public Release

SUMMARY

- Formalized the WebRTC Control Panel as an alternative to the native OS specific Control Panels.

- Increase default system sample rate to 48 KHz.

- Removed support for 32 bit OS installations.

- Corrected various potential server crash issues.

- Corrected various System Administration issues

- Corrected potential VLink TIF alphas display issues.

- Corrected VLink Trunking / VLink TIF issue with server hang on shutdown.

- Corrected issue with the OPUS codec as a supported SIP Codec.

- Corrected a configuration file issue as a result of an out of disk space condition.

- Corrected problem with the display of 8-character unicode VLink Trunking alphas.

CHANGES/FIXES

- VVM: Increased the maximum IP Address string length to accommodate the iPv6 addresses which are 39 characters maximum.

- VVM: Updated a customer specific iPak audio routing customizations, used for election coverages, to include a new iPak. Also updated the naming of existing iPaks to reflect the current operator names.

- VVM: Corrected a problem with the RUDP protocol used between an intercom and the Trunk Master when using Ethernet links with high latency, in particular, with an Ethernet connection carried over a satellite link (with a round-trip delay of approximately 450 ms).

- VVM: Corrected a problem with self-signed certificate generation where the SAN list entries for IPs were erroneously created with the "DNS" prefix instead of "IP" prefix.

- VVM: Corrected a problem with the TIF implementation where with the alphas sent by the RTS MC were not being properly parsed.

- VVM: Changed the VLink installer to add an Installation Type dialog so as to eliminate the need for separate VLink Trunking and VLink TIF installers. The dialog will only be displayed if there is no existing configuration installed.

- VVM: Added 8 RTS Trunk ports to the default VLink Trunking configuration.

- VVM: Changed the default audio mix sample rate to 48 KHz.

- VVM: Added an error message when attempting to install on a 32-bit OS as this is no longer supported.

- VVM: Removed all conditional code which supported the 32-bit OS installation.

- VVM: Restructured the SSL API and supporting files to stream line the process for generating both self-signed and CA signed certificates.

- VVM: Corrected the SSL API to indicate a 422 Unprocessable Entity on a processing error rather than a 500 Internal Server Error.

- VVM: Corrected a potentially critical coding issue where the incorrect target buffer size was used when extracting the SIP host address from the AOR.

- VVM: Corrected a VLink Trunk problem where the server would hang on shutdown.

- VVM: Corrected several issue with the Pacific Interactive Tally implementation. Correct a problem where it was possible that an Alpha Update messages might be sent in the same transmission as a Line Status or Audio Level message, resulting in the Alpha update message being lost. Corrected a problem with the Alpha update message where if multiple updates where available at the exact same time, each update would be sent as a separate message rather than as a single message with multiple Alpha updates, resulting in all but the first update being lost. Corrected the Audio Level message to not send audio level updates for disabled clients as this was unnecessary and an inefficient use of bandwidth.

- VVM: Added support for detecting a configuration change to the Selector Volume Gains in order to update the volume gains dynamically.

- VVM: Change the Activity Log messages displayed on the web client login and logout when using an Anonymous Login to show both the configured and user provided client login name.

- VVM: Changed the 'Authentication' API to remove the obsolete 'ACCESS_TOKEN' from the response.

- VVM: Corrected a problem with the Client Positioning where the connected clients were not properly being displayed.

- VVM: Corrected a problem where both the HTTP and HTTPS services might fail to start when updating from a prior version and retaining the existing System Configuration file. The problem was due to some older System Configuration files having both the HTTP and HTTPS ports set to zero, which was the case in early webserver development.

- VVM: Corrected a problem where the System Configuration file would be lost if there was insufficient disk space remaining to save the file. If the insufficient disk space problem remained unresolved, the system would continue to function properly however upon the next server restart, the server would fail to start due to the missing file. The problem was resolved by first writing System Configuration file to a temporary file and only if that output succeeded was the active System Configuration file deleted and temporary file made to be the active file.

- VVM: Corrected a problem where a SIP connection requiring the OPUS codec was being rejected. The problem was due a missing attribute item required for the OPUS codec which specifies the number of channels.

- VVM: Corrected a problem with the VLink TIF implementation where some RTS originated alphas updates were being lost. The problem was due to an error in the parsing of the 'Alphas' message when it contained more than one alpha update.

- VVM: Added safeguards to ensure that no more than the maximum RTS Master Controller supported 100 phone lines would ever be processed. This was necessary to allow support of more than 100 lines in the configuration as was required by a customer.   All lines in excess of 100 will not have their status conveyed to or controlled by the RTS Master Controller and must therefore be configured for auto-answer.

- VVM: Corrected a problem with Vlink Trunking where the font size of the 8-character Unicode alphas was such that the alpha could not be fully displayed on the selector without cropping. The problem was a result of the Unicode characters received from the Trunk Master using the Full-Width Unicode character set. The problem was resolved by converting the alpha from the Full-Width Unicode character set to the Standard Unicode character set.

- VVM: Corrected a problem with the Virtual Matrix when using a large number channels where the CPU utilization was higher than expected.

- VVM: Corrected an issue with Command File processing which was exposed by a newer version of Visual Studio where an erroneous NULL character in the scanf string prevented the proper parsing of the command files.

- VVM: Corrected a problem where the server could crash when the 'SIP Domain Name' assigned did not represent the SIP server itself. The crash occurred immediately after receiving the response to the DNS query which included a long list of IPs.

- VVM: Changed the TIF API to return an "ERROR" in the Line State to indicate either a loss of communication with the RTS Master Controller or the associated 4-Wire port is not connected.

- VVM: Corrected a problem where the server would hang on shutdown when terminating the SIP Client threads.

- VVM: Corrected a problem with the Web Server startup where the server would fail to start if the private key no longer matched the SLL certificate . The problem was resolved by catching the specific exception and deleting the certificate so that at least the HTTP would start allowing the HTTPS issue to be resolved through the System Administration user interface.

- VVM: Corrected a problem where some video streaming URL definitions were mysteriously disappearing. The problem occurred as a result of the URL being longer than the maximum allocated length of the item data string in the Configuration Data File.

- VVM: Corrected a problem with HTTPS support by changing the Web Server thread to use LoadLibrary instead of GetModuleHandle for ssleay32.dll. The problem was due to GetModuleHandle requiring the library to be already loaded which it was not. Previously this library was already loaded by the Web Tool Kit library, however the newer version of the Web Tool Kit no longer uses this library.

- INSTALLER: Updated the installers to utilize the newer Microsoft redistributables as required by the update to the Platform Toolset .

- VVM: Replaced the browser client audio transport mechanism, which used a Web Socket implementation, with a native WebRTC implementation.

- VVM: Corrected a problem with the Video Streaming API where the streaming state would become orphaned if the browser was closed while video streaming was active. The problem was resolved by always clearing the streaming state upon client logout.

- VVM: Corrected a problem with the playback mechanism for Pre-Recorded Audio Sources where playback would be garbled when a new source was queued while an existing queue was already partially processed.

- VVM: Corrected the Play Sound API to support successive calls to the API allowing multiple WAV files to be queued as previously there was only one Pre-Recorded Sound class allocated per client.

- VVM: Added a Call Notification State API to provide the ability to detect Call Notification activations and deactivations.

- VVM: Corrected a critical problem where the generation of browser client authentication tokens would intermittently crash the server.

- VVM: Corrected a WAV file header parsing issue which resulted in a failure to playback some files.

- VVM: Corrected a problem where the Activity Log message for a VCOM Client logout was not displaying the correct client name when using an anonymous login.

- VVM/VSA/VCP/VDI: Updated all projects to use the Platform Toolset which effectively removes support for Windows XP.

VVM/VSA v5.2.3 - 04/28/2019 – Public Release

SUMMARY

- Corrected SIP Client connection failures after failover to Secondary

- Enhanced SIP call handling to support dynamic IP/Port changes

- Included OPUS codec as a support SIP Codec.

- Corrected various potential server crash issues.

- Corrected long standing licensing issue where the license could be invalidated if the NIC to which the license was bound was disabled or had the network cable unplugged

CHANGES/FIXES

- VVM: Corrected a problem where browser Clients would not logout properly requiring a server restart to correct.

- VVM: Corrected a problem where the server would crash if updating a configuration string with a string that exceeded the string's maximum size.

- VVM: Changed the license request launch URL to include the new default password of "admin".

- VVM: Corrected an intermittent problem with SIP connections where there would be no outbound audio after an RTP media destination address change. The problem only occurred when the SIP client was configured to use the first received RTP packet as the RTP destination address. The problem was due to an RTP packet for the old media destination being received after the media destination address change was requested. The problem was resolved by adding the old media destination address to the "ignore list' and then discarding all pending RTP packets when processing the media destination address change request.

- VVM: Corrected the debug messages for RTP errors to display correctly.

- VVM: Corrected a problem where SIP connections could not be established from an external SIP client to the secondary server using the public (NAT) IP address. The problem was due to using the server private IP address for the internal SIP client.

- VVM: Change the playback of pre-recorded audio files to check for file size changes in addition as the file name change in order to determine whether to reload the WAV file from disk.

- VVM: Added the OPUS codec to the supported SIP codec list as the highest ranked codec due to its 48Khz sampling rate.

- VVM: Changed the Client Positioning API to add a 'maximumofflinetime' parameter to allow specifying the maximum number of hours a client would be included in the positioning results after going offline.

- VVM: Corrected a problem with Failover when running on the secondary server where the IP address of the NAT for the primary server was being used as IP address of the SIP server NAT instead of the IP address of the NAT for the secondary server. As a result, all SIP clients were unable to connect to the SIP server when the secondary server was active.

- VVM: Corrected a problem where SIP outbound calls would intermittently fail to be established due to an invalid P-Asserted Identity.

- VVM: Corrected a problem with the RTS TIF implementation where the server would crash when establishing a connection with RTS MC. The problem was caused by the addition of "standard" SIP clients to the configuration that were not associated with the "TIF" SIP clients.

- VVM: Changed the Resources API to add support for a DELETE request to allow for deletion of an existing file.

- VVM: Added a Dial API to facilitate dialing SIP phones from the browser VCP.

- VVM: Added a separate Client Type for the Video Streaming URL.

- VVM: Added Opus codec support for SIP Devices.

- VVM: Added support to the In-Dialog Invite handler for parsing the SDP in order to check for changes to the RTP IP address and/or port.

- VVM/VCP/VDI: Corrected a server crash which resulted from selecting the Opus codec with at sampling rate of 32KHz. The problem was due to the Opus codec having no support for 32KHz. To correct the problem the sample rate will be forced to 48 KHz if 32 KHz requested.

- VSA: Changed the default System Admin password from no password to 'admin' so that web browsers will offer to save the credentials.

- VVM: Changed the default color scheme.

- VVM: Added a new permission returned during the web based System Administration authentication called 'ALLOW_LICENSING' which indicates whether or not to permit the display of the License dialog.

- VVM: Corrected the Client Positioning API to not return positioning data for clients that have the Geo Mapping Disable option set.

- VVM: Added a Video Streaming API to allow the web based Control Panel to indicate to the server when video streaming has been initiated so that this status can be indicated on the associated selector of all other web based Control Panels.

- VVM: Added the 'PlaySound' API to allowing playing of a user specified WAV file to one or more destinations.

- VVM: Corrected a problem with the web based Control Panel where the login validation would match any client type login name/password including VDI,SIP and VIDEO clients.

- VVM: Added 'TIF VU Meter' licensing support in order to be able to suppress the TIF VU meter display for installations that utilize the Pacific Interactive Tally display.

- VCP: Corrected a problem introduced with the Core Audio API where the volume was not changing in a linear fashion from full volume to minimum volume.

- VCP: Corrected a problem where the audio level could not be muted. The problem was due to attempting to set the volume to a negative value which is not supported.

- VVM: Changed the handling of Direct IP Trunks to use the SIP Domain Name if specified.

- VVM: Corrected a system crash that resulted from an attempt to open a WebSocket with an invalid authorization ID.

- VVM: Corrected the System Settings API to not require Master Administrator level privileges for access.

- VVM: Corrected the playing of an RTSP MPEG4 stream to support audio sample rates other than Mono 44100 Hz. The problem was resolve by saving the audio sample rate and number of channels along with the codec name.

- VVM: Corrected a problem where the video streams would frequently stop after playing for a short while. The problem was resolved by periodically sending a 'Get Parameter' request as an RTSP keep alive message.

- VVM: Corrected a problem with the licensing mechanism where the license would indicate invalid if the NIC that was in use when the license was generated, was no longer connected to a network. The problem was due to the licensing mechanism using an API which did not return the MAC address of the NIC if disconnected.

VVM/VSA v5.1 - 09/14/2018 – Public Release

SUMMARY

- Corrected various issue with browser based System Admin.

- Corrected various SIP implementation issue

- Corrected potential server crash issues.

- Removed support for Windows XP

CHANGES/FIXES

- VVM: Added support for updating all templated clients if any settings were changed for the template base.

- VVM: Added support for restarting the server when the Audio Mix Sample rate or the VCOM Client IP Ports are changed.

- VVM: Corrected a problem with the WebSocket where the thread would intermittently crash on termination.

- VVM: Added checks to confirm presence of SSL certificate files before initializing the secure WebSocket.

- VVM: Removed project support for Windows XP.

- VVM: Increased the maximum allotted size for the configuration data information from 10Mb characters to 20Mb

- VVM: Corrected a calculation error that inadvertently allowed the default RTP port range to be from 10000-42767 instead of the intended 10000-32767 which matches the Cisco standard.

VVM/VSA v5.0.5 - 04/29/2018 – Public Release

SUMMARY

- Corrected VLink TIF issue with RTS MC and Pacific Interface line status updates

- Corrected SIP call transfer issues.

- Corrected potential server crash issue.

- Corrected various issue with browser based System Admin.

- Eliminated backwards compatibility with VCP version 3.x.

CHANGES/FIXES

- VVM: Changed the Audio Input and Output Level Gain Step to detect value changes and set the appropriate flags such that the changes will implement dynamically when changed from the VLink TIF System Administration interface.

- VVM: Changed the configuration methods for updating the Audio Codec and Sample Rate to detect value changes and set the appropriate flags such that the changes will implement dynamically when changed from the Audio Settings web interface.

- VVM: Changed the version compatibility for all Control Panel variations to specify a minimum of v4.0.0.

- VVM: Corrected a Party Line problem where PLs with indexes 50 or greater were non-functional.

- VVM: Corrected a problem that the dropped registration of an active SIP line goes undetected.

- VVM: Corrected a problem with the SIP call handling where the transmit audio would intermittently not work when an already active call was transferred in from another extension on a PBX.

- VVM: Changed the Authentication API to return the configured Connection Timeout. The configured timeout will be use by both the server and the client In the case of the Master System Admin, the default connection timeout will always be used.

- VVM: Corrected a problem with VLink TIF where the 'Number of SIP Lines' and the 'SIP Line Status' messages were erroneously being limited to 40 items per message. In a system more than 40 lines, when sending multiple messages for the Number of SIP Lines, only the second message was used, resulting in only those lines being available in the RTS system.

- VVM: Corrected a problem with VLink TIF where after a SIP line was enabled or disabled, neither the 'SIP Line Names' nor the 'SIP Status Line' messages were being sent to the RTS System.

- VVM: Corrected a problem where the server would crash if a file was uploaded to the 'Graphics Files' directory with a file name length greater than 20 characters.

VVM/VSA v5.0.2 - 01/20/2018 - Private Release

SUMMARY

- Corrected issues with VLink TIF dynamic line status updates

- Correct VLink Trunking issues with IFB Alphas

- Corrected potential server crash issues.

CHANGES/FIXES

- VVM: Build: Changed the default system configuration file to set the default HTTPS port.

- VVM: Corrected a problem with the VLink TIF implementation where the number of active phone lines were only communicated to the RTS Master Controller when the connection was initially established. As a result, if a SIP Client was subsequently enabled, it would not be available for configuration in AZEdit.

- VVM: Corrected a problem with the reversed display of the Secondary Server IP address message.

- VVM: Corrected a problem with Telex Trunking where when connected to a TM10K only 120 IFB Alphas were being reported.

- VVM: Changed the installer to copy the 'SSL Certificates' directory on install and remove the copied files on uninstall.

- VVM: Corrected a problem with Web Server where the server would crash with an unhandled exception if an invalid SSL Certificate was installed.

VVM/VSA v5.0.1 - 01/06/2018 - Private Release

SUMMARY

- Corrected issues with browser based System Administration

CHANGES/FIXES

- VVM: Reduced the browser authentication VSA and VCP token expiration time for to 5 seconds.

- VVM: Changed the Label API for Call Notification selector handling to use the already existing string.

- VVM/VCP: Enhanced the Release Remote Selector Request to support Listen selectors.

- VVM: Corrected the Selector Activity API DELETE request to release the activated selector for the specified client instead of erroneously attempting to release the selector of specified client for the client of the active selector.

- VVM: Changed Selector Activity API to add support for a DELETE request to allow an active talk or listen to be deactivated.

- VVM: Changed the Config/Labels API to suppress the client credentials if requested by a Control Panel.
- Changed the Config/System Settings API to suppress the master admin credentials if requested by a Control Panel.

- VVM: Added a Client Positioning API to allow for retrieval of client GPS Information (if available) and IP Address for Geo Location.

- VVM: Changed the SSL API to restart the Web Server and Web Socket threads after successfully generating or installing an SLL Certificate.

- VVM: Corrected a problem where the Cipher list was not being cleared when being built resulting in an attempt to write beyond the bounds of the allocated storage area if the Web Server was restarted.

- VVM: Changed the WebSocket endpoints from being class members to being dynamically allocated as the endpoint destructor contains critical shutdown code that must be executed. This was necessary to support a Web Socket restart as is required when generating or installing an SSL certificate or executing a 'System Restart'.

- VVM: Corrected a problem with the restoration of talk and listens after a disconnect where a previously latched talks or listens could be restored even if the selector was not actively latched prior to a momentary disconnect.

- VVM: Changed the LABELS API to add support for saving the "Default Selector Assignments to Latch Disable" flag.

- VVM: Implemented SSL API to allow generation and retrieval of a Certificate Signing Request and Private Key for creation of either a CA Signed Certificate or an Self-Signed Certificate.

- VVM: Changed the default values for the Telex Trunk Master and Telephone Interface IP Ports to standard values so that the values are returned by the System Settings API for display in the System Settings dialog.

- VVM: Changed the default values for the Secondary Server IP Ports for Data and Audio to non-zero values so that the values are returned by the System Settings API for display in the System Settings dialog.

- VVM: Changed the Systems settings API to return the default value for the IP address of the Local Interface for Failover.

- VVM: Changed Labels API to add support for Play Selector Activation Messages, Play Client Connection Sounds and Play Battery Status Messages.

- VVM: Changed the default value for the HTTP and HTTPS ports so that the web server will be enabled by default when upgrading an existing installation using a System Configuration file without the HTTP and HTTPS ports specified.

- VVM: Corrected the Client Statistics API to display the RARBD (Receive Audio Rate Before Decoding) which was not displayed due to the use of an incorrect object name.

VVM/VSA v5.0 - 12/14/2017 - Public Release

SUMMARY

- Major release which requires a v5 license.

- Formalized the browser based System Administration as the officially supported platform relegating the System Administration for Windows to legacy status.

- Added support to VLink TIF for configurable Dial Plans.

- Corrected potential server crash issues.

CHANGES/FIXES

- VVM: Added the ability to specify the Local Network Interface for failover so that the network used for the failover connection can be differentiated from the network used for client connections.

- VVM: Changed the License API GET to return all license information and detail.

- VVM: Changed the launch of the web based System Admin to start maximized.

- VVM: Changed the desktop and start menu shortcuts to launch the browser based System Admin.

- VVM: Added the ability to specify the Local Network Interface for failover so that the network used for the failover connection can be differentiated from the network used for client connections.

- VVM: Changed server licensing for the v5 release.

- VVM: Corrected a problem where the server would crash when enabling the Web Server configuration file.

- VVM: Changed the Virtual Matrix installer to launch the Web based System Admin instead of the Windows based System Admin.

- VVM: Changed the License API to return the updated license details in response to the License POST.

- VVM: Corrected the License API to write the new license file with the correct file size.

- VVM: Corrected a problem with the Resource API where files greater than 128Kb could not be uploaded due to this being the default web server maximum request size.

- VVM: Corrected the Resources API POST to return a valid status code as it erroneous indicated an internal server error after a successful operation.

- VVM: Added support to VLink TIF for configurable Dial Plans that allow the dialing from the KP-32 to complete immediately when matching sequences are detected for things like domestic numbers and internal extensions. For non-matching sequences the completion will happen after a configurable dial sequence timeout.

- VVM: Added support for displaying the Client Number (aka Port #) in the Client Statistics.

- VVM: Added a Resource API to allow retrieval of list of Sound Files, Graphics Files and Image Files as well as uploading new Sound Files, Graphics Files and Image Files

- VVM: Corrected the Labels API to parse the return values for the VLink TIF RTS Alpha Function Type and Index.

- VVM: Changed the Labels API to use the standardized Label Category text for the VLink TIF RTS Alpha Function Type value that matches the values returned in the corresponding list request.

- VVM: Changed the SLL API to remove the WSS port since that should be determined from the System Settings API.

- VVM: Corrected the SSL API to return the configured HTTPS port instead of a hardcoded value.

- VVM: Corrected the Labels API to update the Label Type if changed in the API post.

- VVM: Added support for a PBX Primary and Secondary Host Name system setting to simplify configuration of the VLink TIF system such that the SIP Trunk client settings for the SIP Target Host Name defaults to the PBX Host Name if not specified.

- VVM: Changed the System Status API to return the status of the VLink TIF Interface.

- VVM: Corrected a problem with the WebSocket where after an authentication failure when opening the socket, the next message received would attempt to use the invalid client index which likely would result in a server crash.

- VVM: Changed VLink Trunking to support the TM-10K which increased the supported number of trunked systems from 31 to 255. Due to design constraints of the previous implementation, Matrix Index 31 (Matrix Number 32) cannot be used and existing VCOM Clients can support only up to Matrix Number 63.

- VVM: Corrected an issue with VLink Trunking where the initial Trunk Crosspoints message would result a request to resend the Trunk Definition message due to an erroneous discrepancy between the number of actual allocations and the number of allocation specified in the crosspoints message.

- VVM: Changed the Labels API to return the Templates as an array with each element having a Template Type and Label ID for consistency with other APIs.

- VVM: Changed the System Settings API and the Configuration Data File output for VLink Trunking and RTS TIF, if licensed, to always output the Local Interface IP address and the Trunk Master / Master Controller IP Port # so that the web interface can properly display the default field values.

- VVM: Corrected a problem with the Websocket where the server would crash on shutdown if the HTTPS port was not specified.

- VVM: Changed the VR Config API to correct an error introduced with the previous check in that prevented returning the JSON data.

- VVM: Changed the TIF API to add support for a 'show=disabled' parameter to allow display of disabled phone lines.

- VVM: Changed the TIF API to include the ID and ALIAS to allow cross-referencing with other APIs.

- VVM: Changed the TIF API to return DISABLED in the LINE_STATE if the client is disabled.

- VVM: Changed the handling of ALWAYS_ON_VISIBLE/INVISBLE to not assume the presence of the field implies the setting is 'ON' since a recent modifications to the Selector Assignment dialog resulted in the field being sent with the setting being 'OFF'.

- VVM: Corrected a problem where the server could fail to shutdown properly if either VLink TIF or VLink Trunking were enabled.

VVM/VSA v4.4 - 07/08/2017 - Public Release

VVM/VSA v4.3.3 - 10/23/2016 - Public Release

- VVM: Corrected a problem where releasing an outbound call that is still ringing would result in the subsequent call, either inbound or outbound, immediately disconnecting.

- VVM: Corrected a problem when dialing an incorrect number where the call would appear to ring indefinitely. The problem was due to never implementing support for the SIP Session Progress message which is sometimes sent in place of a SIP Ringing message.

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- VVM: Corrected a problem with the SIP User Agent header where the header content could be indeterminate rather than the intended content used to identify the company name, product and version.

- VVM: Corrected a problem where the VVM service was not being shutdown properly on a computer system restart or shutdown but rather was being force terminated.

VVM/VSA v4.3.2 - 10/23/2016 - Private Release

- VVM: Corrected a problem with a server initiated outbound SIP session where when the session was terminated from the server there was a 5 second delay before the call was terminated.

VCP4iOS v4.2.4 - 10/18/2016 - Public Release

- Corrected a problem where a Call Notification Selector was erroneously added to the end of the selector list even though a

dedicated talk selector was already assigned.

- Corrected a problem where the Control Panel would not reconnect and resume operation after an incoming phone call.

- Corrected a problem with the receive packet jitter calculation when an audio codec is configured that uses a 5ms frame size resulting in erroneous values used for the jitter buffer size calculation and also reported for the client statistics.

- Changed the handling of forced packet drops in the case of jitter buffer overruns to take into account a 5ms frame size.

- Corrected an incompatibility problem with iOS 6 preventing the application from being used.

VVM/VSA v4.3.0 - 04/06/2016 - Public Release

VCOM/VLink - VVM/VSA v430 - 04/06/2016

- VVM: Changed the handling of forced packet drops in the case of jitter buffer overruns to take into account a 5ms frame size.

- VVM: Relocated all sound files located in the top level working directory to the 'Sound Files' directory.

- VVM: Corrected a SIP Client problem when an access code is configured where all outbound calls would terminate immediately after connecting if an inbound call was previously terminated due to not entering the access code.

- VVM: Changed the RTS Telephone Interface handling to replace the custom dial plan with a more generallized plan.

- VVM: Changed the RTS Telephone Interface handling to adjust Dial Sequence automatic completion time from 10 seconds to 7 seconds.

- VVM: Changed the behavior of RTS Trunking to delay processing of remote system disconnects to minimize the visible effect of a momentary loss of connection for any given system.

- VVM: Corrected a problem with RTS Trunking where the System Connection state for Matrix Index '0' was not being cleared when the connection to the Trunk Master was lost or the Trunking thread was shutdown. This would leave the selectors for Matrix Index '0' indicting as enabled when they should be disabled.

- VVM: Corrected a problem with RTS Trunking where the selectors for all RTS Alphas associated with a specific system were seemingly spontaneously deleted from all Control Panel Selector Assignments. The selectors once deleted had to be re-assigned manually with the System Administration application.

- VVM: Corrected a problem with the RTS Telephone Interface implementation where the ringing sound played to the Telex system to indicate the call is being connected, was not stopped if the SIP call failed.

- VVM: Corrected a problem with the RTS Telephone Interface implementation where after a call is active, user initiated DTMF digits used for call control were not being sent immediately.

- VVM: Corrected a problem with the RTS Telephone Interface implementation where a call initiated from a KP32 to an external number would potentially result in an abnormal server termination.

- VVM: Changed the configuration file handling to store/retrieve the Label IDs in/from the System Configuration file. This was necessary due to implementation of Web API which allows third party custom applications to be written. In these applications, the Label IDs used for identification of a particular client, party line or fixed group, may be stored externally and therefore must not change after a system restart.

- VVM/VSA: Added a SIP Client option for sending 'digits' when a Control Panel request is initiated for activating and deactivating the interface. This option was added to support SIP Devices that can perform specialized functions such as activation /deactivation of a relay in order to key a two-way radio transmitter.   After the SIP Client Session is active, if configured, the specified 'activation' digit will be sent to the SIP Client when the interface is activating (aka first Talk Selector activated) and the specified 'deactivation' digit will be sent to the SIP Client when the interface is deactivating (aka last Talk Selector deactivated).

- VVM: Corrected a problem when adding new clients where the client appeared to be added correctly in the VSA however the client was in fact not added correctly in the VVM. The problem occured only after previous deletions.

- VVM: Updated the Copyright information showing in the file properties.

- VVM/VSA/VCP: Added system option, configurable in the VSA System Settings, to allow the assignment of a selector for 'Self', primarily for testing and diagnostic purposes. When this option is enabled, a Talk Selector can be assigned to a Control Panel via the Selector Assignments, allowing that Control Panel to talk to itself.

- VVM/VSA/VCP: Added a per client option, configurable in the VSA Selector Assignments, to allow setting the default for talk selector assignments to be latch disabled when assigned. This option is used both by the VSA Selector Assignments as well as the VCP for Android Selector Assignments.

- VVM: Changed the Web API for 'config/labels' to add Actiivty Log and Debug Log messages to indicate when Labels are added/changed/deleted.

- VSA: Changed the Client Statistics to add a 'Description' column to allow for sorting by client description. Since this column will likely only be used for sorting it is positioned as the last column.

- VSA: Changed the Client Statistics to add user option to choose which client type of Control Panel, Device Interface and/or SIP Client to display.

- VSA: Changed the Client Statistics such that the Client Selector Name column width is set to only the width necessary to display the longest item.

- VSA: Changed the Client Statistics to not display the warning color based on 'Packet Loss Since Login'. As a result packet loss will only cause the warning color to be displayed if there was any packet loss in the last minute.

- VSA: Changed the Client Statistics to make the 'Duration', 'Battery', 'IP Address' and 'Version' columns sortable.

- VVM: Add support for a special 'Call Notification' Selector to allow user placement of the Call Notification selector implemented originally only on the iOS and always placed as the last selector. The Call Notification selector allows all incoming calls to be indicated in the same selector position even if a selector for the caller exists elsewhere. Previously if a call was received from a caller and a selector did not already exist, a selector for that caller would be permanently added to that client's Selector Assignments. The Call Notificaiton selector is currently supported on only iOS and Android.

- VVM/VSA: Added support for playing optional audible 'Connection' sounds when the client disconnects and then reconnects to the server as a result of a loss of network connectivity. No message will be played if the user manually disconnects and/or reconnects from/to the server. These sounds will be played to all clients that have the 'Play Connection Sounds' option enabled in the Client Options.

- VVM/VSA: Added support for playing optional audible 'Selector Activation' messages when the client activates/deactivates any talk or listen selector. These messages will be played to all clients that have the 'Play Selector Activation Messages' option enabled in the Client Options.

- VVM/VSA: Added support for playing optional audible 'Battery Status' messages at 30%, 25%, 20%, 15%, %10 and 5% of battery remaining. These messages when enabled in the System Settings will be played to all clients that have the 'Play Battery Status Messages' option enabled in the Client Options.

- VVM/VSA: Added support for playing an audible 'Low Battery Alert Displayed' message at the request of the iOS Client when it determines that the iOS 20% Battery Alert Level is displayed.

- VVM: Changed the handling of Labels ID to allow for duplicate Label IDs to be detected when the Configuration Data file is parsed. Although in theory a duplicate Label ID should never happen, a previous bug since corrected allowed this situation to occur. If a duplicate Label ID did exist in the file when it was parsed, it resulted in a failure to parse correctly with the possible loss of a significant portion of the configuration. This potential failure is now prevented by detecting duplicate Label IDs during parsing and discarding the duplicate.

- VVM: Corrected a problem with Client Statistics where the client connection state change time was being set to the current time when resetting the client statistics. As the client connection state change time is now persistent, this value should never be reset.

- VVM: Corrected the Selector Assignment mode Label List request mechanism to exclude all Labels that have the restriction enabled for 'No Local Assignment by User'.

- VVM: Added an Activity Log message to indicate when the Selector Assignments configuration has been changed via the client's "Selector Assignment" mode.

- VVM: Change the handling of VCP for Droid to not automatically assign a selector for an incoming call for version 4.3.0 and upward as incoming calls are now handled by the Call Notification selector capability.

- VVM: Corrected a problem with RTS Trunking where the VCP talk/listen selector states for Telex selectors were not being restored after a momentary client disconnect/reconnect.

- VVM: Corrected a problem with the Selector Activity API where an erroneous selector state could be returned for a VDI or SIP if the selector configuration was changed while the device was not active.

- VVM: Corrected a problem with SIP Clients where the Activity Log message to indicate 'SIP Session Activation' would indicate the wrong RTP Destination IP Address if configured to NOT use the SDP for the RTP Destination.   The message will now indicate that the address is "TDB" and a second message displayed when the address is determined based on the address of the first packet received.

- VVM: Corrected a problem where a SIP Registered Trunk would not answer a call if an Invite was received without an SDP but instead would reply with a 400 response indicating "can't provide an offer". The problem, which has existed since inception of the product, was due to improperly handling the response for this unusual situation.

- VVM: Changed the handling of the Registration Time from using the master profile to adding as a parameter when creating the Registration message. This was done in an effort to eliminate a verified but unreproduceable problem where received Invites were being ignored until only after a re-registration was sent.

- VVM: Corrected a RTS Trunking problem where activation of a listen selector for a trunked audio source could result in the source audio be directed to an incorrect destination. The problem was due to a known bug in the Trunk Master where a 'Trunk Allocation' update message could intermittently not be sent.

- VVM: Corrected a RTS Trunking problem with the assignment restrictions where the restrictions specified in the remote systems for 'Assignment by Administrator' and 'Assignment by User' were erroneously swapped during parsing.

- VVM: Changed the RTS Trunking behavior of the Trunk Allocation request to always enable the 'Request Target to Listen Via Auto-Table'. This will facilitate correct operation when Auto-Tables are in use and have no impact when they are not.

- VVM: Changed the Autenthication API to return a series of 'Allow' and 'Supports' flags so that the System Administration implementation can determine which options to allow and which features to display.

- VVM: Corrected a problem with the Config/System API where the PUT for the Trunk Master Main IP Address was parsing the wrong JSON field.

- VVM: Corrected a problem with the Config/System API where the return code erroneously indicated JSON content instead of no content.

- VVM: Corrected a problem with the Config/Labels API where the status code was not being check properly resulting in a failure to parse all label information as well as a failure to report that parsing was successful.

- VVM: Corrected a problem with the Selector Activity API where the selector states were being returned when the device was 'connected' as opposed to 'active'. For a SIP client, the device is 'connected' when registered but not 'active' until the session is started.

- VVM: Changed the output methods for the Web Server 'Labels' API to return all the label type specific details such as the VCOM Client Options, Audio Settings and Selector Assignments. The label type specific details will only be returned when requesting a specific Label ID and not when requesting the list of Labels.

- VVM: Added a Web Server Authentication API for receiving and validating a client login name and password as a prerequisite for accessing any other API. Once a login name and password is validated, an Authorization token will be returned to the client. This Authorization token must be included in all subsequent API requests or the request will be rejected. All Authentication tokens will expire after 10 minutes of API inactivity.

- VVM: Changed the 'config' and 'status' Web Server APIs to accept a list of Label IDs rather than just a single Label ID. The list of Label IDs must be presented in '<ID>[,<ID>] format with a comma delimiter separating the IDs.

- VVM: Changed the 'action' Webserver APIs for forcing a failover or failback and for restarting the system from an HTTP PUT request to a HTTP POST request.

- VVM: Corrected a problem with the Web Server where the server startup would result in an abnormal program termination if the specified 'Web Resources' directory did not exist.

- VVM: Added a REST API to support forcing a failover which will manually transfer control from the active server to the standby server ('api/v1/action/forcefailover').

- VVM: Added a REST API to support initiating a system restart which will result in the termination of all active server threads server followed by the reactivation of all server threads, essentially rebooting the server ('api/v1/action/systemrestart').

- VVM: Changed the parameter string used to set the mime type to correctly set plain text.

- VVM: Added a plain text response to indicate 'not supported' for any unsupported method used on a supported API.

- VVM: Corrected a problem with the 'config/labels' API where processing the request would result in the web server thread hanging.

VVM/VSA/VDI/VCP v4.2.3 - 11/01/2015 - Public Release

- VVM: Added a preliminary REST API to allow updating all configuration items associated with the System.

- VVM: Added a preliminary REST API to allow retrieval of all information and status associated with the System.

- VVM: Added a preliminary REST API to allow adding and removing members from a Party Line. In addition to VDI and SIP clients assigned the PL, the GET method will also include VCP clients which have a selector assignment for the PL. Similarly, the SET method expects VCP clients and will add or remove the selector assignment for the PL as appropriate.

- VVM Corrected a problem with the Telex TIF Tally system where intermittently the Telex System would lose both control and status of the SIP phone lines. The problem occurred as a result of the Telex standby Master Control (MC) very briefly erroneously indicating it was the active MC before again indicating it was the standby controller. As a result of the erroneous indication, the server switched to using the "new" active MC even though it was actually still the standby MC. The problem was resolved by not acting on the indication of the active MC but rather evaluating the states of both MCs to prevent a switch of control in the case where both MCs indicate they are active. As a result of this change if the active MC is reset, there will be a brief delay before control is switched to the newly active MC while the connection to the reset MC is timed out.

- VVM: Corrected a problem with the Telex TIF Tally system where the implementation of the 'SIP Port Status' where separate data structures were being maintained for each Master Controller (MC). As a result, if control was switched from the main MC to the backup MC or vice versa, the state of any active SIP phone lines sent to the newly active MC would be reported incorrectly. The problem was resolved by making the SIP Port Status data structure common for both MCs.

- VVM: Corrected a problem with the Telex TIF Tally system where the server would crash whenever a '#' key was dialed from an RTS KP-32. The problem occurred when querying the audible key press confirmation tone due to an invalid filename pointer as a result of a syntax error in the file name list. Also as a result of this error, there would be no audible key press confirmation tone for the '*' key.

- VVM: Corrected a problem with the Telex TIF Tally system where dial tone was sent perpetually to the RTS system after a seize if there was no other dialing activity. To prevent this problem a timeout was implemented such that after a period of no dialing activity, the line is automatically released and the dial tone removed.

- VVM/VSA: Added support for an 'Auto-Answer Access Code' to prevent access to sensitive audio streams from unauthorized callers to a SIP client configured for Auto-Answer. When an Access Code is configured, an 'Auto-Answer Notification Message' should be also configured in order prompt the user to enter the access code either the default message or a user provided message. When the Access Code is entered correctly the caller will be joined to the pre-configured selector assignments and a "Connect" sound will be played to all listeners. When the caller subsequently disconnects, a "Disconnect" sound will be played to all listeners. If the Access Code is not entered correctly, the caller will be summarily disconnected.

- VVM: Added a preliminary REST API for 'Party Line Activity' to retrieve the party line active talk and listen membership states. If no specific ID is given to specify a specific party line, the Party Line Activity for all party lines will be retrieved.

- VVM: Added a preliminary REST API for 'Selector Activity' to retrieve the client active talk and listen selector states. If no specific ID is given to specify a specific client, the Selector Activity for all clients will be retrieved.

- VVM: Corrected a problem with the new selector volume gain change implementation in the VVM v4.2.2 used only by the VCP4Windows v4.2.2 where the wrong variable was used for changing the selector volume gain possibly resulting in the inability to properly change the gain. The problem only occurred if both the system and client Audio Output Level gains were set to non-zero values.

VVM/VSA/VDI/VCP v4.2.2 - 08/24/2015 - Public Release

- VVM/VCP: Corrected a problem with the mechanism to restore the previously active talk and listen selectors after a Control Panel disconnect where the selectors to be restored could be lost if the login failed before the selectors to restore were re-activated by the Control Panel.

- VVM/VCP4Windows: Corrected a problem where the templating of Selector Assignments would preclude the use of the Control Panel for Windows adjustable Selector Volume Gains as a result of the Selector Volume Gain values being stored with the Selector Assignments. The problem was that since the Selector Assignments are duplicate from the base template on each Control Panel login, any adjustment to the Selector Volume Gains made by the Control Panel templated to the base template would be overwritten by those of the base template on a subsequent login. Backwards compatibility has been maintained between older VCPs and the new VVM.

- VVM/VSA: Corrected a problem where when using client login credentials for the System Administration, the client login credentials would be displayed in the System Settings.

- VVM: [RTS Only] Added user interface for the previously existing internal option to 'Ignore 8 Character Unicode Alphas'.

- VVM: Added support for the re-sampling of sound files so that their sample rate matches the configured Audio Mix Sample rate. This was necessary in order to support a 48 KHz Audio Mix Sample rate since all sound files presently are set to a 32 KHz sample rate.

- VVM/VCP/VDI: Added support for converting a buffer size greater than one audio frame to allow for conversion of sound files. In the case of an audio frame, if the buffer size is not converted to the expected output length the audio frame is discarded. In the case of a sound file, no frames will ever be discarded.

- VVM: Increased the size of the SIP Target User Name and Host Name from 20 characters to 40 characters.

- VVM: Added support for a Secondary Target Host in order to support both a Primary and a Secondary Host for Direct IP and Registered trunks.

- VVM: Added voice activity detection for Fixed Groups in order to indicate voice activity based on the voice activity of its members.

- VVM: Corrected a problem where the SIP server initialization could potentially hang if the server was abnormally terminated during the prior SIP server initialization. During the SIP server initialization several backup database files are automatically created and immediately deleted after initialization completes. If however these files are not deleted, then any subsequent SIP server initialization will hang when attempting to create the files.

- VVM/VSA/VCP4Windows: Added preliminary support for Fullband 48 KHz audio and the OPUS codec in order to facilitate the future implementation of these capabilities in all clients.

- VSA: Enabled the Automatic Gain Control checkbox in Audio Settings for VCP4iOS client types.

- VDI: Added support for ASIO 24 bit integers as this is the default format expected by the Dante Virtual Sound Card.

- VDI: Added support for ASIO 32 bit floats as this is the default format required by Wine ASIO.

- VVM: Added preliminary support for a Web Server in order to facilitate the future implementation of the System Configuration via a Web Browser as well as the implementation of a REST API.

- VVM: Added preliminary support for a REST API which allows for adding, editing and deleting VCOM Clients, Party Lines and Fixed Groups.

- VVM/VSA: Added support for a configurable SIP Registration Expiration Delay Time. Previously the Registration Expiration Delay Time was hardcoded to 5 seconds.

- VVM: [RTS Only] Corrected a critical problem with RTS Trunking where the server would intermittently hang during shutdown or failover requiring the server to be forcibly terminated.

- VVM/VSA: Changed the handling of SIP Registered Trunks to allow the specification of a Target SIP Proxy Server.

- VVM: Corrected a critical problem introduced in v4.2.0 with the addition of the display of the SIP Registration and Session Failure Response Codes and Reasons which could potentially result in an abnormal server termination. Although the potential for this problem existed, it is believed that this problem was unlikely to be encountered during typical system operation. The only known occurrence of this problem was due to an unrelated bug which generated an internal SIP server error.

- VVM: Corrected a problem with the handling of a received SIP Re-invite sent without an SDP which would result in the SIP Session being terminated. The problem was due to the wrong method being called to respond to the re-invite which resulted in an internal SIP server error.

- VVM: Corrected a critical problem introduced in v4.0.1 with the change to support dedicated audio send sockets, where the audio send thread for a specific client could hang, possibly indefinitely. Although the potential for this problem existed for an extended period of time, it is believed that this problem was unlikely to be encountered during typical system operation.

- VVM/VSA: Added support for the mix audio sample rate of 48000 Hz in preparation for adding full band codecs to improve the audio quality of the system. For now the default mix audio sample rate will remain at 32000 KHz.

- VSA: Corrected a problem with the installer where the application failed to install properly in silent mode.

VVM/VSA v4.2.1 - 04/22/2015 - Private Release

- VVM: Added Activity Log messages to indicate when System Configuration information has changed from the System Administration application so that it if necessary it can be determined which user made the changes.

- VCP4Windows: Corrected the calculation of the screen area used for selector resizing and re-layout to adjust for the omitted top bar when in full screen mode.

- VCP4Windows: Changed the handling of Selector Spacers which were previously identified by implication, to explicitly identify Spacers to eliminate any possibility of ambiguity and ensure correct handling and display of the Spacers.

- VCP4Windows: Corrected a problem where multiple Selector Row Breaks were not displayed correctly if the Row Break occurred at end of a "full" selector row. The problem was resolved by ignoring the Row Break if a "full' selector row would have resulted in a natural row break.

- VCP4Windows: Added support for the insertion of a Selector Row Break in the first selector position.

- VCP4Windows: Corrected the calculation of the screen area for the active display to account for an offset from the base setting (0, 0) of the principle display when displaying full screen on a non-principle display.

- VCP4Windows: Corrected a problem with panel resizing and selector re-layout where if selectors were added or removed it could result in missing or duplicated selectors.

- VVM: Changed the handling of the error message which indicates an SNMP initialization failure such that the error message is only displayed one time. This was necessary to eliminate repeated displays of the failure message since SNMP initialization is re-attempted if required whenever an SNMP message needs to be sent.

- VSA: Corrected a problem where any SIP Automatic Dial Sequence entered from the Client Options dialog would be immediately lost.

- VCP4Windows: Corrected a problem where the insertion or removal of any USB device could result in the inability to receive audio.

- VCP4Windows: Corrected a problem where, when using Telex Trunking, the Selector Assignment dialog shows the local intercom in the System List three times, once as the 'Local' intercom and twice as the Telex named intercom.

- VVM: Corrected a problem with the transmission of DTMF Tones sent by an internal SIP Client as RTP Events where the tones would not always be decoded properly by the far end, potentially resulting in a failure to establish a call.

- VVM: Corrected a problem where SIP Direct IP Trunks were non-functional when operating on the secondary (failover) server. The problem was due to the SIP Host being set to an invalid value of 0.0.0.0 for the internal SIP Client resulting in its failure to register with the internal SIP server. The problem was corrected by changing the SIP Domain to be the IP address of the active Local Network Interface.

- VVM: Changed the Activity Log message which displays to indicate a registration failure to not be restricted to only a SIP Registered Trunk. It was previously anticipated that there would never be a non-programmatic registration error with the internal SIP server however there is no valid reason to restrict it.

- VVM Changed the SIP Invite Session initiation and failure messages from Debug Log messages to an Activity Log messages. The failure message also now includes the failure response code and reason.

VVM/VSA v4.2.0 - 04/13/2015 - Private Release

- VVM: Added a text description to the SIP 403 message sent when the user is determined not to be authorized for its identity.

- VVM: Removed all checks to determine if a user is authorized for its identity to work around a problem with a specific SIP client that seemingly erroneously sent its local IP in the 'From' field instead of the server IP resulting in a failure to match the provided host to the server realm.

- VVM/VSA/VCP4Windows/VDI: Added a configurable Connection Timeout value per VCP/VDI client in order to facilitate connections over satellite where the latency of the connection can add 1 or more seconds to the request/response time.

- VSA: Corrected a problem with the deletion of clients that are used as a base templates where references to that template are not also deleted. While this problem in itself does not create any immediate problem, the presence of the invalid references in the System Configuration file does create a subsequent problem when the system is restarted due to a Label ID being generated when the invalid reference is encountered.

- VSA: Added a Connection Status report which shows the current login state for each client including the last connect time to allow for easy determination as to which client logins are not being utilized.

- VSA: Added a License Information report which shows how many of each licensable type have been licensed as well as how many of each type are utilized and how many are remaining.

- VVM: Added support for persistently retaining the last connect time of a client so that administratively it can be determined the last time a particular client has logged into the server. Previously the last connect time would be set to the server activation time which meant that after any server restart or failover it was not possible to know the actual last connect time for clients that have not logged since the server re-activated.

- VVM: Removed the forced assignment of PL Defaults to Control Panels that have no selectors.

- VVM/VSA: Changed the handling of the System Configuration file update mechanism such that the file update can be scheduled for a future time. Previously when and configuration item was modified the file update would be completed 1 second after the last modification. This change was necessary so that newly added configuration items that contain operational information and thus modified frequently would not result in the constant update of the configuration file. The existing configuration items that contain settings information will now result in the update of the configuration file 10 seconds after any modification is made.

- VVM: Corrected a problem with the Control Panel Selector Assignments function where the list of available labels for a selected Telex system and category would not always be displayed as complete.

- VVM: Corrected a critical problem with failover that could result in a server crash after the data connection is restored between servers which are both are active due to prior loss of the data connection. The problem was caused by the active server sending its configuration to the deactivating server before the server deactivating server had completed its shutdown.

- VVM: Corrected a problem where the default SIP Client Options were not being sent to the failover server. As a result, the configuration between the primary and the secondary server would not be properly synchronized if the default SIP Client Options in the configuration file differed from the runtime initialization values. This would potentially result in different behavior for SIP Clients when operating on the secondary server instead of the primary server.

- VVM: Corrected a problem with Geomapping where the historical positioning would show the client as being previously located at the 0,0 position. The problem was due to the android clients initially reporting their position as 0,0 presumably due to the actual positioning information not being available. To avoid the problem the server was changed to ignore the positioning information if reported at the 0,0 position.

- VCP4Windows: Corrected a problem with the restoration of Talk and Listen Selectors where the selectors would not properly be restored if greater than 50 selectors were to be restored.

- VCP4Windows: Corrected a problem where the 'Release Remote Talk Request' was releasing both the active Talk and the Listen on a split Talk/Listen selector instead of only the Talk. In the case of a release remote talk request due to a phone line disconnection, only the Talk must be released otherwise the operator will not receive any ring indication for subsequent calls.

- VVM: Changed the display of version numbers to separate the build number with a dash ('-') rather than a period. ('.').

- VVM: Changed the processing of all Telex Telephone Interface messages sent to the RTS Adam which convey the details and state of the 'SIP Lines' to exclude any configured SIP clients from these messages that are set for 'Disable Client Login / Connection'. This was done to allow a specific customer with multiple systems to have nearly identical configurations in all systems even though some systems require fewer lines than others.

- VVM: Corrected a problem where the P-Asserted Identity header, the Remote Party ID header and the User Agent header would not be included in the Invite message if the SIP Client is configured to not include the SDP with the Invite request.

- VVM: Added a SIP 'User Agent' header to the SIP Invite message to allow identification of Intracom VVM originated invites as well as the specific server version in network traces.

- VVM: Corrected a problem with SIP Registered trunks where a loss of connection with the SIP Registrar or a restart of the SIP Registrar could result in failure of the SIP Session to be re-established. The problem was due to not checking for a registration failure while waiting for the SIP Session to re-activate. As a result of not checking for a registration failure, there was never any attempt to re-register. As a result of no valid registration, all attempts to establish the SIP session would be rejected. The problem was corrected by checking for a registration failure and terminating the attempt to activate the SIP Session until the re-registration is complete.

- VVM: Added a debug message to indicate when an Invite Request is being sent for a SIP Client with a fixed SIP Target User Name.

- VVM: Corrected a problem where an active SIP Session would not terminate if the SIP client was configured for 'Disable Client Login' while the session was active.

- VVM: Added a check before attempting to send an SNMP message to initialize the SNMP interface if not previously initialized. This facilitates updating a license to enable SNMP without requiring a restart of either the Primary or Secondary servers.

- VVM: Corrected a problem with all SNMP messages where an extra zero value was inserted in each message after the 'vcom' object identifier. Since the already published MIB file did not reference this extra value, any SNMP manager strictly using the MIB may fail to identify the SNMP messages.

- VVM: Corrected a problem with SNMP messages where extraneous data was being appended to the OID in every SNMP message. The problem was due to incorrectly setting the message length to the number of bytes in the OID rather than the number of integer values.

- VVM: Corrected a problem with SNMP where the SNMP Traps were not being generated during a System Administration application Forced Failover or Forced Failback operation. The problem was due to only checking for a failover state change when the server was active and as such never updated the failover state when the server went inactive. As a result of this correction, both the primary and the secondary servers will report all failover state changes regardless of whether active or inactive.

- VVM: Corrected a problem where the System Administration application would always show 'Default' in the Title bar after the Virtual Matrix installation even on a clean machine. The problem was due to erroneously specifying the Default profile when automatically launching the System Administration application from the Virtual Matrix installer for the purposes of displaying the System Identification code.

- VVM: Increased the delay time before launching the System Administration application from the Virtual Matrix installer for the purposes of displaying the System Identification code from 2 seconds to 4 seconds in order to provide more time for the Virtual Matrix to be initialized. This was done as it was observed that the System Administration application would frequently fail the first communication attempt with a message indicating it was unable to connect to the Virtual Matrix.

- VVM: Corrected a problem with an incoming SIP call when the 'Inbound Session Activation' was set to 'On Talk Selector Activation' where the configured 'Auto-Answer Notification Sound' was erroneously being played after the call was answered by a talk selector activation. The problem was corrected by removing the check for the 'Inbound Session Activation' being set to 'On Talk Selector Activation' as a condition to initiate playing of the auto-answer sound.

- VSA: Renamed 'SIP Target Host' to 'SIP Target Host Name' for clarity of purpose.

- VSA: Renamed 'SIP Registrar or Proxy IP Address' to SIP Target Server IP Address (optional). This was necessary as it was determined that the server uses the SIP AOR which includes the SIP Target Host Name in the registration request and does not use what was previously indicated as the SIP Registrar or Proxy IP Address.

- VSA: Adjusted the spacing of several SIP Client Option dialog items for display uniformity.

- VVM: Corrected a potential problem caused by the fact that SIP Server IP Port was always being appended to the SIP AOR even if set to default. In the specific case of a SIP Registered Trunk, the specification of the SIP Server IP Address with a port number are in fact optional. As it is potentially possible that the host port number could be specified in the SIP Target Host Name, a problem could arise where both port numbers are appended resulting in an invalid AOR. The problem was resolved by never appending the port number if set to the default port of 5060. While not an ideal solution, this change allows for a port number to be specified either in the SIP Target Host Name or in the SIP Server IP Address.

- VVM: Changed the handling of the P-Asserted-Identity header and the Remote Party ID header to not insert either header if the Local Party ID is blank.

- VVM: Corrected a problem with the P-Asserted-Identity header and the Remote Party ID header where the headers were being rejected by a particular SIP PBX. The problem was due to an extraneous space between the URI scheme ('sip:') and the user name.

- VVM: Corrected a problem where the failure of a SIP Invite Session prior to the session activation would not result in the proper termination of the Invite Session.

- VVM: Corrected a problem where a SIP Registered Trunk would not proceed to initialize the SIP Client if the SIP Server IP Address was not specified however as it turns out the SIP Server IP Address is not actually required for a SIP Registered Trunk.

- VVM: Corrected the Activity Log message which indicates the registration attempt to show the SIP Target Host Name and not the SIP Server IP Address.

- VVM: Added a Debug Log message to indicate if an initiated SIP Invite Session failed prior to the session activation.

- VVM: Corrected a problem with Domain Authentication where any login name greater than 11 characters would not be authenticated properly.

- VCP4Windows: Corrected a problem with the handling of audio device insertion and removal which occurs while the client is logged in where the audio device change will not properly be recognized. As a result, new audio devices could be used and removed audio devices appeared to be still available.

- VVM: Updated SystemConfiguration.txt to include the defaults for the newer options supported by the current server version.

VVM/VSA/VCP4Windows/VDI v4.1.0 - 01/21/2015 - Public Release

Production release. This release corrected a critical problem with the VVM service where a restart might prevent the OS from starting properly if the network connection is not available when the service starts.

- VVM/VSA/VCP4Windows/VDI: Replaced the Voice Activity Detection (VAD) algorithm used since the inception of the product which was overly sensitive to low level background noise and would trigger the VAD indicator even when there was nothing audible. The false VAD indications were especially evident on the 4-Wire hoot circuits where low level bleed back from the 4-Wire phones made it practically impossible to determine who was actually speaking.

- VVM: Corrected a problem with the Virtual Matrix installation script where the 'Sound Files' directory was not being included in the installer.

- VVM: Corrected a problem with SIP Registered Trunks where an incoming call would effectively be auto-answered even though the SIP client's 'Inbound Session Activation' was configured for 'On Talk Selector Activation'. The problem was determined to occur only when that SIP Client was placed into a Party Line with another already active SIP Client. As a result of being in the same Party Line, the already active SIP Client would send an interface activation request to the SIP Client with which it was conferenced, resulting in any incoming call being answered immediately as if a Talk Selector was activated. The problem was resolved changing the behavior of SIP Clients configured as Telephone Interfaces such that it never sends Interface Activation requests to any other client. This modification may potentially need to be reconsidered if there is later determined to be a requirement that a SIP Registered or Direct IP Trunk needs to initiate an Outbound Session on another SIP device with which it is placed in the same Party Line.

- VVM: Corrected a critical server issue when using the service which could prevent the OS from restarting properly after a reboot if no network connections were available. The problem was introduced by a modification in the v407 release which changed the behavior of the server to wait indefinitely for a network connection rather than terminate. Furthermore if the network connections did become available they would never be detected since the list of available network adapters was never updated.

- VVM: Corrected a critical server issue where two exact simultaneous login requests from the same client could be processed and accepted resulting in a server crash due to a suspected resource allocation conflict. Typically a second login request would be rejected with a duplicate login message however a design flaw in the check for a duplicate login left a window of several microseconds where if two login requests were processed at the exact same time it was possible for both to be accepted. The possibility of exact simultaneous login was previously thought to be practically impossible however it is suspected that a client side network issue of some sort is delaying a login request such that the subsequent login request as a result of the automatic login being enabled arrives at the server at the exact same instant in time.

- VVM: Changed all SIP Client thread debug messages to use debug filter.

- VVM: Changed the activity log message for a VCP/VDI logout from 'Client logout' to 'VCOM Client logout' to match the login message.

- VSA: Added support for selection of PCM 5ms codec when the client type is configured for Android, MAC or Linux. Also added support for the PCM 20ms codec however this selection is only available if the VSA is enabled for debugging.

- VVM: Corrected a problem with Geo Mapping where false 'socket' error messages were being generated after sending commands to the Geo Mapping server. The problem was as a result of waiting for a response from the Geo Mapping server for every command sent even though many of the commands would never generate a response from the server. The problem was resolved by waiting for a response from the Geo Mapping server only for those commands which generate a response.

- VVM: Corrected a problem where an invite from an external SIP client to an internal SIP Client would disrupt any already active SIP session. The problem was resolved by sending a '486 Busy Here' message to the external SIP client when a new server invite session is received and a SIP session is already active.

- VVM: Changed the default Auto-Answer Delay time from 5s to 0ms (the previous defacto default before adding the Auto-Answer Delay) as in most cases the incoming call should be answered as quickly as possible.

- VVM/VCP4Windows/VDI: Changed the messaging methodology for restoring talks and listens after a client disconnect and re-connect from a Restore Talk/Listen State bitmap to individual Restore Talk/Listen Selector messages which indicate the Label ID to restore. The Restore Talk/Listen Selector messages will be sent to all client versions 4.1 or greater. The main purpose of this change was to correct a long standing usability issue where active Talk/Listen selectors to a disconnecting VCP or VDI would be permanently deactivated even if the disconnect was due to a very brief loss of network connectivity. With this modification, if a client disconnects and then reconnects, and the disconnect duration is within the configured time frames specified for the restoration of talk or listen selectors after a disconnect, then any talk and listen selectors that were previously active to that client prior to its disconnect, will be automatically restored.

- VSA: Changed the Client Statistics Column Legend dialog to include an explanation of SA Jitter and RA Jitter.

- VVM: Corrected a problem with SIP Registered Trunks and SIP Direct IP Trunks where changing the SIP Client Login Name and Password (aka SIP User Name and Password) resulted in the SIP Client becoming non-functional until either the SIP Client Type was changed (and then changed back) or the SIP Server was restarted.

- VVM: Changed the VSA Domain Authentication validation to be case insensitive when comparing the provided login name and password against the configured login name. This was required due to the fact that the Windows OS user name itself can be case insensitive which intermittently results in a failure of the VCP to authenticate depending on how that user name was entered.

- VVM: Corrected a problem where the wrong internal SIP Client could be matched to a new external SIP Client registration. The problem only occurred if the user name of the external SIP Client was a sub-set of another already configured internal SIP Client user name (aka user name was 100 while 1000 also existed).

- VVM: Corrected a problem with internal SIP Clients configured to send Invites to external SIP Clients, where the audio would stop being received after a period of time. This problem only occurred when the external SIP Client used SIP Session Timers and that external SIP client insisted on sending the initial SIP Re-invite. The problem was due to sending an invalid SDP in response to the first SIP Re-invite as it was never expected that an Re-invite would be received when the Invite originated by the internal SIP Client.

- VVM: Added support to the VVM installer to include the default Sound Files from the '\Sound Files' directory.

- VVM: Added SNMP support to the VVM Installer so that when installing on Windows Server 2008 or Windows Server 2012 and the OS SNMP service exists, the SNMP service will be automatically configured to load the VCOM SNMP Agent DLL. If the SNMP service is already running, it is stopped first and then restarted afterward so that SNMP will be fully functional after the VVM installation in complete.

- VVM/VSA: Added support for a configurable RTP Keep Alive Method to allow for selection of 'either none' or 'Unknown RTP Payload Type'. In the case of 'Unknown Payload Type', the payload type can be specified. Previously, the RTP Keep Alive Method was permanently set to 'Unknown Payload Type' using 121 as the payload type.

- VVM/VSA: Added support for a configurable RTP Timeout so that active SIP sessions will terminate after the timeout period if no RTP packets are received. The RTP Timeout can only be specified if the SIP Device does not support silence suppression or supports the sending of the RTP Keep Alive Packets per RFC 6263. An RTP Timeout value of zero (the default value) will disable the RTP Timeout mechanism.

- VSA: Added support for a configurable SIP Registration Expiration Time to allow the expiration time to be modified specifically for SIP Registered Trunks. The default SIP Registration Expiration Time is 300 seconds which is recommended per RFC. For all internal SIP Client Registrations the registration expiration time remains unchanged at 60 seconds.

- VSA: Added support for SIP Session Timers and a configurable SIP Session Expiration Time to allow SIP Session Timers to be enabled or disabled. The default SIP Session Expiration Time is 0 which disables the

SIP Session Timer. When the SIP Session Expiration Time is non-zero, the SIP Session Timer per RFC 4028 will be enabled for the SIP Client which will result in the sending of periodic Re-Invites or Updates. By default the Session Mode is set to Caller Refreshes however this may be re-negotiated by the Callee.

- VVM: Corrected a problem where VCP/VDI/SIP clients would not appear in Geo Mapping if there were Unicode characters in the client's Selector Name. The problem was due to the fact that Unicode characters can never be used in a URL. To temporarily work around the problem the Selector Name is no longer sent to the Geo Mapping sever as the "Full Name" which is not actually a significant change since the current Geo Mapping implementation made no use of it. The problem still exists if there are Unicode characters in the client Login Name however there is no simple work around for this problem at this time other than to change the Login Name to not use Unicode characters.

- VVM: Corrected a problem where Geo Mapping would become non-functional after a "System Restart" initiated by the System Administration application. The problem was due the Geo Mapping connection state not being set to "Not Initialized" when the thread was terminated.

- VVM: Corrected a problem where no ringing sounds would be sent to any VCP or VDI listening to a SIP client that is not set to immediately auto-answer when an incoming call is received.

- VVM/VSA/VCP4Windows: Modified the Keep Alive implementation so that the server does not enable the Keep Alive mechanism for Android clients that do not support the Keep Alive as these clients would crash when the Keep Alive message was sent. The server only sends the Keep Alive when the time required by the network stack to send all login messages to the client exceeds the Keep Alive time since the application level only knows when those messages were placed on the stack and not when they were actually sent.

- VVM/VSA/VCP4Windows: Modified the Keep Alive implementation so that the sender's Auto Disconnect Check time is sent in the message detail. If the client receives an Auto Disconnect Check Time from the server that is different from the value it is using, it will use the server's Auto Disconnect Time as its new value. In this way it will be possible, if it is deemed necessary, to extend the default 3.5s client Auto Disconnect Time for connection stability to accommodate the added latency expected over satellite.

VVM/VSA v4.0.8 - 11/24/2014 - Private Release

- VVM: Removed permanent activation of Label In-Use Indication on session activation for SIP Registered Trunks and Direct IP Trunks.

- VVM: Added SNMP support for Virtual Matrix Shutdown message.

- VVM: Corrected message size used for sending SMNP Failover message.

- VVM: Added delay after sending SNMP Virtual Matrix Shutdown message to ensure the SMNP Agent DLL has time to process the message.

- VVM: Changed SNMP initialization messages sent to Activity Log.

- VVM: Corrected a problem when responding to a SIP re-Invite sent from a callee, where the SDP was erroneously being regenerated resulting in a loss of received audio due to the callee correctly sending audio to the changed RTP port however the server was still listening for audio on the previous RTP port.

- VVM/VSA: Added support for a configurable SIP Registration Expiration Time to allow the expiration time to be modified specifically for SIP Registered Trunks. The default SIP Registration Expiration Time is 300 seconds which is recommended per RFC. For all internal SIP Client Registrations the registration expiration time remains unchanged at 60 seconds.

- VVM/VSA: Added support for SIP Session Timers and a configurable SIP Session Expiration Time to allow SIP Session Timers to be enabled or disabled. The default SIP Session Expiration Time is 0 which disables the SIP Session Timer. When the SIP Session Expiration Time is non-zero, the SIP Session Timer per RFC 4028 will be enabled for the SIP Client which will result in the sending of periodic Re-Invites or Updates. By default the Session Mode is set to Caller Refreshes however this may be re-negotiated by the Callee.

- VVM: Changed the handling of SIP Registered Trunks to detect a registration failure, display that failure in the Activity Log and re-initiate another registration attempt. Prior to this change, once a registration failure occurred, there was no further attempt to re-register the SIP Client which required a user intervention to correct by disabling and then re-enabling the SIP Client.

- VVM: Corrected a memory leak with the handling SIP Clients where several classes that were created, were never being deleted. The memory leak occurred every time the external SIP Client Registration was removed or the SIP Client was disabled.

- VVM: Changed the VCP Domain Authentication validation to be case insensitive when comparing the provided login name and password against the configured login name. This was required due to the fact that the Windows OS user name itself can be case insensitive which intermittently resulted in a failure of the VCP to authenticate depending on how that user name was entered.

- VVM/VSA/VCP4Windows: Added support for a Keep Alive mechanism on the TCP connection for all clients in order for them to work correctly over a satellite connection. When operating over satellite a flaw in the current communication protocol was exposed which can result in the inability of a client to complete the login sequence. During a client login sequence, the client does not send any packets to the server while it waits for all data to be received. However, over a slow connection such as a satellite, the login sequence can take longer to complete than the sever side connection timeout of 3.5 seconds resulting in the server erroneously terminating the connection. To correct this problem all clients must implement the transmission of a Keep Alive message if there has been no other packet sent for 1.25s.

- VSA: Corrected support for passing the IP Address and Port as well as the Login Name and Password on the command line. This feature was previously implemented however was not functional due to some critical code that was somehow omitted during final integration.

- VSA: Corrected a problem with the parsing of the SIP server and STUN server IP Address fields where a random IP port would be assigned when the IP address was entered without a port number. The problem was due to incorrectly using the converted parameter for the port number even though the return value indicated the parameter was not converted.

- VVM: Corrected a problem when running the SIP server on a port number other than 5060 where SIP client invite would receive a "relaying forbidden" message. The problem was resolved by including the port number in the SIP client's AOR.

Explanation of acronyms used

VVM = VCOM Virtual Matrix (includes both service and application)

VSA = VCOM System Administration

VCP = VCOM Control Panel (All Platforms including iOS, Java, Android and Windows)

VCP4iOS = VCOM Control Panel for Apple iOS

VCP4Windows = VCOM Control Panel for Windows

VDI = VCOM Device Interface

SNMP = Simple Network Monitoring Protocol