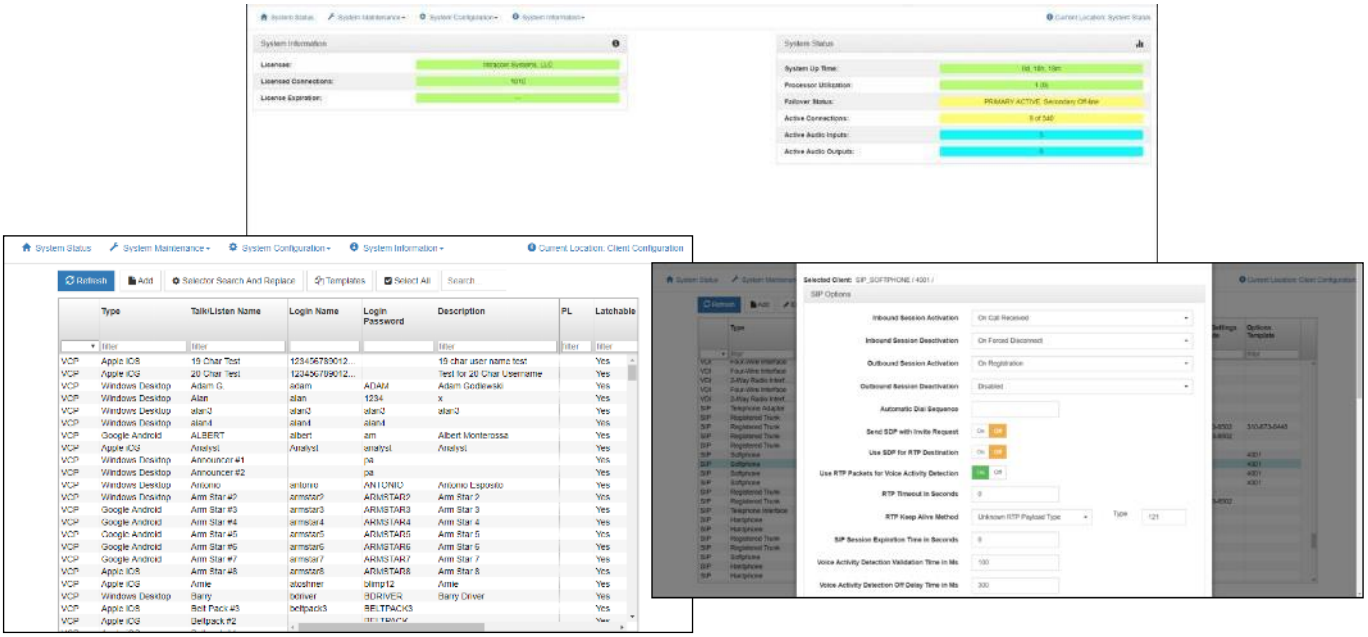
VLINK System Administrator User Guide



U.S. Patents # 8660039, 8942141, 9357077

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INTRODUCTION

ABOUT VLINK

VLINK is configured via the client-side browser based System Administration application, which allows for dynamic configuration from any workstation. This application can also be run on the server hosting the VLINK Virtual Matrix and is installed automatically along with it.

This document provides information on how to install, configure, and use the VLINK System Administration application.

SYSTEM REQUIREMENTS

The only requirement for running the VLINK System Administration application is a web browser. Supported browsers include Google Chrome and Microsoft Edge.
For VLINK Virtual Matrix system requirements please refer to the VLINK Virtual Matrix User Guide.

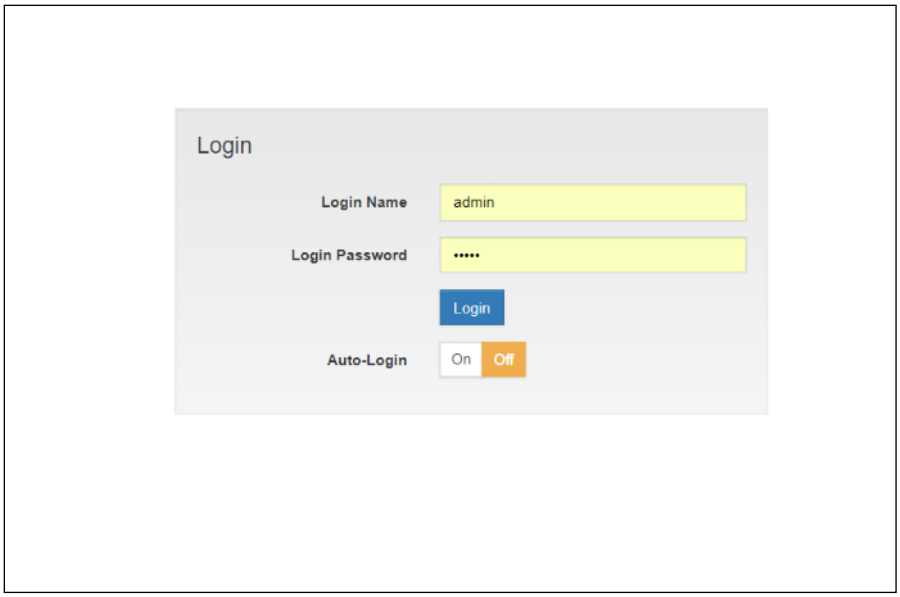
LOGGING IN

If you are using the VLINK System Administration application from the same machine VLINK is installed on:

After VLINK is installed on the machine, a shortcut will be placed on the desktop called 'VLINK System Administration.' Click on the shortcut. A browser window will appear. The VLINK System Administration application can also be accessed by typing 'localhost' into the URL box and pressing enter.

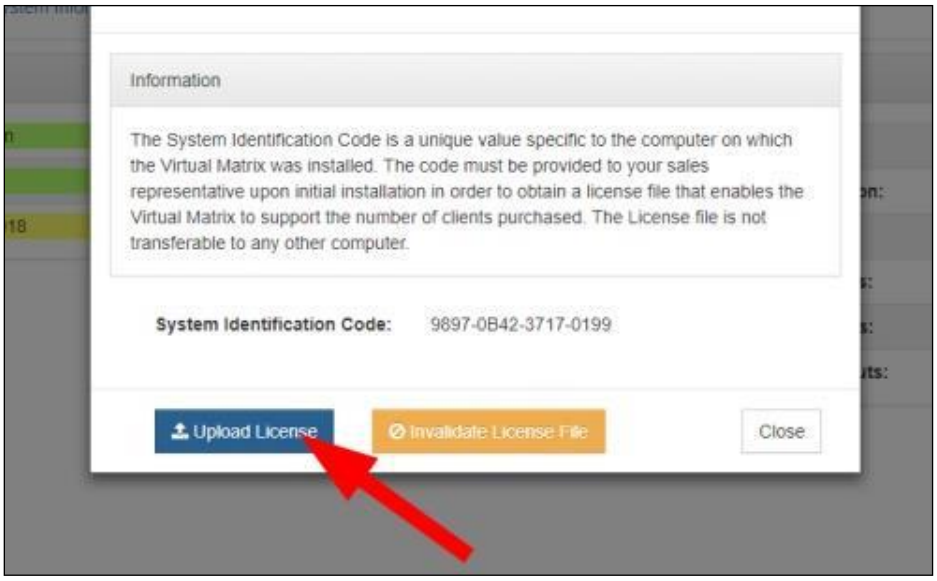
If you are using the VLINK System Administration application from a different machine:

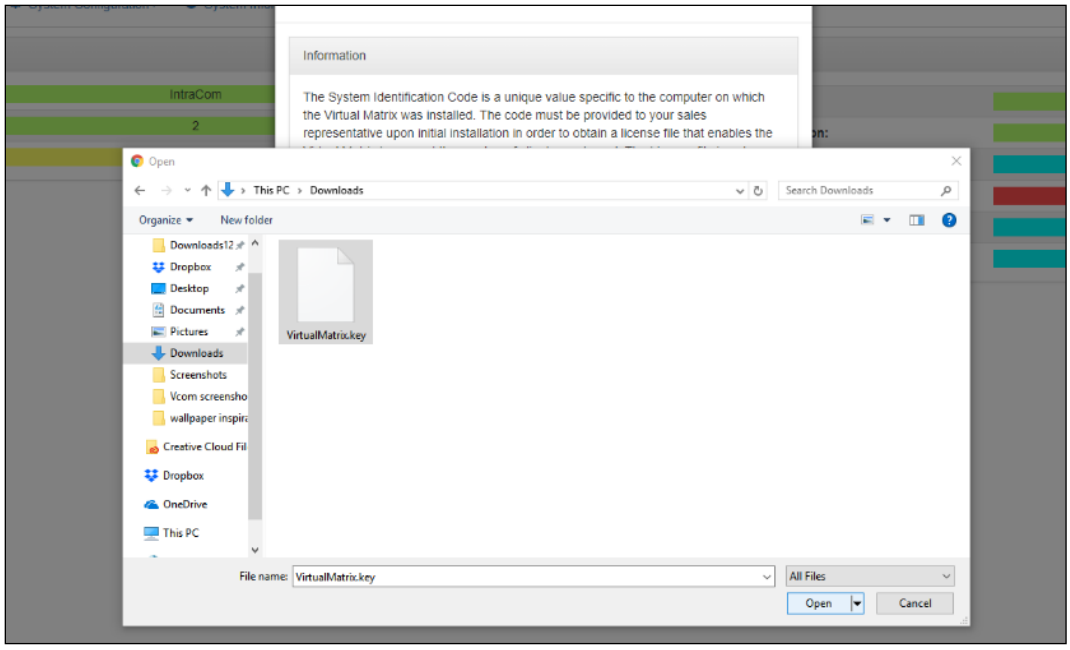
Open a browser window and type the local IP address of the machine running VLINK into the URL box and press enter.



The default master 'Login Name' and 'Login Password' are both 'admin'. Once logged in, credentials can be changed. After logging in you will be prompted to upload a license file. You will be shown your 'System Identification Code,' which is a unique value specific to the computer on which the Virtual Matrix was installed. This code must be provided to your sales representative upon initial installation to obtain a license file that enables the Virtual Matrix to support the number of clients purchased. The license file is not transferable to any other computer. When the license file is received, click 'Upload License'.

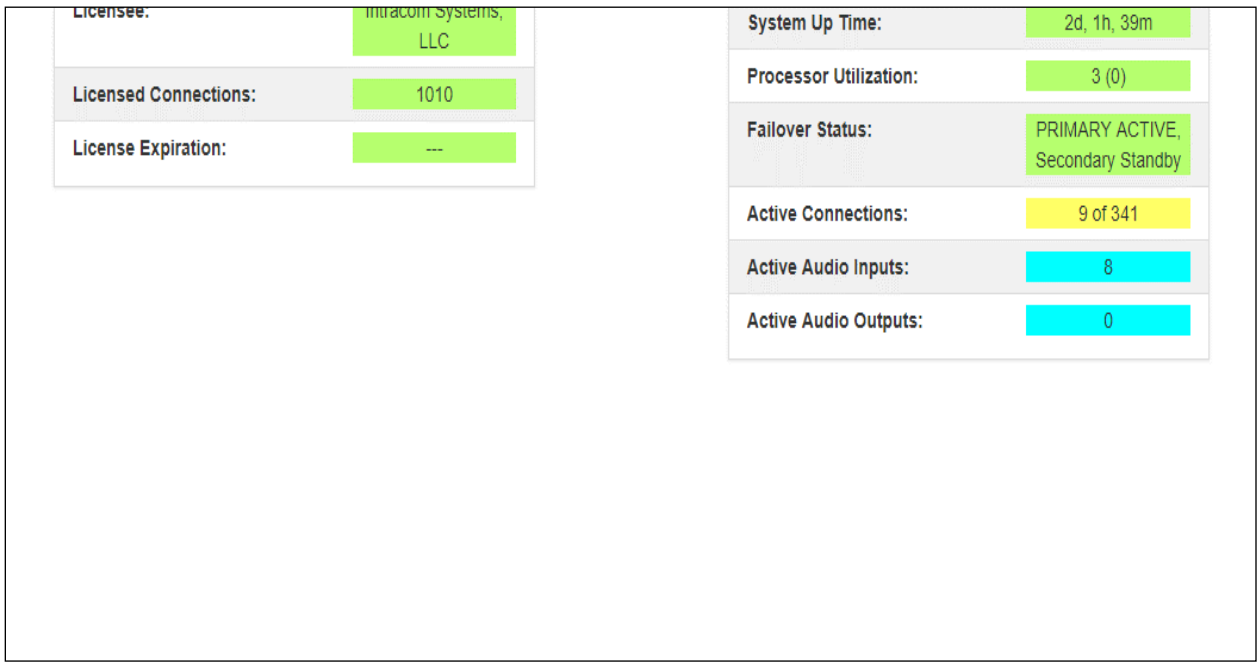
Note: When utilizing multiple NIC (Network Interface Card) devices, please note that in the event of a change in network connection, the license key will no longer work.





Locate the file and select ‘Open’ in the Windows Open File dialog box.

OPERATION



SYSTEM INFORMATION

System Information fields are color-coded. Red indicates a problem; Yellow a potential problem; and Green normal operations. Blue fields are informational.

The 'System Information' area in the upper, left-hand side of the System Administration main window displays the name of the licensee, the number of clients licensed, and expiration date. 'License Expiration' will typically display '---' which means a system has been purchased and a perpetual license has been granted. Demo, trial, and rental systems will display a numerical value in days reflecting the limited duration for which a license has been granted.

SYSTEM STATUS

System Status fields are color-coded. Red indicates a problem, Yellow a condition that needs attention, and Green normal operations. Blue fields are informational.

The 'System Status' on the right side of the System Administration main window displays general system metrics.

System Up Time: Displays how long the VLINK Virtual Matrix has been running in days, hours, and minutes.

Processor Utilization: Displays CPU utilization of the server or PC hosting the VLINK Virtual Matrix.

Failover Status: Displays the operational status of the Primary and Secondary servers when Failover is licensed and configured.

Active Connections: Displays how many clients are connected to the Virtual Matrix at any given moment out of how many clients are configured. Clients include VLINK Control Panels, VLINK Device Interfaces, and SIP Devices.

Active Audio Inputs: Displays how many active audio channels are being streamed from clients into the Virtual Matrix at any given time.

Active Audio Outputs: Displays how many active audio channels are being streamed out of the Virtual Matrix to clients at any given time.

SYSTEM CONFIGURATION

SSL CONFIGURATION

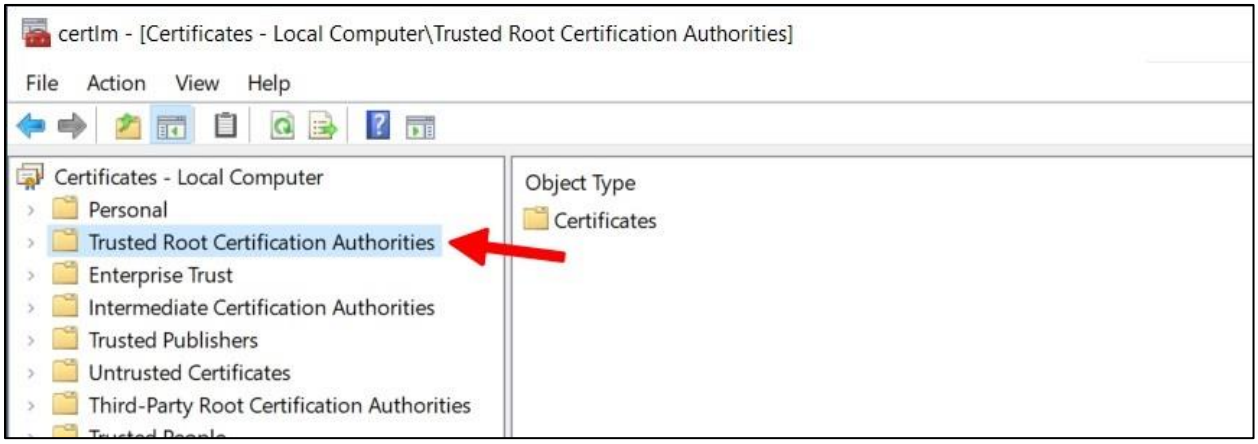
For secure communications and access to the Microphone and Camera in Google Chrome, an SSL certificate is required. If you are a VLINK Cloud customer, SSL is already enabled on your server and you may skip this portion of the guide. VLINK comes with a self-signed certificate preinstalled. When accessing the system, Google Chrome will display a warning message that the site is insecure. It is recommended to add the certificate to your computer's certificate store however this is not required for VLINK to function properly.

If your server is to be used from the public internet, it is recommended to generate an SSL certificate though a Certificate Authority. If you are using Google Chrome and only require access to the Microphone and Camera, it is possible to configure Google Chrome to allow access to the Microphone without the need for a SSL certificate (see Configuring Google Chrome to treat VLINK as secure).

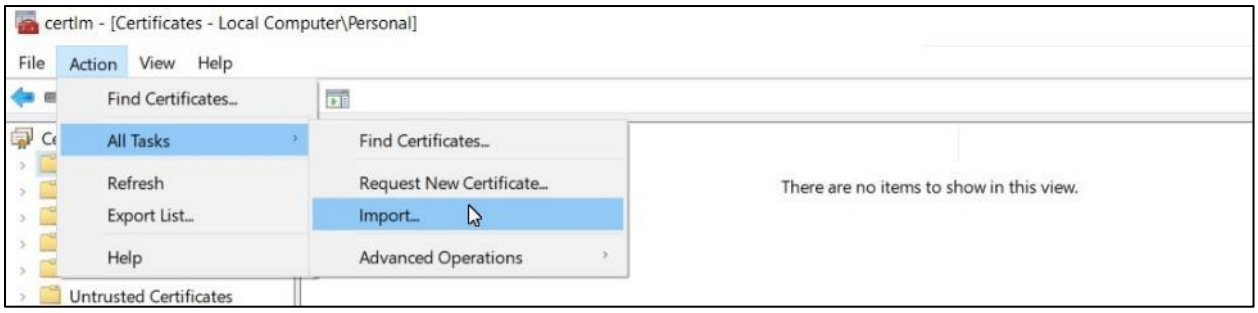
Note: The VLINK WebRTC app for Android does not require SSL

Adding the self-signed certification to the computer's certificate store:

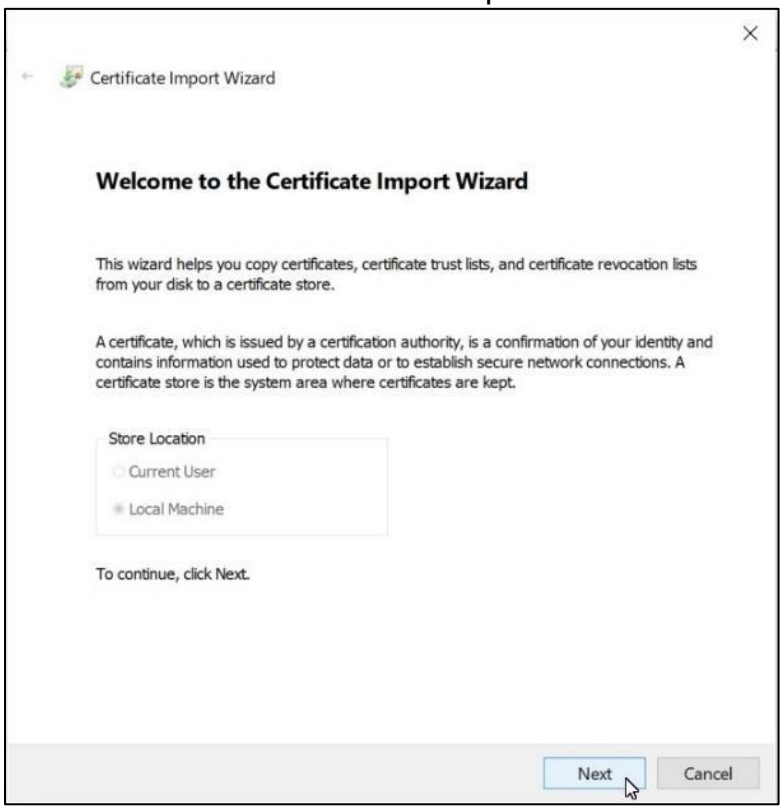
Note: The SSL certificate must be installed on every computer that will access VLINK to avoid browser warnings.



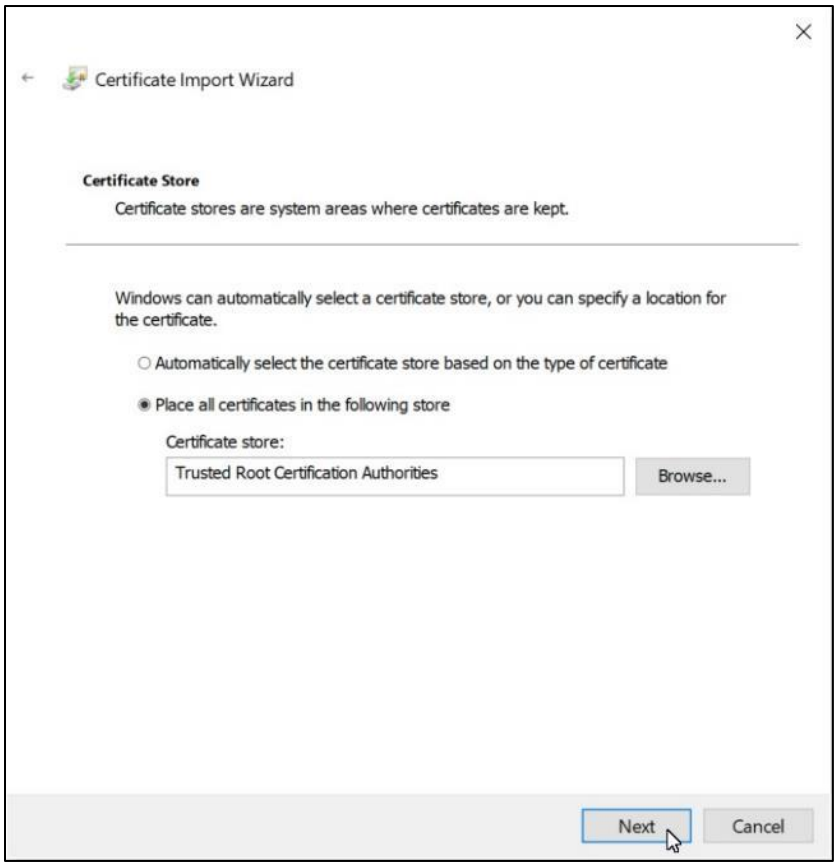
Launch the windows certificate manager by entering "certmgr.msc" in the start menu. Select the "Trusted Root Certification Authorities" folder.



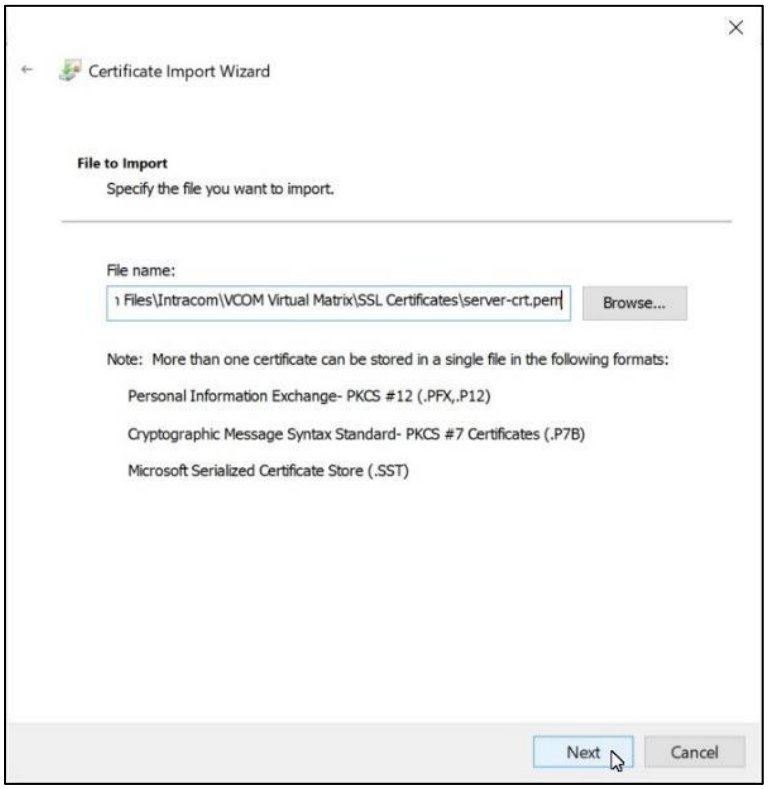
From the action tab select All Tasks then select Import.



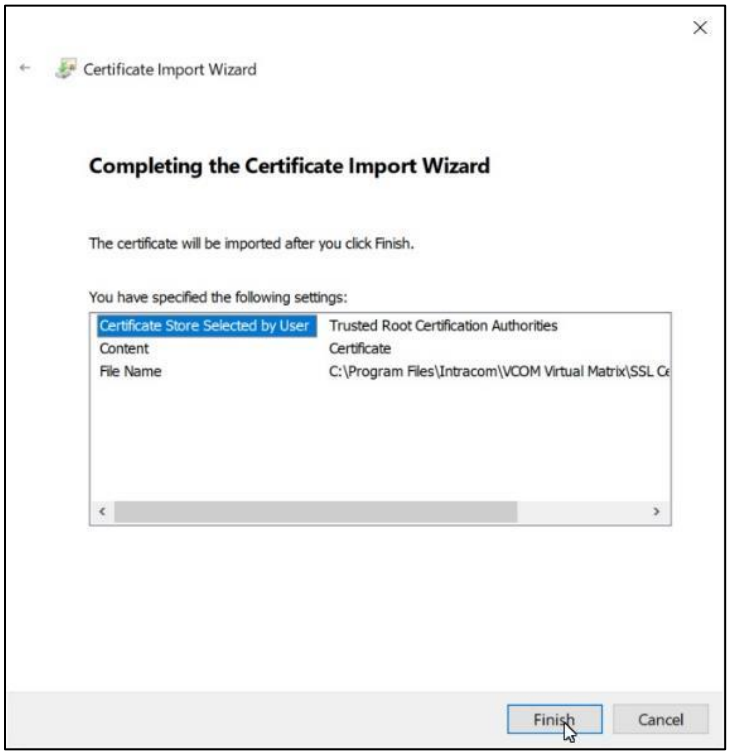
Click Next.


Certificate Import Wizard
Certificate Store
Certificate stores are system areas where certificates are kept.
Windows can automatically select a certificate store, or you can specify a location for the certificate.
Automatically select the certificate store based on the type of certificate
Place all certificates in the following store
Certificate store:
Trusted Root Certification Authorities
Browse...
Next
Cancel

Click Next.

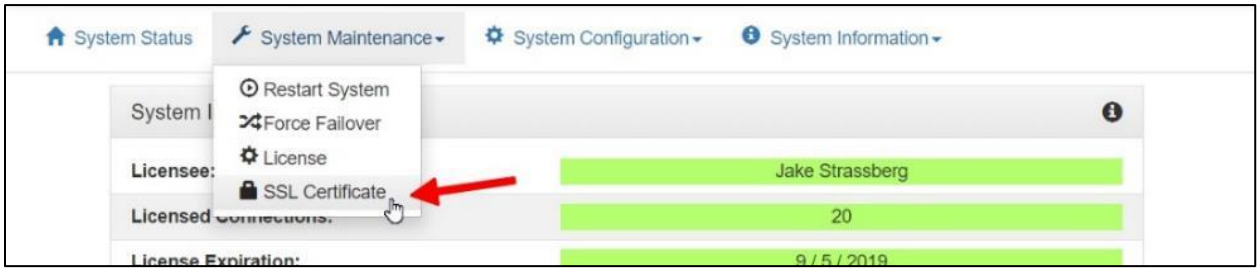

Certificate Import Wizard
File to Import
Specify the file you want to import.
File name:
ı Files\Intracom\VCOM Virtual Matrix\SSL Certificates\server-crt.pem
Browse...
Note: More than one certificate can be stored in a single file in the following formats:
Personal Information Exchange- PKCS #12 (.PFX,.P12)
Cryptographic Message Syntax Standard- PKCS #7 Certificates (.P7B)
Microsoft Serialized Certificate Store (.SST)
Next
Cancel

In the File name field enter “C:\Program Files\Intracom\VLINK Virtual Matrix\server-crt.pem”. Your certificate location will be different if VLINK Virtual Matrix was not installed in the default directory. Click Next.

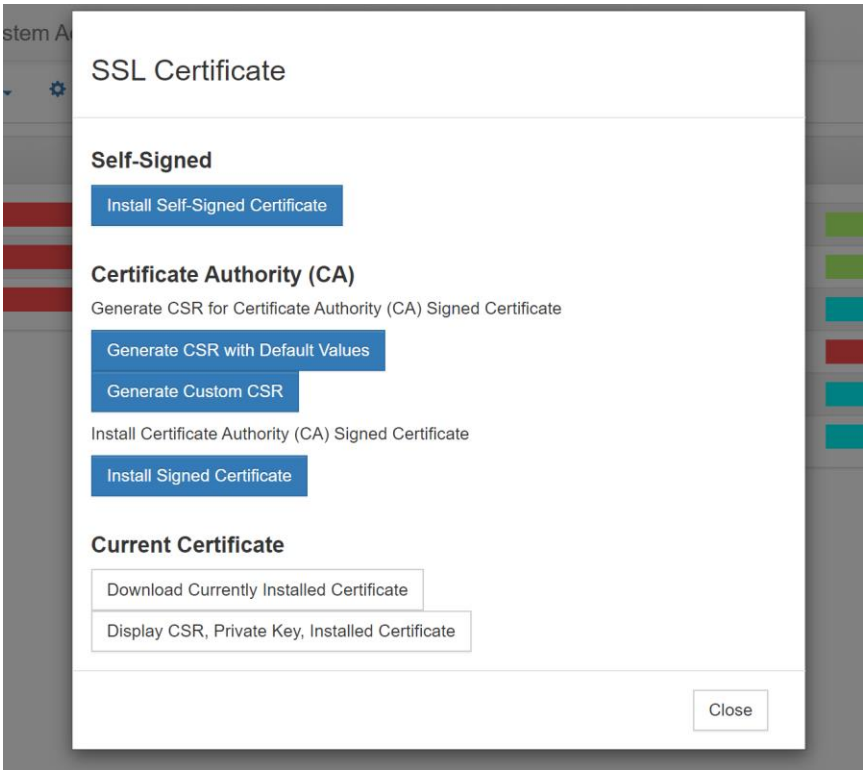


Click Finish then restart Google Chrome.

Generating a self-signed SSL certificate:



From the System Maintenance menu click SSL Certificate.



Install Self-Signed Certificate
Generates and installs a self-signed certificate onto the VLINK server.

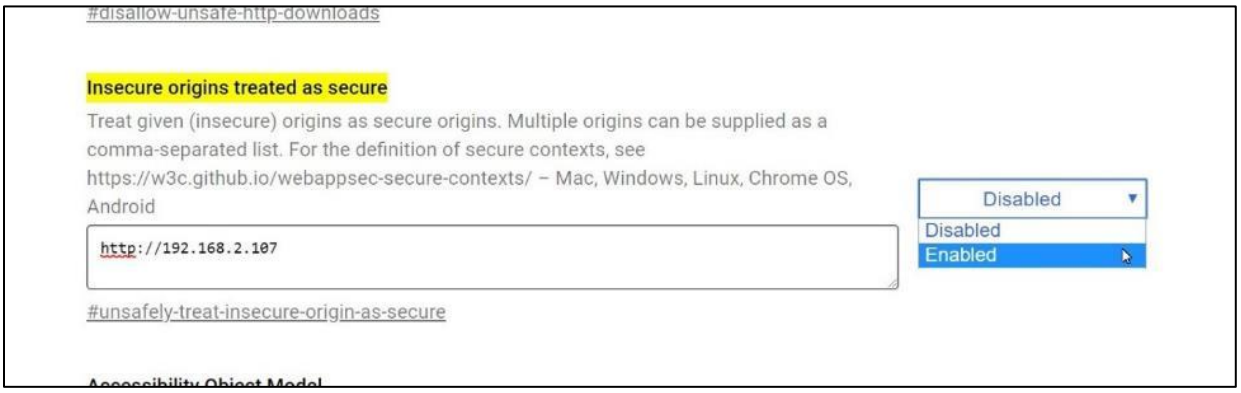
Generate CSR with Default Values
Prompts the user for the Domain Name(s) with which to generate a CSR to send to a Certificate Authority.

Generate Custom CSR
Prompts the user for Domain name(s), Organization Name, County Name, and State/Province. These values are used to generate a CSR to send to a Certificate Authority.

Install Signed Certificate
Installs a certificate generated by a certificate authority.

Configuring Google Chrome to treat VLINK as secure (Not recommended):

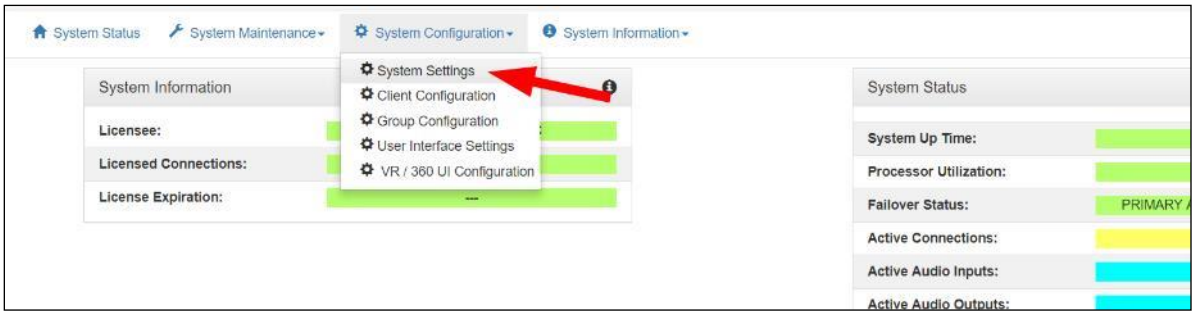
In the address bar enter “chrome://flags/#unsafely-treat-insecure-origin-as-secure” then press enter.

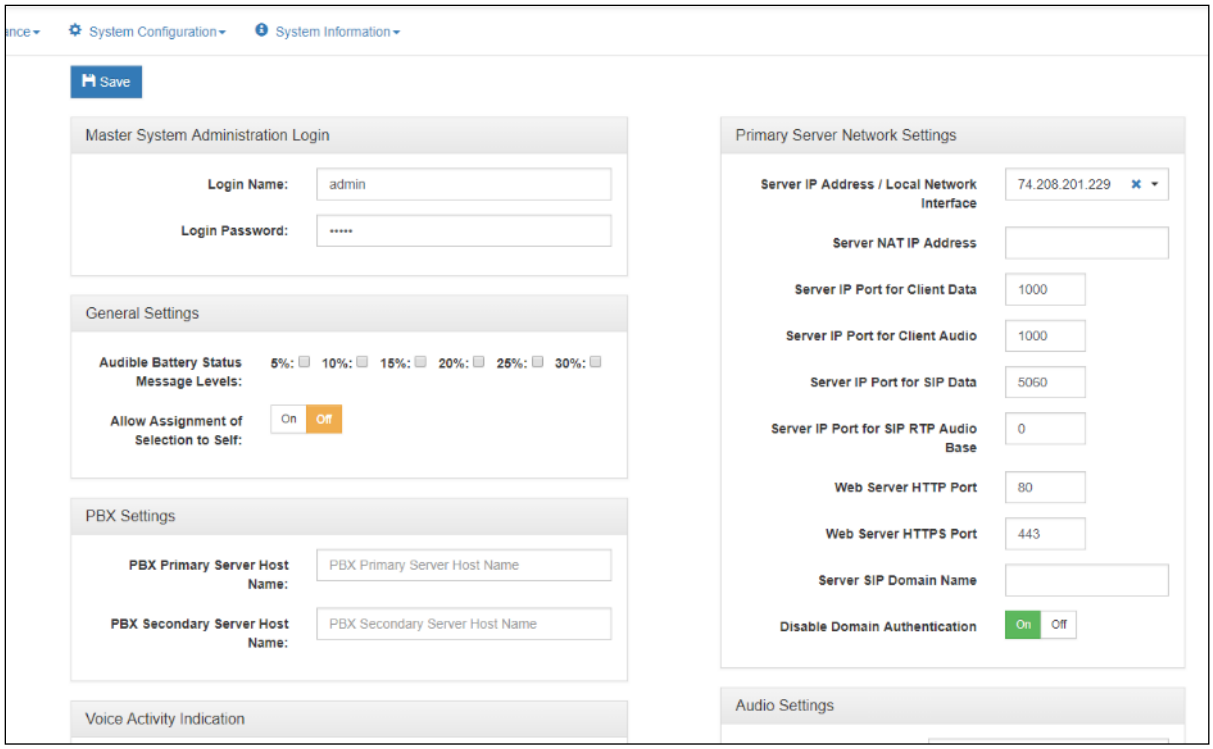


Type in “http://” followed by the local IP address of your server. On the right drop-down menu select “Enabled” then restart Google Chrome. Redo these steps on every computer that will access the Control Panel.

VLINK System Settings

The ‘System Settings’ option can be found in the drop-down menu under the ‘System Configuration’ tab.





Master System Administrator Login

- Displays and allows you to edit the master system administrator login name and login password

Primary Server Network Settings

- Enter the 'Server IP Address,' which is the IP address of the server hosting the Virtual Matrix

- Enter the 'Server NAT IP Address', which is the public IP address that is used to route data to the Server's IP address within the LAN. This is only used when Network Address Translation (NAT) is being used on your LAN

- Enter the Server IP Port for VLINK Client Data under 'Server IP Port for VLINK Client Data / Audio' (default: port 1000), which controls the TCP/IP port that all client side Control Panels and Device Interfaces use to transport data to the Virtual Matrix. The system has no restriction other than reserved ports. If the Virtual Matrix is behind a firewall and external access is required, a Port Forwarding entry must be added to route all traffic on this port to the internal Virtual Matrix IP address

- Enter the Server IP Port for Client Audio under 'Server IP Port for VLINK Client Data / Audio' (default: port 1000), which controls the UDP port that all client-side Control Panels and Device Interfaces use to transport audio to the Virtual Matrix. The system has no restriction other than reserved ports. If the Virtual Matrix is

behind a firewall and external access is required, a Port Forwarding entry must be added to route all traffic on this port to the internal Virtual Matrix IP address

- Enter the Server IP Port for SIP Data under 'Server IP Ports for SIP Data / RTP Audio Base' (default 5060) to define the IP Port for the integrated SIP Server. In general this value will never be changed as this is an industry standard port number. However, the value must be changed if multiple VLINK Virtual Matrix instances are to be run on the same physical computer

- Enter the Server RTP Audio Base under 'Server IP Ports for SIP Data / RTP Audio Base.' By default, when set to zero all SIP RTP (Real-time [Audio] Protocol) sessions will establish the IP port number randomly in the range of 10000-42767. In many situations this is perfectly adequate. However, if the audio must transverse a firewall, it is not practical or safe to open such a large range of addresses. As such, by specifying an RTP Audio Base port the system will assign IP ports sequentially upward from the base port. Once an IP Port is assigned to a SIP client, it will never change unless the Base Port is itself changed

- Enter the 'Server SIP Domain Name' to define the optional SIP Domain for the integrated SIP Server. If the SIP Domain is specified, it can be used as the SIP Proxy Name and the Registrar Name when configuring SIP clients. Regardless of whether the SIP Domain is specified, the Virtual Matrix IP Address can always be used as the Proxy Name and the Registrar Name

Secondary (Failover) Server Network Settings

If you wish to set up a failover server for a second Virtual Matrix, and have purchased 'Redundancy', enter the relevant values. In an instance of a hardware or connectivity failure client applications will reconnect to the failover server. Refer to Section 6 of the VLINK Virtual Matrix User Guide for a detailed description of the failover capability

- Enter the 'Server IP Address,' which is the IP address of the server hosting the Virtual Matrix

- Enter the 'Server NAT IP Address', which is the public IP address that is used to route data to the Server's IP address within the LAN. This is only used when Network Address Translation(NAT) is being used on your LAN

- Enter the Server IP Port for VLINK Client Data under 'Server IP Port for VLINK Client Data / Audio'

- Enter the Server IP Port for Client Audio under 'Server IP Port for VLINK Client Data / Audio'

- The 'Server IP Port for Failover Data' is the port that is used for the two Virtual Matrix servers to communicate with each other

Audio Settings

- The ‘Audio Mix Sample Rate’ controls the maximum sampling rate supported by the Virtual Matrix and thereby dictates the maximum fidelity for all Client connections. There are 3 possible settings: Narrowband (8 KHz), Wideband (16 KHz), and Ultra-Wideband (32 KHz), which is the default setting. Narrowband is approximately the same fidelity as a phone connection while Wideband more closely approximates the fidelity of a professional analog, hardware-based Intercom system. Clients by default will be set to the System's Audio sampling rate. However, the client audio sampling rate can be specified at a lower rate but never at a higher rate. Higher audio sampling rates have more significant requirements both in computational speed and network bandwidth so careful consideration must be made when choosing this setting with respect to server hardware and network infrastructure

- The ‘Audio Output Level Gain (Post-Mix)’ adjusts the output level from the VLINK Virtual Matrix to the Control Panels and Device Interfaces in 6dB intervals 3 times to a maximum of 18dB. This is a global change throughout the system

Voice Activity Indication

- Use the ‘Voice Activity Indication Color’ feature to change the text color and background color used to indicate voice activity on a given selector. The system interchanges its base colors (yellow text / navy background) with the selected activity indication colors (variable). For typical applications, the default color provides a subtle but noticeable indicator. For some applications, such as maintenance panels for hoot and holler systems, a more pronounced indicator (black text / white background) is generally required
- [Default: White text / light navy background]

Failover Settings

- Set the amount of time to delay the failover from coming online in place of the Primary Server

- Automatic Failback is used to define when to restore the Primary Server. Choose the setting that best fits your procedures

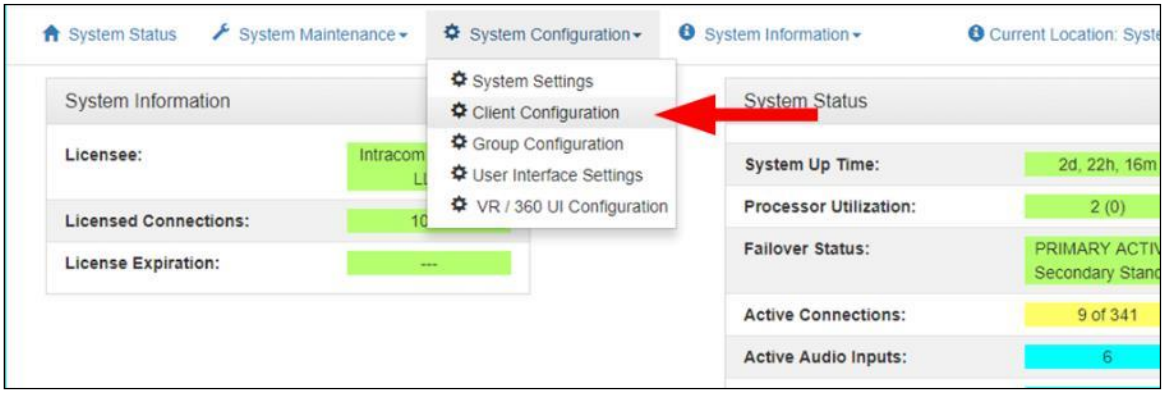
Geo Mapping Server Network Settings

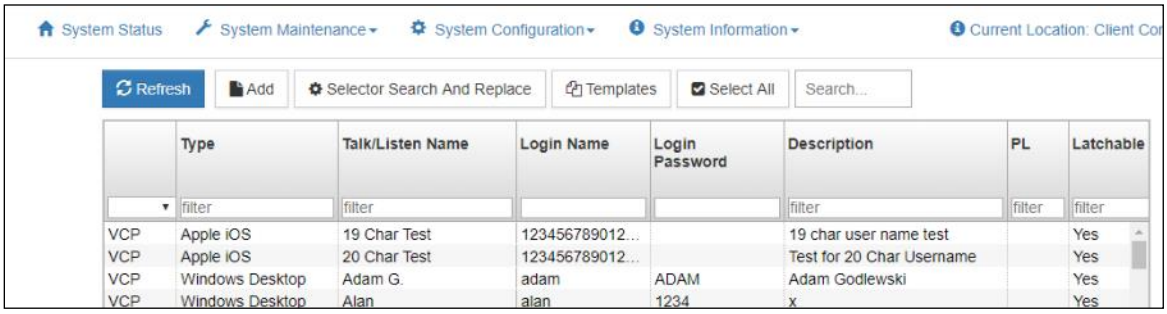
- Enter the Geo Mapping Server IP Address which is the IP Address of the Geo Mapping Server

- Enter the Geo Mapping Server Port number which is the port number the Geo Mapping Server uses for connections

VLINK Client Configuration

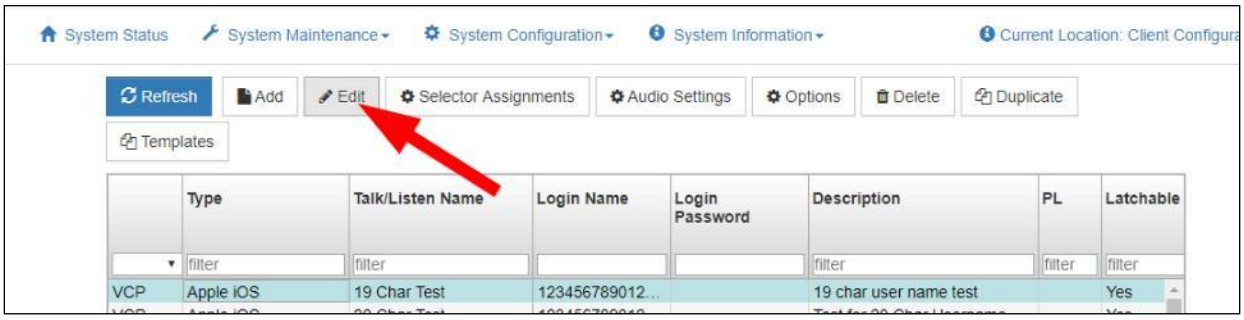
The Client Configuration option can be found in the drop down menu under the 'System Configuration' tab.

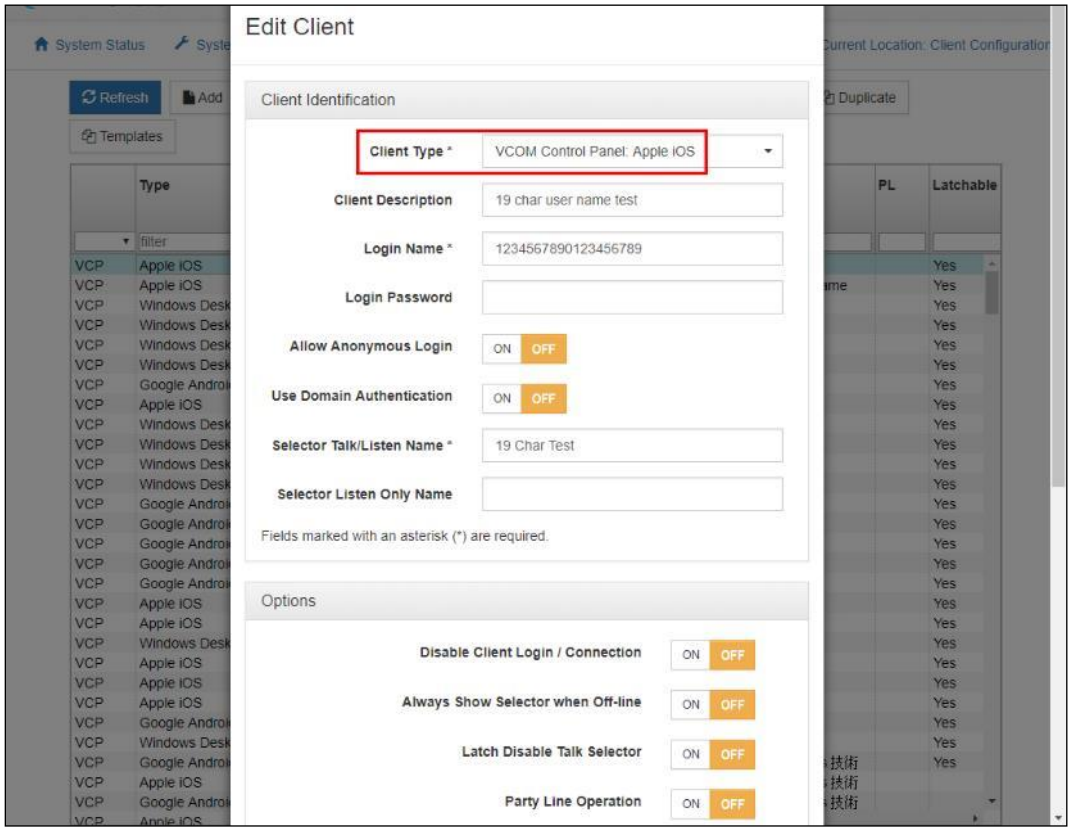




The upper section of the 'Client Configuration' window displays all configured users and devices, log in names, passwords, selector labels, client type, and if the given channel is set as a party line.

Click on a client description name and then 'Edit' to change a parameter, 'Delete' to delete a user or device, 'Add' to create a new client, or 'Duplicate' to copy a client's settings to a new client.





Client Identification

Client Type: Specifies the type of Client that allows the system to modify the internal operational behavior for proper operation of specific, such as VLINK Control Panels, Device Interfaces, Two-Way radios, and SIP devices. SIP device options are:

- 'SIP Device: Analog Telephony Adapter (FXS)': also referred to as an "ATA" enables an analog telephone to function as an IP phone, sitting between your network and analog telephone.

- 'PSTN Telephone Interface (FXO)': VoIP gateway enables analog phone lines or T-1s with phone service into VLINK

- 'SIP Device: Softphone': enables connection to one of the many available softphones designed for PCs, Macs, and handhelds, including the iPhone.

- 'SIP Device: Hardphone / Stand alone IP Phone': enables a connection to a physical IP phone to VLINK.

- 'SIP Device: Intersystem/PBX Direct IP Trunk': enables a connection between PBX/SIP Servers and VLINK using IP settings.

- ‘SIP Device: Intersystem/PBX Registered Trunk’: enables a connection between PBX/SIP Servers and VLINK using Registration of clients.

- ‘SIP Device: Four Wire Interface’: enables connection of a SIP based Four Wire Interface to transport audio into VLINK.

Client Description: The description given to identify the client that is used exclusively in the System Administration application. This allows a complete description of a user (typically first and last name) that cannot otherwise be assigned to the selectors due to space restrictions.

Login Name: The name assigned to a user or device and used to login a Control Panel or Device Interface to the Virtual Matrix. Select ‘Allow Anonymous Login’ to allow a user to login a Control Panel by entering any login name he or she chooses followed by the designated password. The chosen login name will appear on the selector.

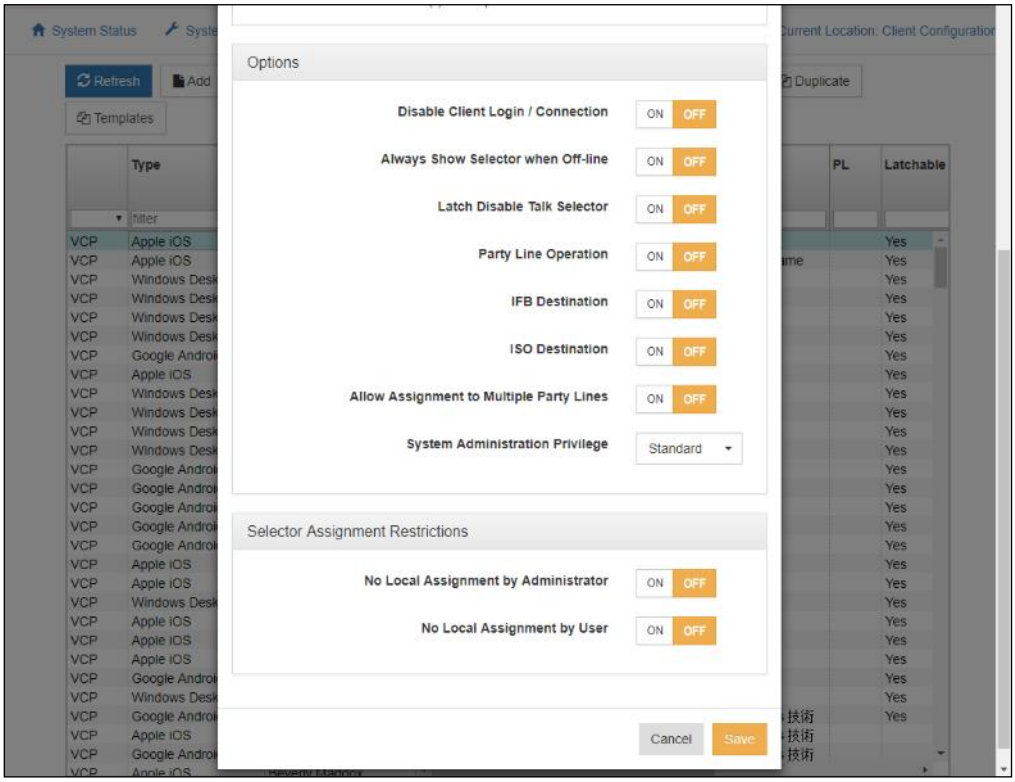
Login Password: The password assigned to a user or device and used to login a Control Panel or Device Interface to the Virtual Matrix

Selector Talk/Listen Name: The alphanumeric identifier that appears on Control Panel ‘Talk only’ and ‘Talk/Listen’ selectors.

Selector Listen Only Name: The alphanumeric identifier that appears on Control Panel ‘Listen only’ selectors. This is generally only assigned when a client has split functionality for the audio input and output as with a Program Feed input and IFB output.

Use Domain Authentication (LDAP): This is used when a Client is outside the LAN, **and** your LAN uses authentication to provide secure connections to remote users. If this option is enabled VLINK queries the OS to retrieve the already authenticated Windows user name. It then uses that name along with an encrypted password unique to that client to log in to the VVM. The Use Domain Authentication option must also be checked on the Control Panel.

Note: This feature is only available on the non web-based VLINK Control Panel.



Options

Disable Client Login/Connection: This allows the System Administrator to “turn-off” certain clients when they are not needed to login to the system.

Always Show Selector when Off-line: Specifies that the selector for this Client will be visible even if off-line on a VLINK Control Panel that is configured to hide the off-line selectors. It is generally used for VLINK Device Interface clients that should, under normal operation, never go off-line thus allowing VLINK Control Panel operators to more easily notice an unanticipated disconnect event.

Latch Disable Talk Selector: Select to operate associated selector as a momentary, meaning that an audio path will only persist as long as the selector is clicked and held.

Party Line Operation: This specifies that a given client operates like a Party Line. This means that anyone talking to that client will also talk to anyone listening to that client and anyone listening to that client will also hear everyone talking to that client.

IFB Destination: Designates a client as an IFB Destination that causes the system to interrupt any assigned listen or program feeds to the destination when a Control Panel initiates a talk path to the destination. This setting is typically used with on-air talent who need to be constantly monitoring the on-air program feed but periodically take cues from the director or producer.

ISO Destination: Designates a client as an ISO Destination that causes the system to interrupt any assigned listen or program feeds to the destination when a Control Panel initiates a talk path to the destination and automatically activates a return talk path from the destination back to the Control Panel. Additionally, the talk paths in both directions are isolated so that the conversation is kept private. This setting is typically used with cameras when the director or producer needs to isolate a particular camera from the camera PL to provide private instruction.

Allow Assignment to Multiple Party Lines: Allows a client to be remotely assigned to Party Lines through Party Line Assignment Mode.

Standard System Administration Privileges: Gives System Administrator privileges to the client.

Master System Administration Privileges: Gives Master System administrator privileges to the client.

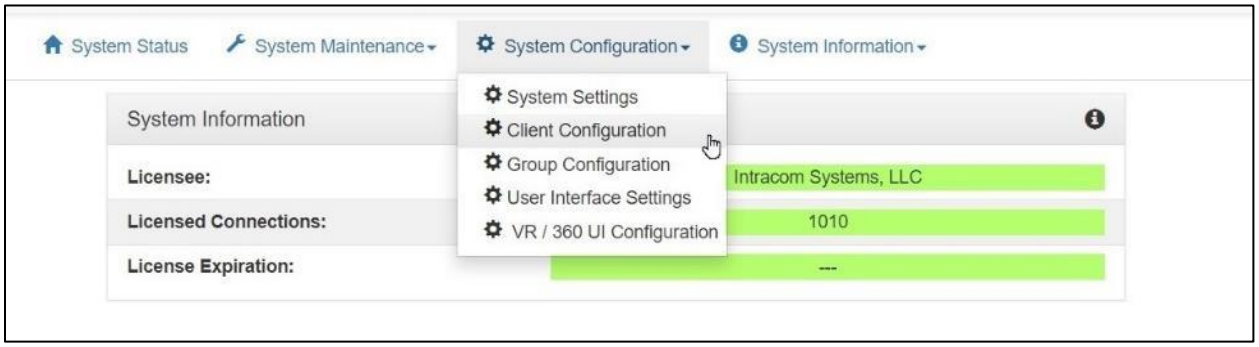
Selector Assignment Restrictions

- No local Assignment By Administrator: This prohibits local assignments to be made by an Administrator. Assignments can only be done through the System Administration application
- No Local Assignment By User: This prohibits local assignments to be made by a User. Assignments can only be made by an administrator from the System Administration application

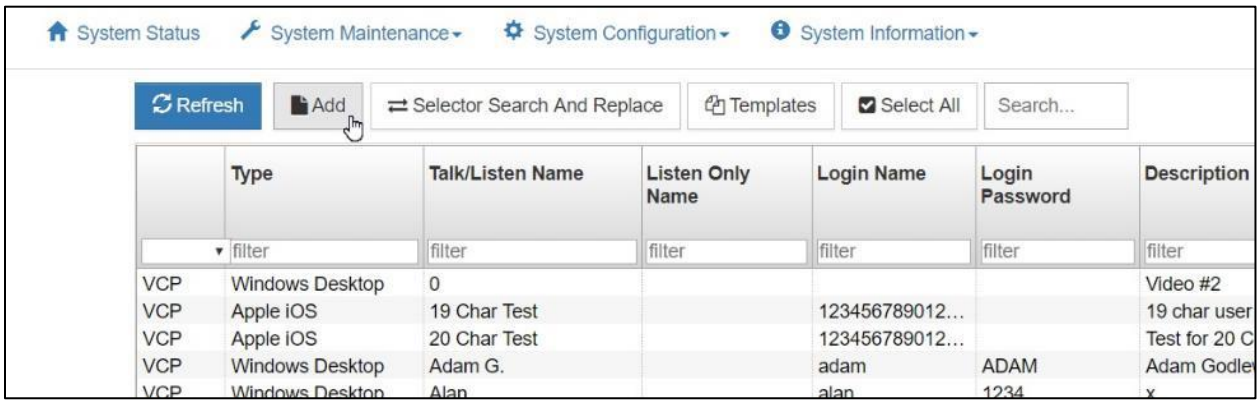
ADDING A SIP LINE TO VLINK

Intracom Systems does not provide SIP lines. The lines must be purchased through a 3rd party phone line provider. Once a SIP line is purchased you will be provided a phone number, username, PBX IP, and you will set up a password.

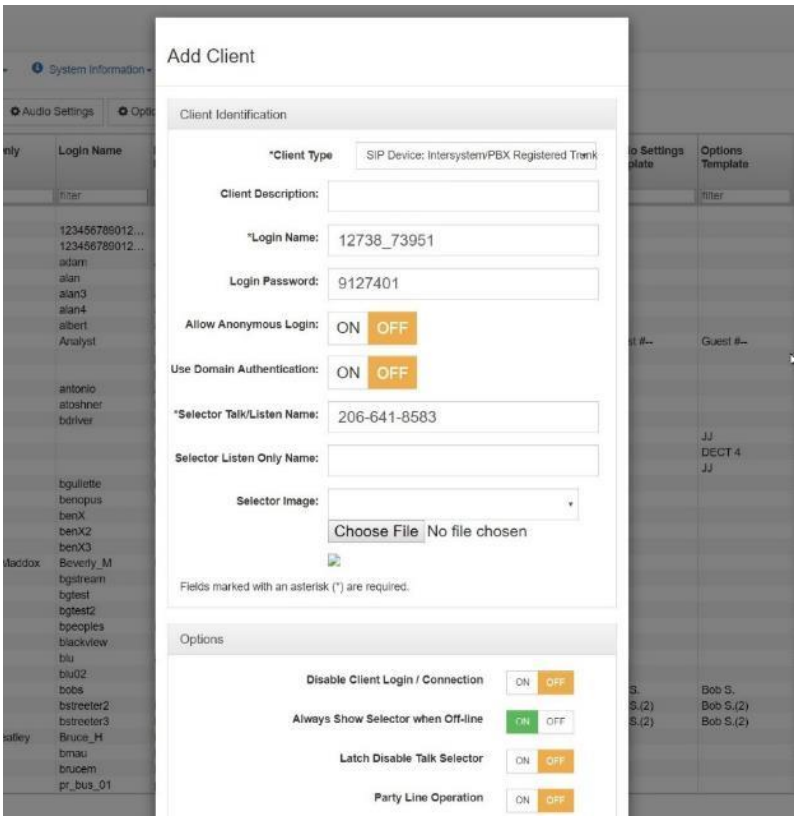
Note: If you are connecting a Cubic / Vocality RoIP device to VLINK we have dedicated documentation available as a separate guide.



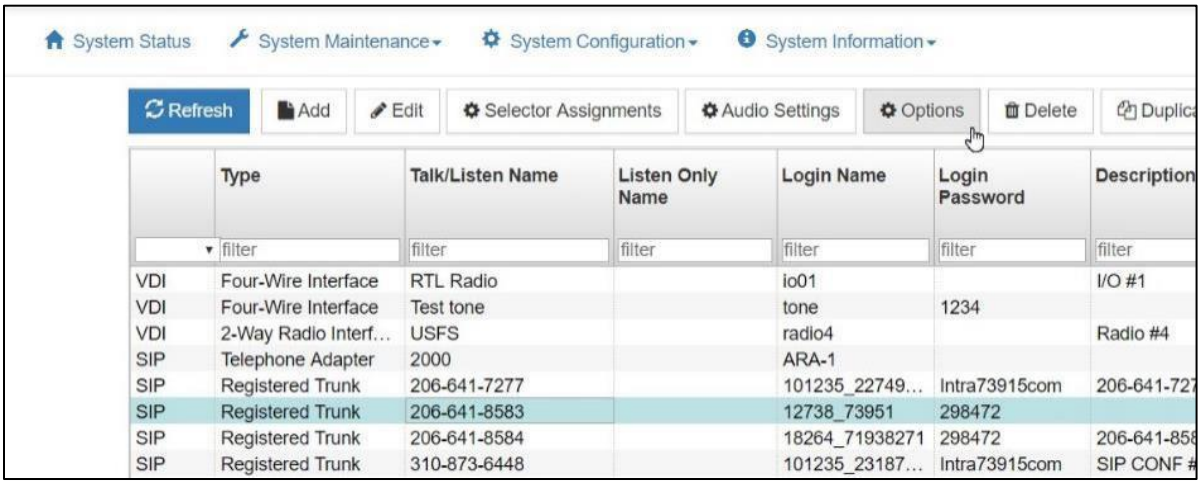
From the System Configuration tab click Client Configuration.



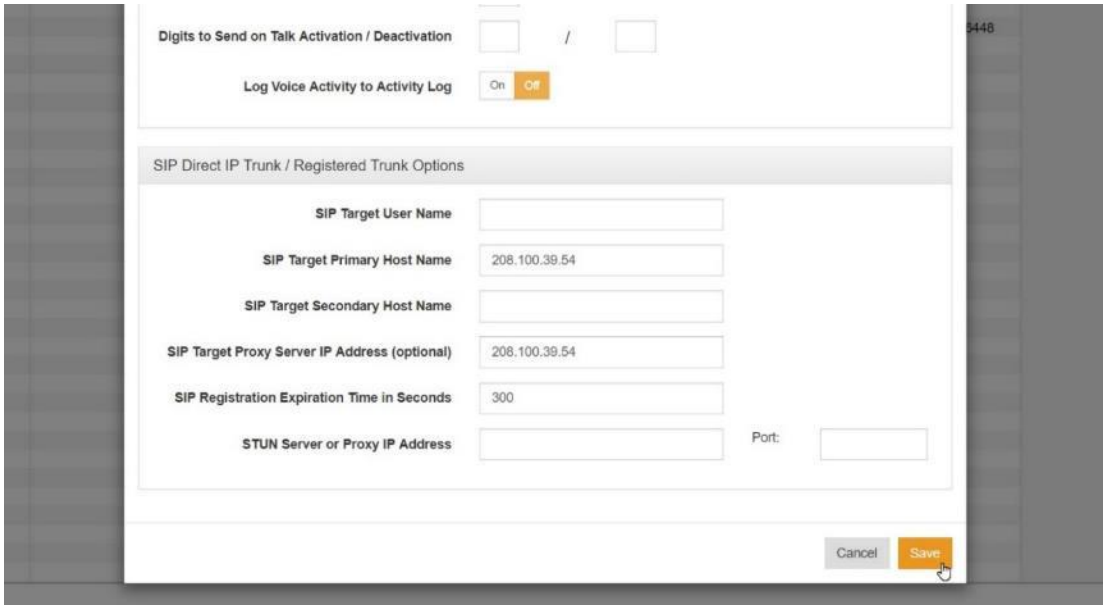
Click Add.



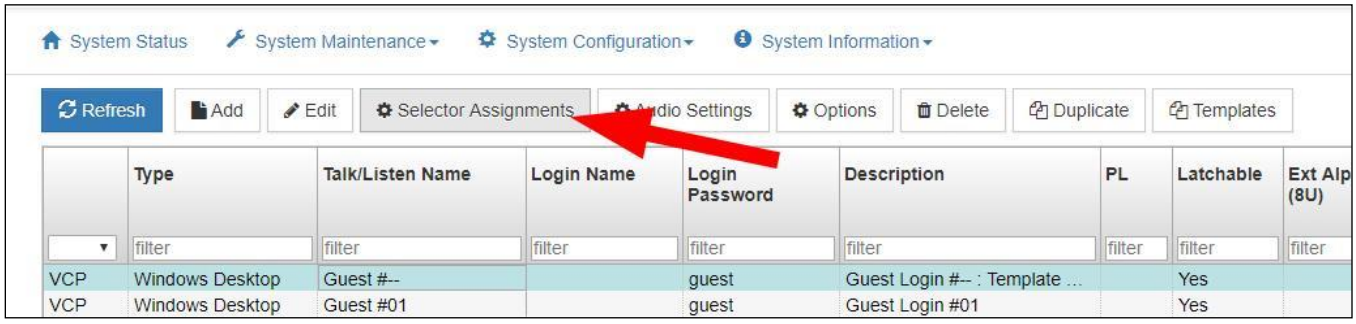
On the Client Type drop-down menu select “SIP Device: Intersystem/PBX Registered Trunk”. For Login Name enter in the username given by your SIP provider. The Login Password will be the password you set up with the SIP provider. The Selector Talk/Listen Name is the phone number. In the options panel below enable “Always Show Selector when Off-line”. Click save at the bottom of the menu.

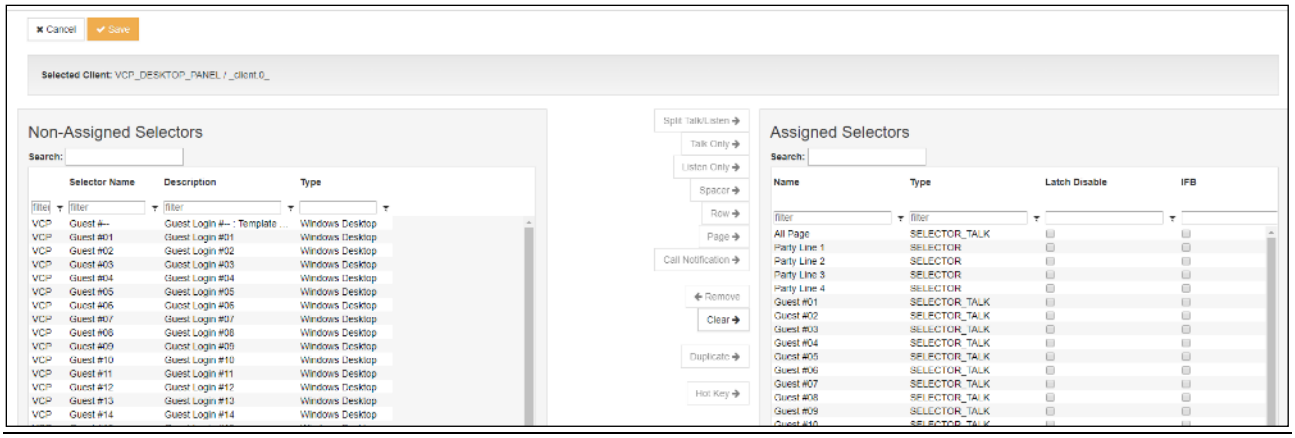


Click on the client you just made to highlight it then click Options.



Scroll down to the bottom of the menu. For the SIP Target Primary Host Name enter in the PBX IP given by your SIP provider. You can also enter in the SIP Target Proxy Server IP Address, although this step is optional.





VLINK Selector Assignments

This page allows assignment of a Control Panel's selectors. The selectors can be the label associated with any other Control Panel, Device Interface, Party Line, or Fixed Group.

- Highlight a selector in the left-hand portion of the screen and select 'Split Talk/Listen' to add the desired selector with a combination Talk/Listen path
- Highlight a selector in the left-hand portion of the screen and select 'Talk Only' to add the desired selector with a Talk Only path
- Highlight a selector in the left-hand portion of the screen and select 'Listen Only' to add the desired selector with a Listen only path
- Highlight a selector in the right-hand portion of the screen and select 'Remove' to delete an assigned selector
- Select 'Clear' to delete all assigned selectors
- Select 'Defaults' to restore selected client to the default for all clients
- Latchable – Configures the selected Selector to be latch disabled
- IFB – Not yet implemented
- ISO – Not yet implemented
- Speaker Dim – Configures the selected Selector to dim speaker to prevent acoustical feedback
- Hot Key – Assigns a keyboard key to the selected Selector so that the key can be used to activate and deactivate the Selector on a VLINK Control Panel

- ‘Selectors to Display per Row’ dictates the number of selectors to be displayed per row on a VLINK Control Panel

- ‘Default Assignments to Non-Latchable’ assigns new selectors as Latch Disabled

- ‘Selector Activation Method’ is an option available only for non-Control Panels and indicates under what circumstances the assigned selectors are to activate. The possible options are as follows:
    - ‘On This Client Connect’ – This default option results in the selectors be activated as long as the client is connected

    - ‘On Other Client Disconnect’ – This option results in the selectors being activated only when a specific client as designated by the Selector Activation Detail is disconnected allowing for a redundant audio input of a critical feed

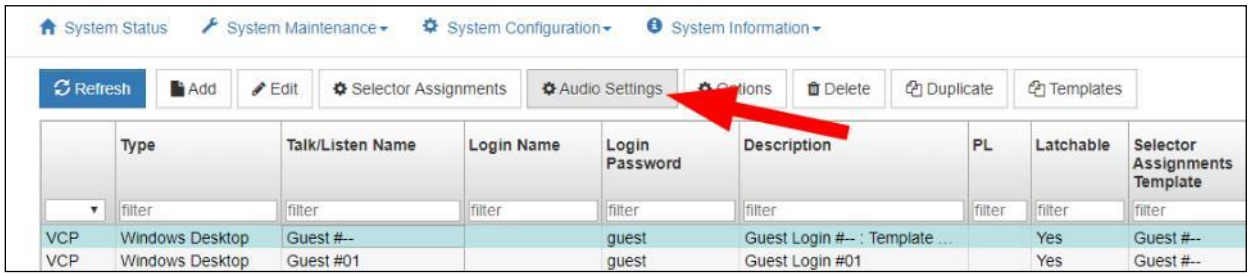
    - ‘On Voice Activity Detection’ – This option results in the selectors being activated only when Voice Activity is detected allowing for a device like a two-way radio to provide In-Use indication when the channel is active

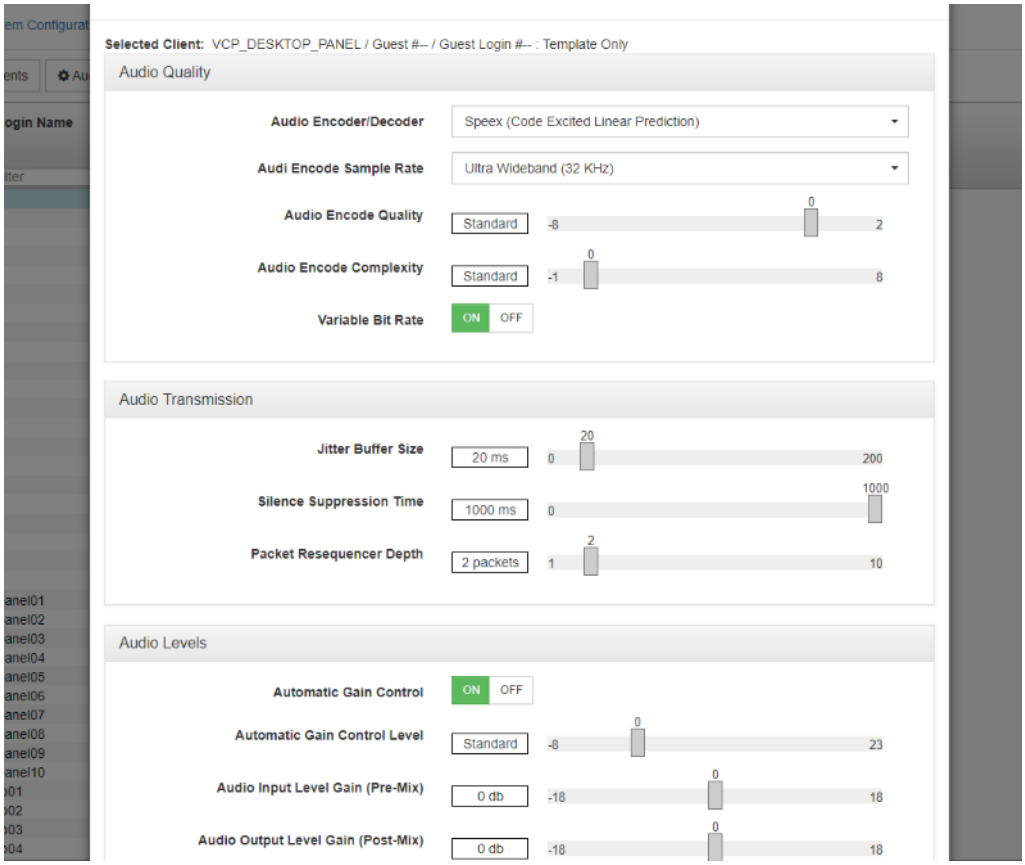
    - ‘On Logic Input Activation’ – This option results in the selectors being activated only when an external logic Input signal is detected via the Device Interface application, allowing for a device like a two-way radio to provide In-Use indication when the channel is active

    - ‘On DTMF Tone Detection’ – This option, available only for SIP clients, results in the selectors being activated only when the selector’s corresponding DTMF code is detected. The corresponding DTMF codes are the order in which the selectors appear in the list from 1 to the

    - number of selectors. A DTMF code of zero turns off any previously activated selector

VLINK Audio Settings





Audio Quality **Audio Encoder/Decoder**: This setting allows selection of a different encoder/decoder. For VLINK Control Panels and VLINK Device Interface clients the choice is between a High Compression / Low Bitrate Codec used for Internet connectivity and a Low Compression / High Bitrate codec that may be used for Local Network connectivity to slightly reduce latency. For SIP Devices the codec specified is preferential codec used when negotiating which codec to use with the SIP Device.

Audio Encode Sample Rate: This setting controls the sampling rate supported by the Clients and thereby dictates default fidelity for the Client connections. This setting is typically the same as the System Audio Sampling rate however it can be specified at lower rate but never at a higher rate (refer to System Audio Sampling Rate for additional detail). Higher audio sampling rates have more significant requirements in

computational speed and network bandwidth. Careful consideration must be given when choosing this setting with respect to client hardware and the client network connection.

Audio Encode Quality: The VLINK System codec achieves compression at the expense of fidelity of the input speech signal. Unlike some other speech codecs, it is possible to control the tradeoff made between quality and bit-rate.

Audio Encode Complexity: With the VLINK System codec, it is possible to vary the complexity allowed for the encoder. This is done by controlling how the search is performed with an integer ranging from 1 to 10. For normal use, the noise level at complexity 1 is between 1 and 2 dB higher than at complexity 10, but the CPU requirements for complexity 10 is about 5 times higher than for complexity 1. In practice, the best trade-off is between complexity 2 and 4, though higher settings are often useful when encoding non-speech sounds.

Audio Transmission

Variable Bit Rate: Allows the system's codec to dynamically change the bit rate at which audio is being encoded. As sounds like vowels require a higher bit rate to achieve good quality as compared to "s" and "f" sounds, this setting efficiently achieves the best sound quality within the given confines. The system can be set for variable rate or fixed rate. While this setting improves the quality of speech, it conversely degrades the quality of music and should therefore be disabled when using a program feed.

Jitter Buffer Size: This setting specifies the depth of the jitter buffer in milliseconds. In network-based communications, the delivery time of audio packets across the network may not be uniform. This characteristic is known as jitter. As such, audio received from a network connection must be buffered to compensate for this so that a continuous, time-relative stream of audio can be delivered to the consumer of the audio. Different network topologies will have different jitter characteristics. For example, a public Internet connection will have significantly more jitter than an internal local network. The effect of a jitter buffer that is too small will result in audio gaps. The value is specified in milliseconds.

Silence Suppression Time: Ceases all transmission of audio data when no voice activity is detected from a Control Panel or Device Interface after the specified time lapse. This virtually eliminates background noise during multiparty conferences. However, it may initially be disconcerting to some individuals as the 'comfort noise' typically associated with analog systems is suppressed. Additionally, this feature minimizes the overall required network bandwidth. The value is specified in milliseconds in the range of 100-1000 ms or can be turned off entirely.

Packet Re-sequencer Depth: This setting specifies the number of packets that are stored when waiting for an out of sequence audio packet. In some network topologies, even if UDP packets are sent in sequential order, they are received non-sequentially. These packets must be re-sequenced before use. After the maximum re-sequencer

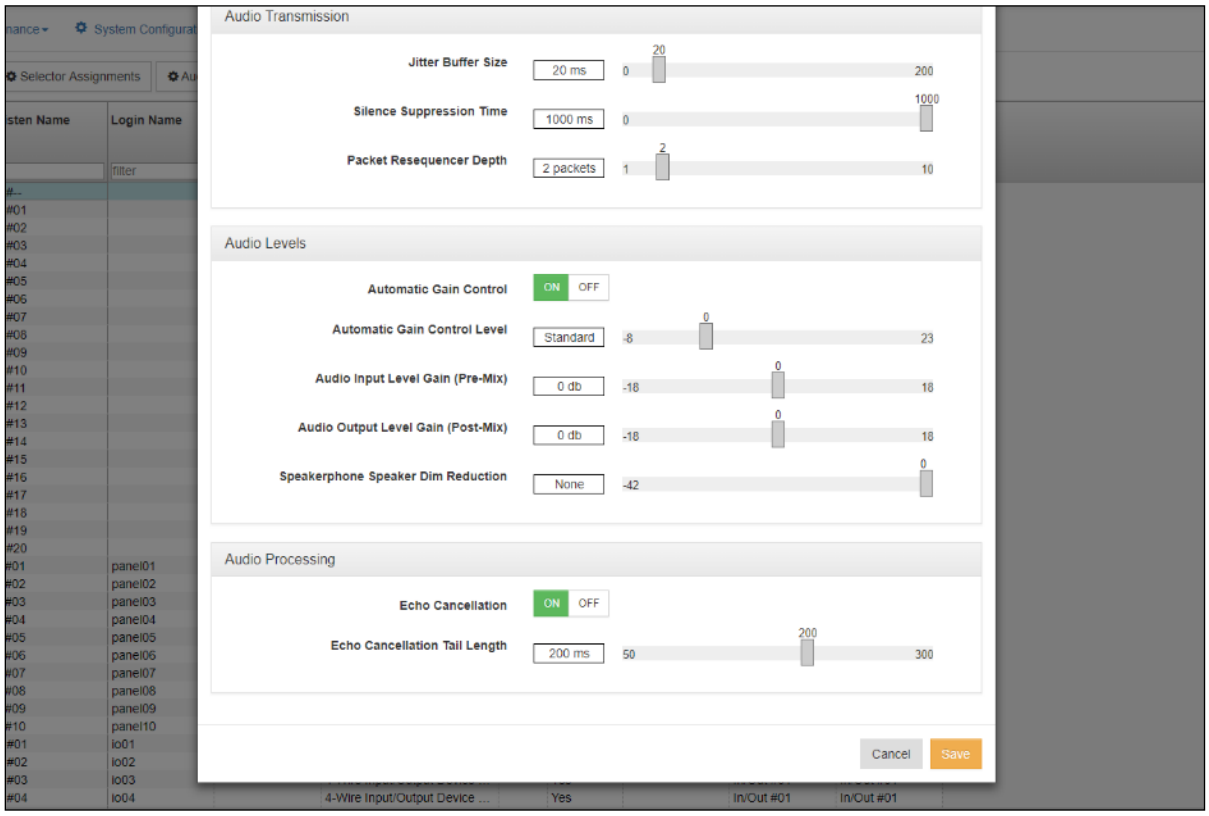
depth has been reached, the packet being awaited is declared to be lost and the re-sequencing is re-started at the next earliest received packet. Valid settings are from 2 to 10 packets.

Audio Processing

Automatic Gain Control (AGC): This setting enables or disables AGC on the audio path from Client to the Server. AGC automatically increases or decreases the audio level such that the client presents a uniform audio level to the Virtual Matrix. AGC is primarily appropriate for use with a Control Panel when used with a headset microphone. In some situations where there is a high amount of background noise or some return audio leakage the AGC may incorrectly amplify the noise to normal audio levels.

Automatic Gain Control (AGC) Level: This setting increases or decreases the sensitivity of the AGC. Increasing or decreasing the sensitivity of the ACG changes the behavior of the AGC such that it adapts faster or slower, respectively, to audio levels not considered to be at uniform level. Decreasing the sensitivity may be useful in cases where there is a high amount of background noise or some return audio leakage.

Audio Input Level Gain (Pre-Mix): This setting controls the audio input level sent from the Client to the Virtual Matrix. This setting is typically used only when the client's audio input device does not provide a sufficiently audible level (as heard by all other clients) and does not have a local gain control to compensate. The value can be adjusted a maximum of +/-18dB in 6dB steps.



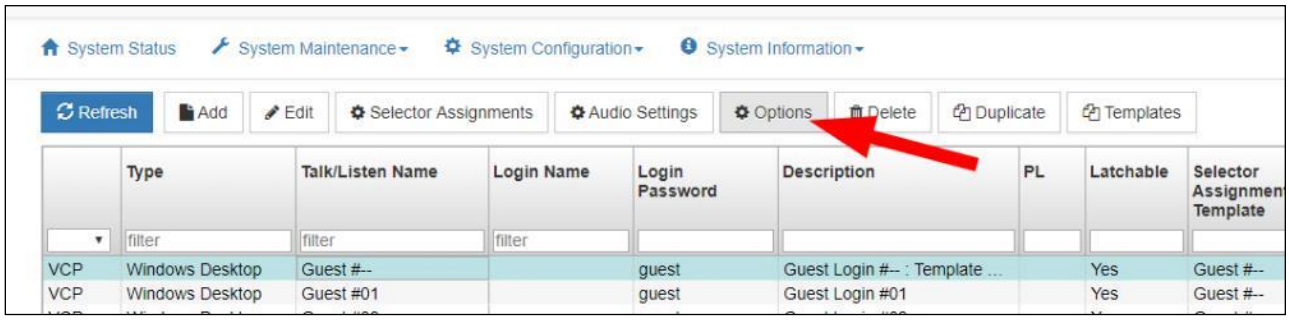
Audio Output Level Gain (Post-Mix): This setting controls the audio output level sent to the Client from the Virtual Matrix. This setting is typically used only when the client's audio output device does not provide a sufficiently audible level and does not have a local gain control to compensate. The value can be adjusted a maximum of +/-18dB in 6dB steps.

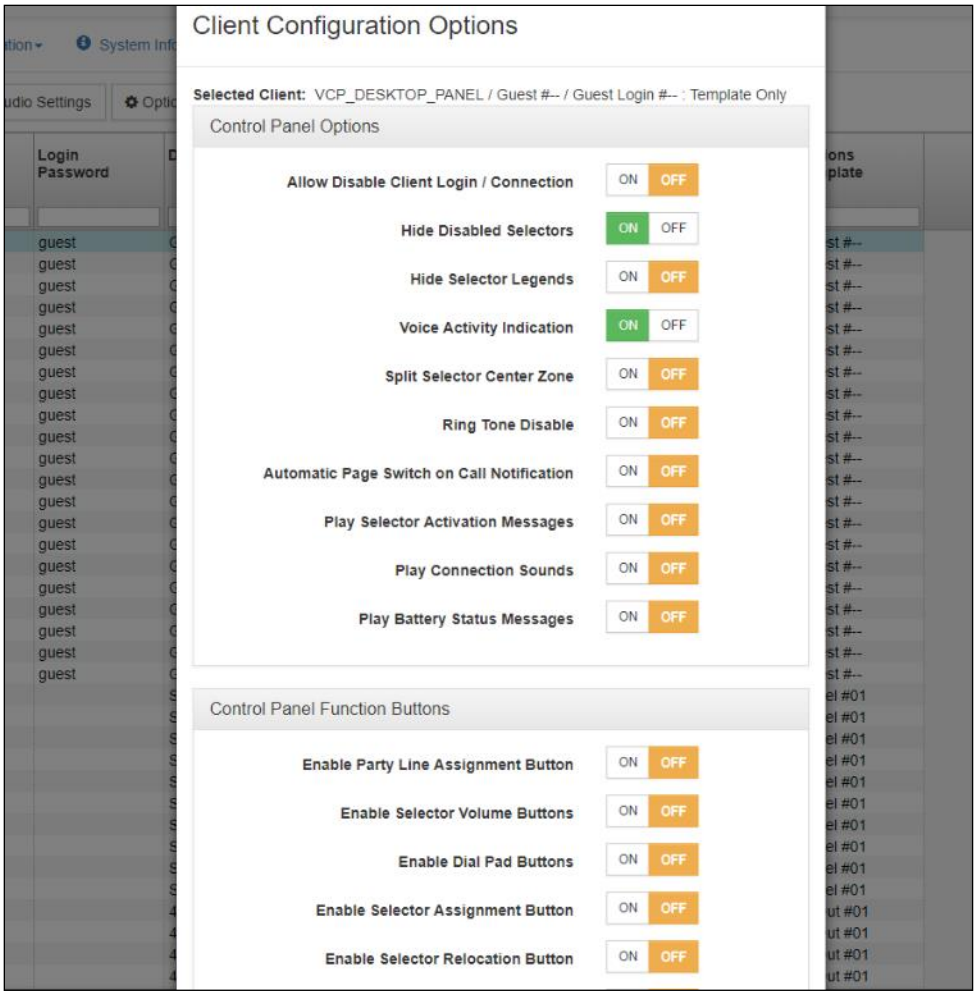
Speakerphone Speaker Dim Reduction: This setting dims the incoming audio output by a configured amount when you activate the talk key.

Echo Cancellation: This setting enables or disables the client's Echo Cancellation. Echo Cancellation is useful if there is any return audio leakage from the client's speaker back to his microphone. This may result in an audible echo heard by any other client that is talking and listening to the client with the return audio leakage.

Echo Cancellation Tail Length: This setting controls the duration the echo canceller waits to receive the echo before it begins the cancellation process. The recommended tail length is approximately a third of the room reverberation time. For example, in a small room, reverberation time is in the order of 300ms, so a tail length of 100ms is recommended.
Audio Encryption: This setting is a licensable option for security measures. Clients may choose from following options: No Encryption, AES 128, or AES 256.

VLINK Client Configurations Options





Control Panel Options

Allow Disable Client Login/Connection: This setting allows the client's login capability to be disabled.

Hide Disabled Selectors: This setting hides selectors assigned to other clients that are not logged into the system. When the clients come online, their selector will dynamically appear.
Hide Selector Legends: This setting hides the overlaid selector legends displayed on listen selectors ('L') and talk selectors ('T').

Voice Activity Indication: This setting is used to visually indicate voice activity on Control Panel selectors, represented by selector text and background color switching between base state (yellow text / navy background) and default activity indication colors (white text / light navy background) or selected activity indication colors (variable). Voice Activity Indication is only available if the Control Panel has the ability to listen to or is being talked to by the client indicating voice activity.

Split Selector Center Zone: This setting allows the center of the selector to be used to activate both the talk and listen selectors for split talk/listen selectors.

Ring Tone Disable: Disables Ring Tone loaded in SIP Client Options for the selected user.

Automatic Page Switch on Call Notification: This setting will cause the pages (when multiple pages are used) to switch to the page with the current incoming call from another user.

Play Selector Activation Messages: This setting will cause the unit to announce “Talk on/ Talk Off” or “Listen On/Off” when activating the selectors.

Play Connection Sounds: This setting will cause the unit to play a sound every time you connect or disconnect to the network.

Play Battery Status Messages: This setting will announce the status of the battery depending on the percentage of battery left – (5, 10, 15, 20, 25, 30)

Control Panel Function Buttons

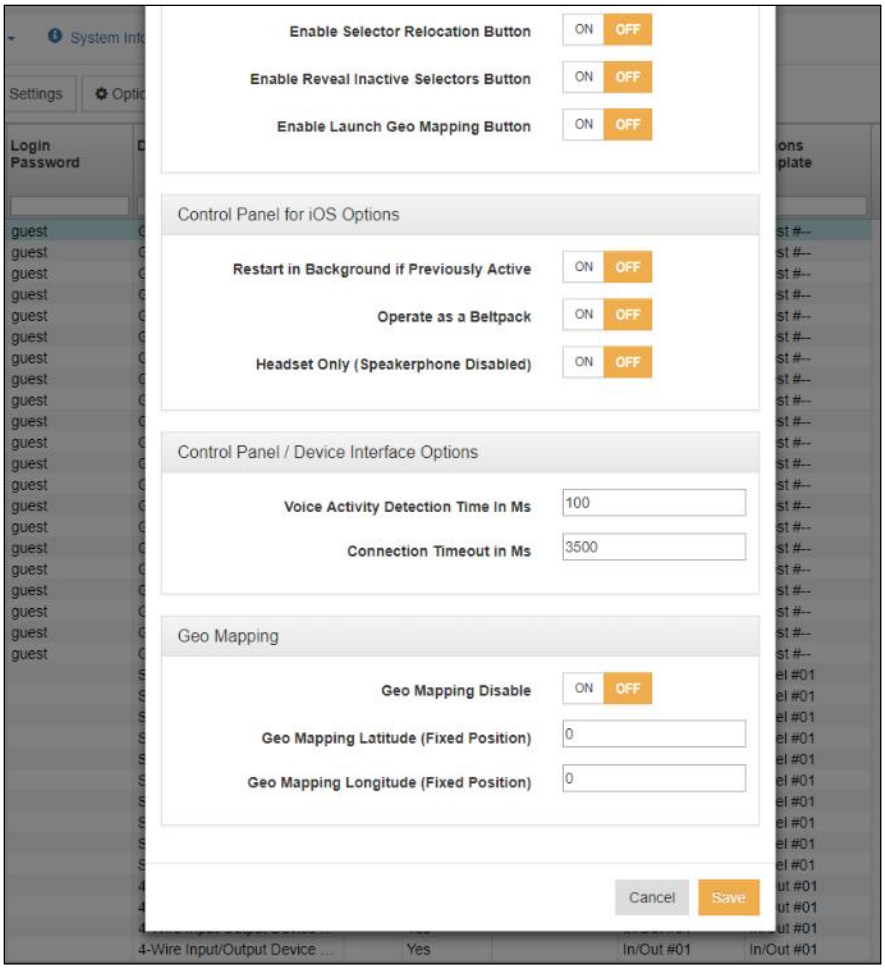
Enable Party Line Assignment Button: This will enable the PL Assignment button on the Control Panel allowing the user the capability of assigning other “devices” to PLs.

Enable Selector Volume Buttons: This enables the “Selector Volume” button that gives the user a quick key to adjust individual selector volumes.

Enable Dial Pad Button: This enables a button that enables the Dial Pad when a SIP device is being accessed to dial out.

Enable Selector Assignment Button: This enables a button that gives the user the capability of assigning selectors through the VCP GUI via a pop-up window.

Enable Selector Relocation Button: This enables a button that allows the user to rearrange the order of their selectors to customize to their particular needs.



Enable Reveal Inactive Selectors Button: This allows the user to disable/enable inactive (not logged in) selectors.

Enable Launch Geo Mapping Button: This enables a button that will launch a web page directed to the Geo Mapping Server to display the Geo Mapping web application.

Control Panel for iOS Options

Restart in Background if Previously Active: This will restart the VCP in the background when the device is restarted.

Operate as a Beltpack: This will use the "Beltpack Template" that allows for oversized buttons on an iOS device such as the iPhone/iPod Touch which can also be used with Screen covers with cutouts for these oversized buttons.

Disable Integrated Mic and Speaker: This will disable the use of an internal mic and speaker on the iOS device.

Control Panel/Device Interface Options

Voice Activity Detection Time in Ms: This allows the amount of time to be set for Voice Activity Detection to turn on for the selected user.

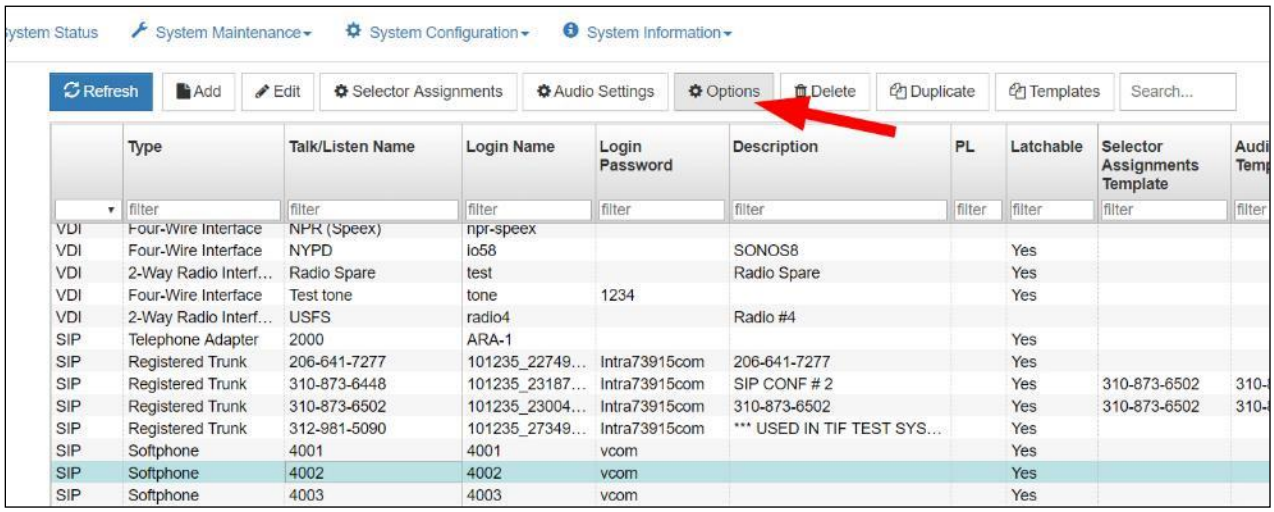
Geo Mapping

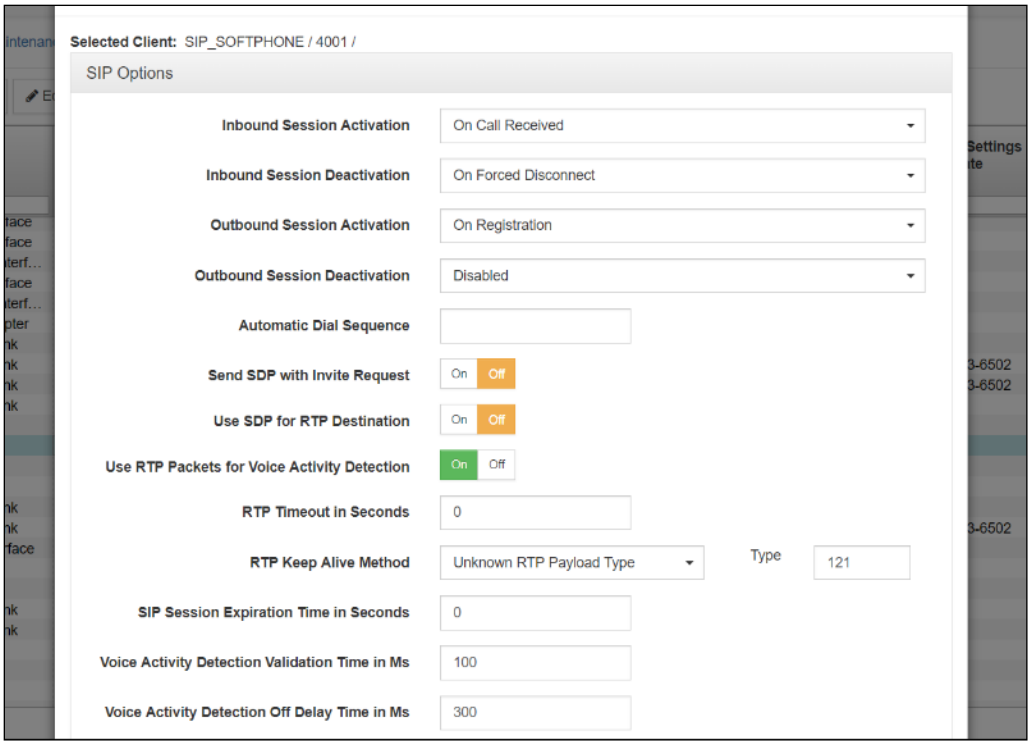
Geo Mapping Disable: This will disable the selected user's access and data sent to the Geo Mapping server.

Geo Mapping Latitude/Longitude: This allows the GPS coordinates of the selected user to be set when used as a fixed position. This is not used for mobile devices.

VLINK SIP Options

See "Appendix A" for an explanation of SIP and application examples.





SIP Options

- **Inbound Session Activation**: This setting specifies how the Virtual Matrix handles the activation of a call initiated by the SIP client. If configured for 'Disabled,' the call initiated by the SIP client is ignored. If configured for 'On Call Received (Auto Answer),' the call initiated by the SIP client will be automatically answered by the VLINK Virtual Matrix. If configured for 'On Talk Selector Activation,' the call initiated by the SIP Client will be indicated on the associated Control Panel selectors and the call will be answer only if a VLINK Control Panel activates the talk selector associated with the SIP Client

- **Inbound Session Deactivation**: This setting specifies how the Virtual Matrix handles the deactivation of a call initiated by the SIP client. If configured for 'Disabled,' the call initiated by the SIP client can never be disconnected by the Virtual Matrix. If configured for 'On Forced Disconnect,' the call initiated by the SIP client can only be disconnected by a VLINK Control Panel using the 'Disable Client Login' feature. If configured for 'On Talk Selector Deactivation,' the call initiated by the SIP Client will be disconnected when all VLINK Control Panels deactivate the talk selectors associated with the SIP Client

- **Outbound Session Activation**: This setting specifies how the Virtual Matrix handles the activation of a call initiated to the SIP client. If configured for 'Disabled,' the Virtual Matrix cannot initiate any call to the SIP client. If configured for 'On Registration,' the Virtual Matrix will initiate a call to the SIP client as soon as the SIP client makes its presence known through a process known as 'Registration.' If configured for 'On Talk Selector Activation,' the Virtual Matrix will

initiate a call to the SIP client when any VLINK Control Panel activates the talk selector associated with the SIP Client

- **Outbound Session Deactivation**: This setting specifies how the Virtual Matrix handles the deactivation of a call initiated to the SIP client. If configured for 'Disabled,' the call initiated to the SIP client can never be disconnected by the Virtual Matrix. If configured for 'On Forced Disconnect,' the call initiated to the SIP client can only be disconnected by a VLINK Control Panel using the 'Disable Client Login' feature. If configured for 'On Talk Selector Deactivation,' the call initiated to the SIP Client will be disconnected when all VLINK Control Panels deactivate the talk selectors associated with the SIP Client

- **Automatic Dial Sequence**: This setting specifies a dial sequence to be dialed as soon a call is established with a SIP Client. To insert a delay in the dial sequence, use 'P' to insert a 5 second delay and 'p' to insert a 1 second delay

- **Send SDP With Invite Request**: This setting changes the default behavior of calls initiated by the Virtual Matrix to allow compatibility with devices that do not conform to proper SIP implementation, specifically the Raytheon ARA-1. Normally when the Virtual Matrix initiates a call to the SIP client, it does so without sending a Session Description Protocol (SDP), so that it can subsequently control the codec selection

- **Use SDP for RTP Destination**: This setting changes the default behavior of the Virtual Matrix to allow strict conformance with the Real-Time Protocol specification. Normally the Virtual Matrix ignores the RTP IP address specified in the SDP and uses the actually received RTP IP address as the SIP specification as written and does not account for SIP clients behind NAT firewalls that typically alter the IP address of the packet

- **Use RTP Packets for Voice Activity Detection**: Many SIP device employ Silence Suppression to eliminate unnecessary network traffic. When a device supports silence suppression, this option can be enabled such that Voice Activity will be indicated whenever RTP audio packets are received. If this option is not enabled, The Virtual Matrix will indicate Voice Activity by analyzing the content of the audio stream

- **RTP Timeout In Seconds**: During an active SIP call, there should be a constant and ongoing flow of RTP audio packets or RTP Keep Alive packets between the client and the server in additional to the SIP control packets. In some cases the RTP audio packets might be blocked even though the SIP control packets are not. This option specifies the time in seconds to wait for RTP audio packets before assuming there must be a problem with the connection and disconnecting the call. Setting this value to zero disables the check for an RTP Timeout

- **RTP Keep Alive Method**: When a SIP device supports silence suppression, it is often necessary to send RTP keep alive packets when audio is not being sent in order to ensure that no firewalls or routers in between close the associated ports due to inactivity. This option specifies the type of RTP Keep Alive Method as not all SIP devices can accept the RTP Keep Alive Methods specified by RFC 6223

- **SIP Session Expiration Time In Seconds**: During an active SIP call there is often a constant and ongoing flow of SIP control packets between the client and the server. In some cases the SIP control packets might stop due to a loss of network connectivity. This option specifies the time in seconds to wait for the SIP control packets before assuming there must be a problem with the connection and disconnecting the call. Setting this value to zero disables the check for a SIP Session Expiration
- **Voice Activity Detection Validation Time in Ms**: This allows the amount of time to be set for Validation Time to register as a Voice Activity event. For example, there needs to be 100ms of audio for Voice Activity Detection to turn on to pass audio in the picture above

- **Voice Activity Detection Off Delay Time in Ms**: This allows the amount of time to be set for Voice Activity Detection to be turned off. This is the amount of silence allowed after a user is speaking

- **Call Notification Ring Tone**: A file path can load an audio file to be played when an incoming call is placed

- **Auto-Answer Notification Message**: This option allows the selection of an audio wave file to be played as the notification message to the caller to indicate that the connection has been established between the SIP client and the VLINK Virtual Matrix. User provided files can be uploaded to the server provided the files are in a 16 bit mono audio format, preferably set to the same Audio Mix Sample Rate of the Virtual Matrix

- **Auto-Answer Delay Time In Ms**: This option sets the delay time in milliseconds after which the VLINK Virtual Matrix will automatically answer a received SIP call. Typically this value will be set to zero such that the call is answered immediately. In some situations it is desirable to delay answering the call so that VLINK Control Panels listening to that line will hear a ring signal

- **Auto-Answer Access Code**: This option enables the requirement that an Access Code must be entered by the caller if the VLINK Virtual Matrix automatically answers a received SIP call. The Access Code may consist of up to 10 digits. When configuring an access code, it is recommended to configure an appropriate Auto-Answer Notification Message which prompts the caller to enter the access code. The provided "AccessCode.wav" message is provided for this purpose

- **Use 16kHz for G.722 Timestamp**: Sets bandwidth to 16kHz when the G.722 CODEC is used

- **Internal SIP Client User Name Prefix**: Character placed before the Client's User Name when dialing, most commonly used is the asterisk

SIP Direct IP Trunk Registered Trunk Options

In most situations, the VLINK Virtual Matrix itself acts as the SIP Registrar and SIP Server for SIP Clients to connect to it. However, when connecting to other SIP Registrars and SIP Servers, the VLINK Virtual Matrix can itself act as a SIP Client when a client is configured as a SIP Direct IP Trunk or a SIP Registered Trunk using the following options:

- **SIP Target User Name**: This specifies the fixed user name to use as the Address of Record (AOR) when sending SIP registration and/or SIP invite requests. If using a VLINK Control Panel to dial out through a PBX, this field must be left blank, as the dialed phone number will be User Name

- **SIP Target Primary Host Name**: This specifies the target primary host name or IP address of the target SIP Registrar or SIP Server to use when sending SIP registration and SIP invite requests

- **SIP Target Secondary Host Name**: This specifies the target secondary host name or IP address of the target SIP Registrar or SIP Server to use when sending SIP registration and/or SIP invites. The secondary host name will only be used in the case where a secondary (failover) server is available and the primary server is not reachable. If no secondary (failover) server is available, this field must be left blank

- **SIP Target Proxy Server IP Address (optional)**: This specifies the IP address of the SIP Proxy, if a SIP Proxy is used

- **SIP Registration Expiration Time In Seconds**: This option specifies the time in seconds that the Virtual Matrix is expected to refresh the registration with the target SIP Registrar. If the registration is not refreshed with the specified time frame, the SIP session will be terminated

- **STUN Server of Proxy IP Address**: This option enables SIP client transversal through a NAT by specifying the IP address of a STUN Server which will inform the Virtual Matrix of the public IP address of the SIP Client so that it can be included when sending SIP registration and SIP invite requests

Geo Mapping

- **Geo Mapping Disable**: Check this box to disable the use of geo mapping on any given SIP periphery

- **Geo Mapping Latitude (Fixed Position)**: Input the fixed latitude position for a geographically static SIP periphery

- **Geo Mapping Longitude (Fixed Position)**: Input the fixed longitude position for a geographically static SIP periphery

Templates

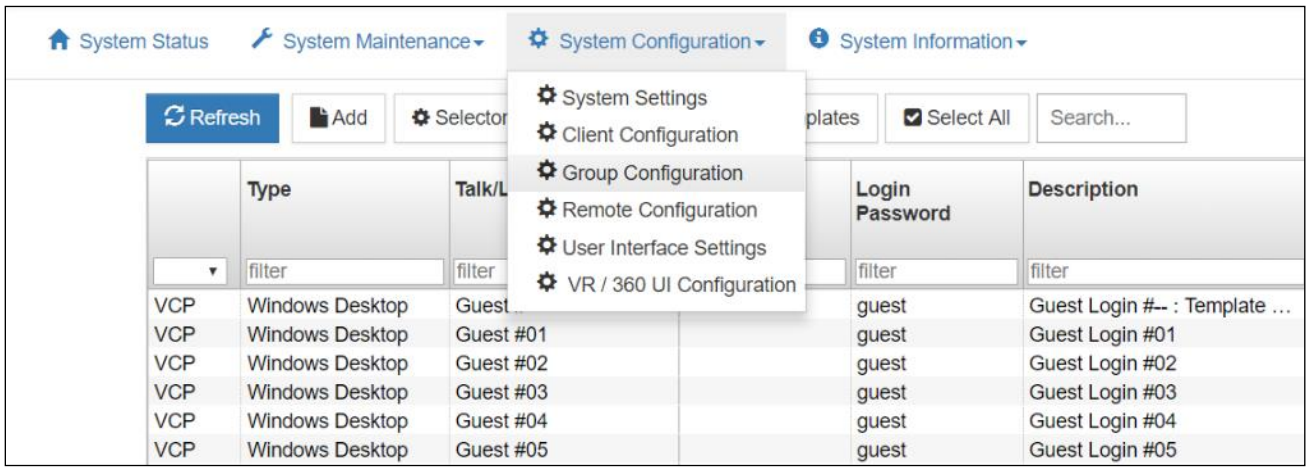
In the lower right hand corner of the main Client Configuration window, 'Templates' allow you to apply a given client's selector assignments, audio settings, and/or options to one or more other clients.

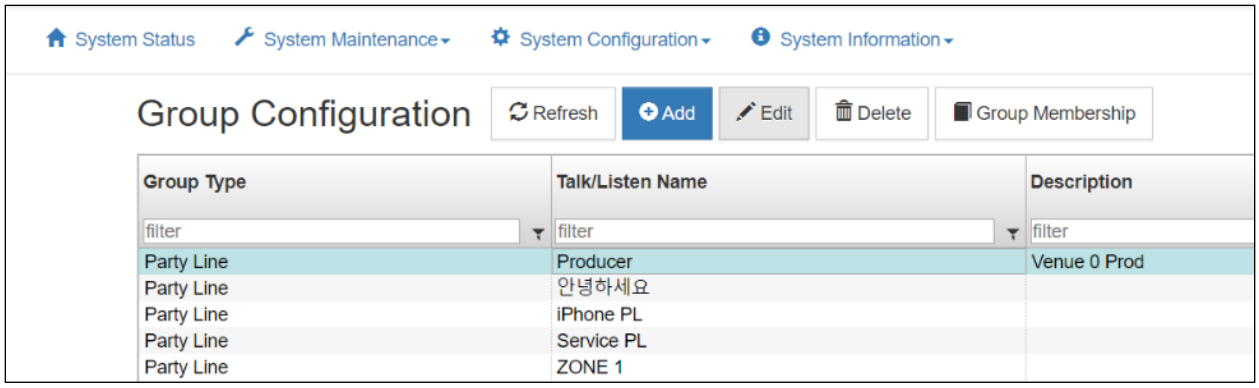
First select the client you want to use as the template and then using the **CTRL key**, select one or more other clients you want to apply your template's settings to. You will see the 'Templates' section become enabled. Click on the appropriate 'Link' button to populate selector assignments, audio settings, and/or options from your template to your other selected clients. If you later want to remove applied client settings from your template, use the appropriate 'Unlink' button.

VLINK Group Configuration

The 'Group Configuration' tab found in the 'System Configuration' area is used to add/edit/delete Party Lines and Fixed Groups, change selector names, and change group membership.

The main 'Group Configuration' window displays configured Party Lines and Fixed Groups.

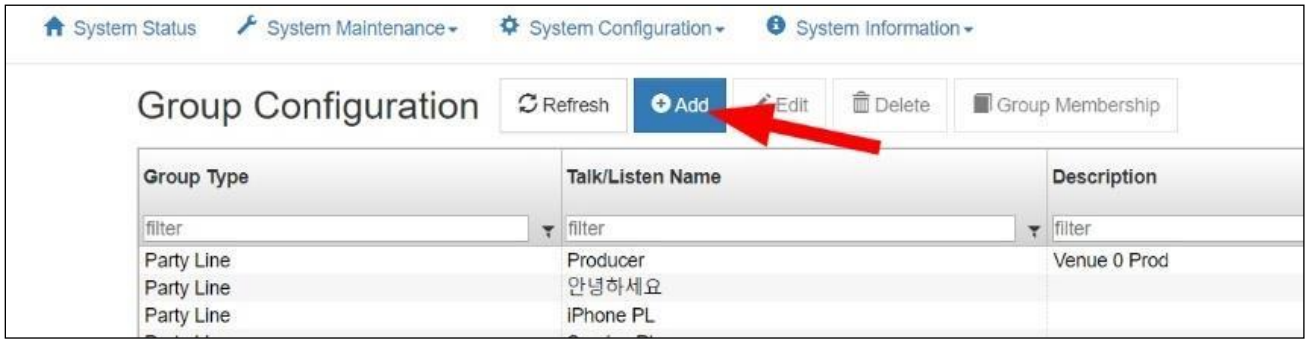




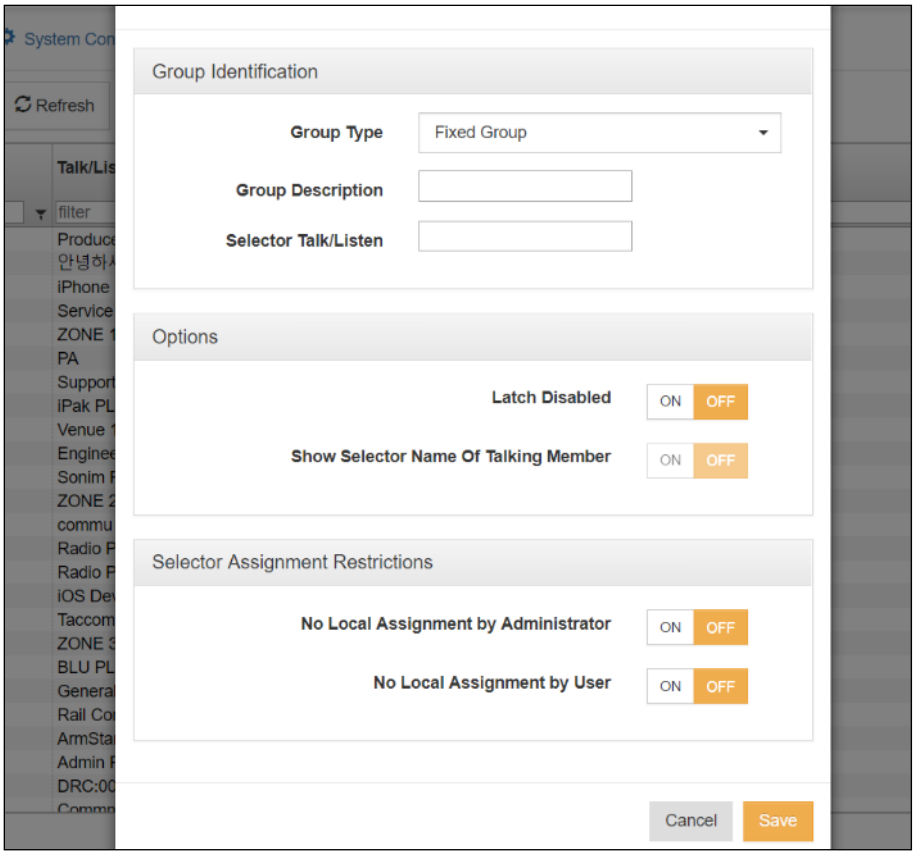
To edit a Party Line or Fixed Group, highlight the one desired and click 'Edit' to bring up the 'Group Configuration Add/Edit' window.

To delete a Party Line or Fixed Group, highlight the one desired and click 'Delete.'

To add members to a Fixed Group, click 'Selected Group' in the lower portion of the 'Group Configuration' window. Highlight a non-group member from the list of users and devices in the column on the left and click 'Insert' to add as a group member. To remove a group member highlight the user or device from the column on the right and click 'Remove.' Click 'Clear' to clear all group members.



To add a Party Line or Fixed Group, click the 'Add' button.



Type: Choose from the drop down box 'Party Line' or 'Fixed Group'

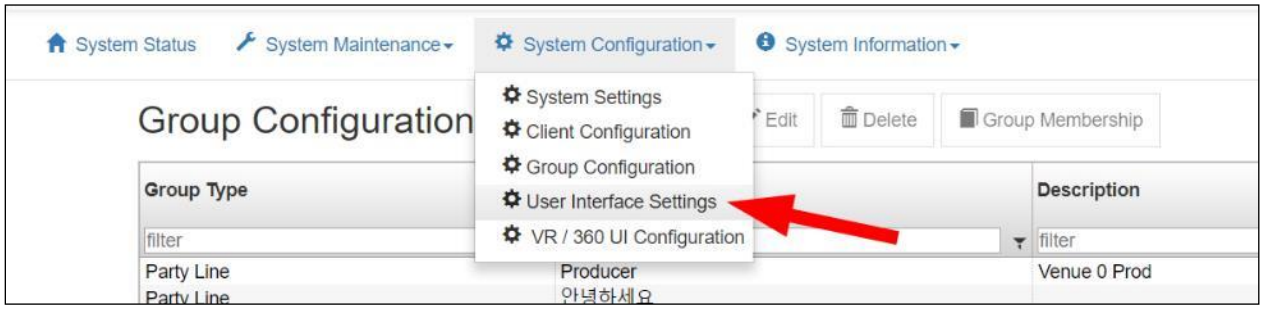
Description: You may add a short description for the Party Line or Fixed Group or leave it blank

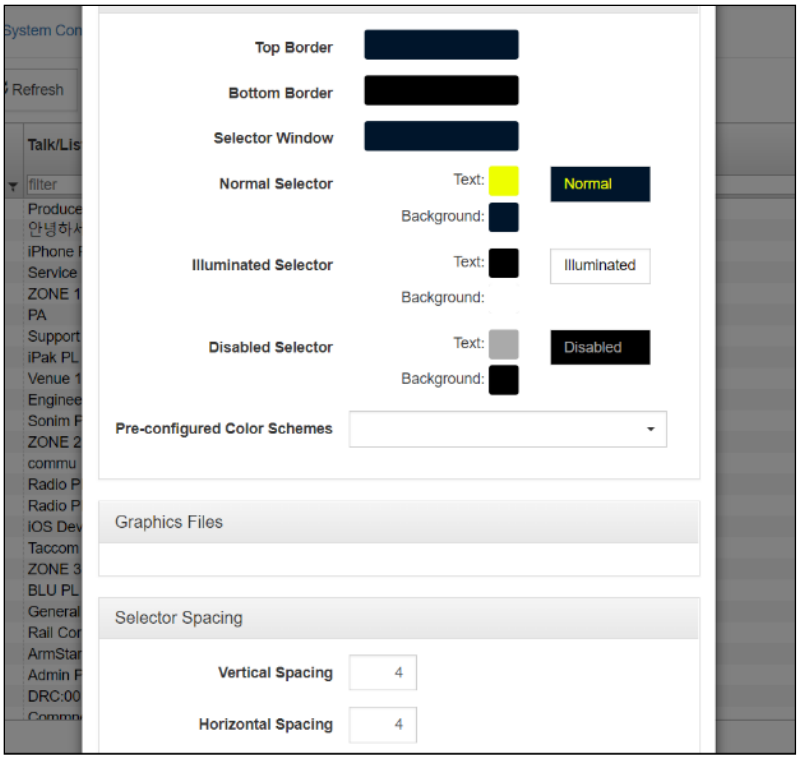
Selector Talk Label: Add a name of up to 10 alphanumerical characters which will appear on the selector

Latch Disable: Select latch disable to have the selector work as a momentary key

VLINK User Interface Settings

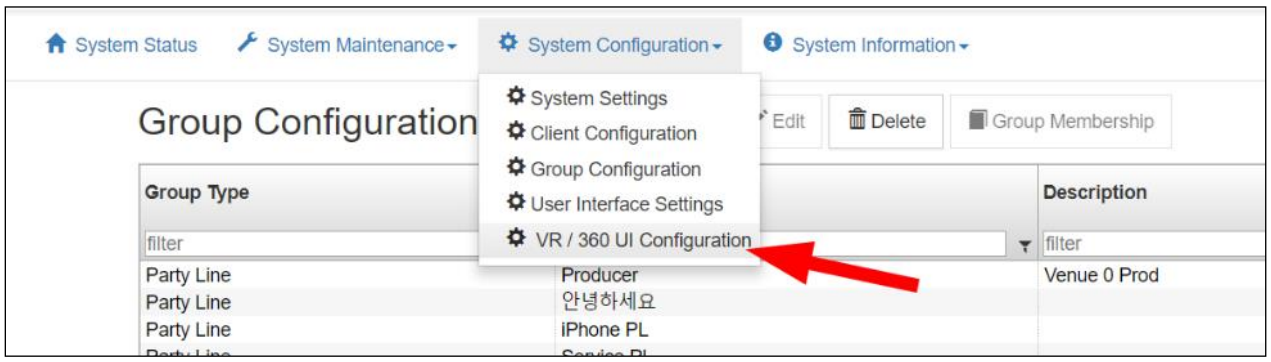
This allows customizations of the VCP GUI such as the Colors (Background and Text), Selector Spacing, and 24-bit PNG Graphic files that can be uploaded, like a logo.



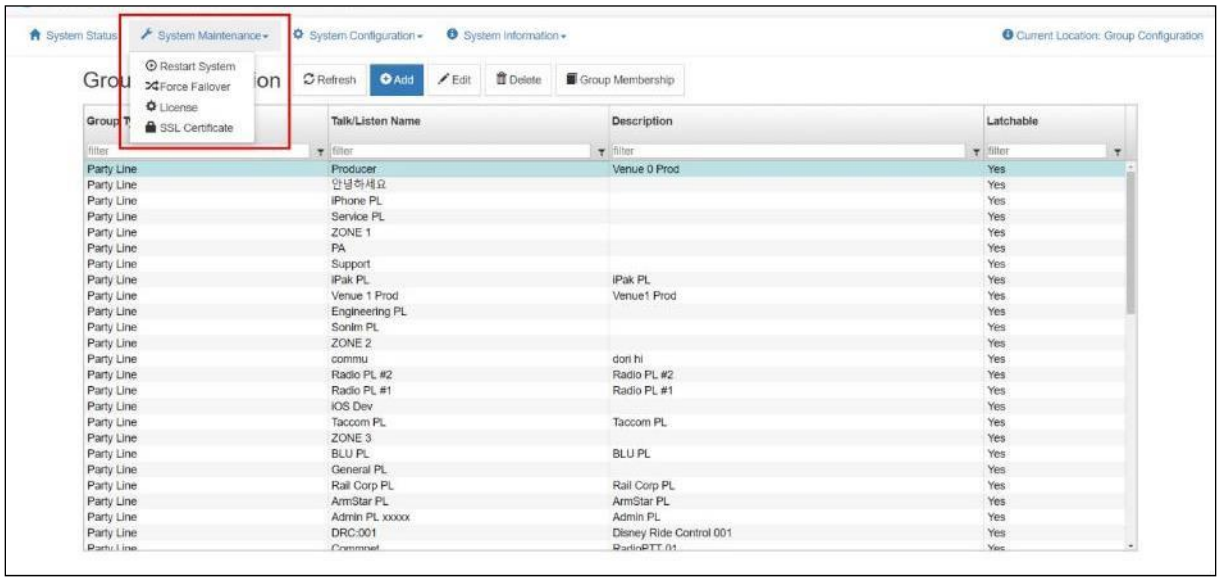


VR / 360 UI Configuration

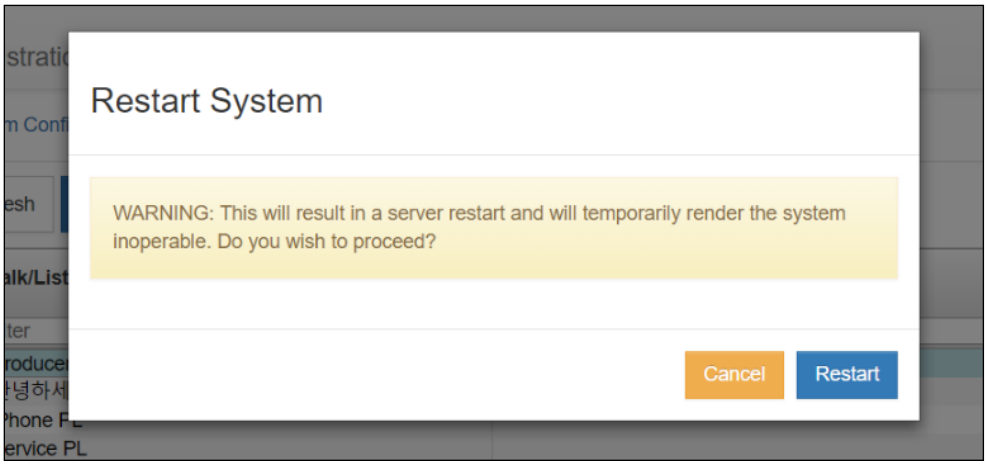
This allows you to name, add, and remove 360 video streams. To set a default stream click the checkbox next to the desired stream.



SYSTEM MAINTENANCE



Restart System: This feature restarts the system. A warning message will first appear noting that selecting this feature will temporarily render the system inoperable and asking if you wish to proceed. This feature is only available to the system administrator logged in with the master username and password

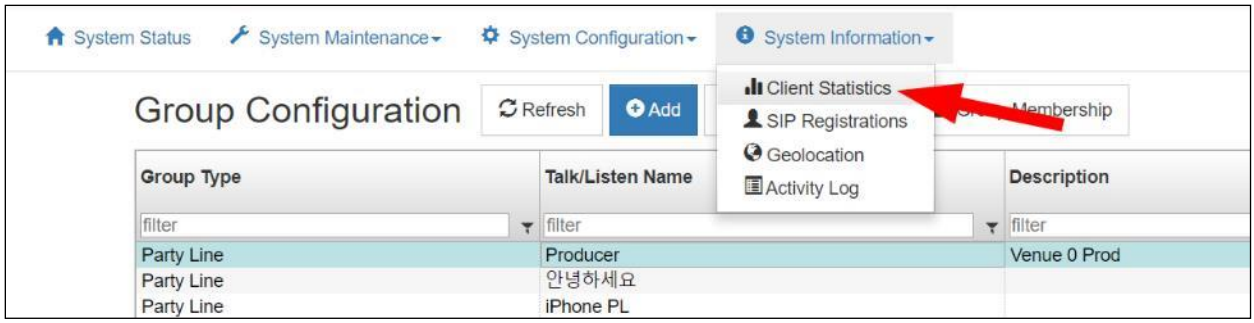


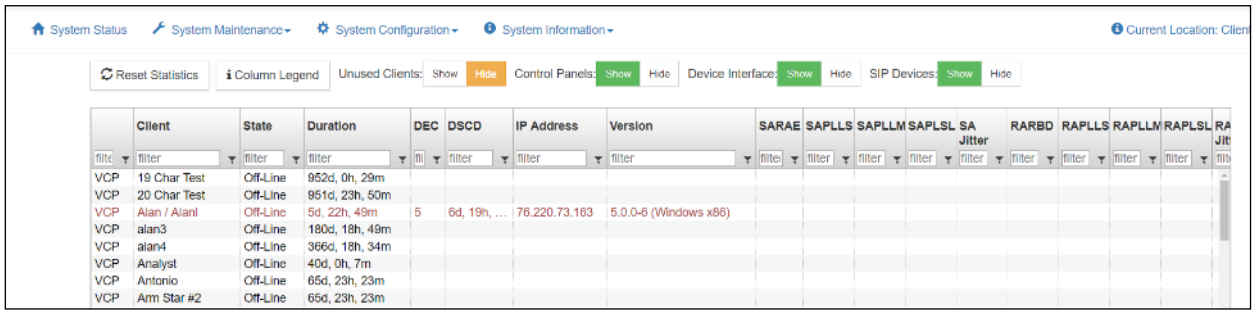
Force Failover: Forces the Primary to relinquish control to Secondary Server

Force Failback: Forces the Secondary to relinquish control to Primary Server

SYSTEM STATISTICS

Client Statistics: Displays individual send and receive audio and packet loss statistics for all client connections

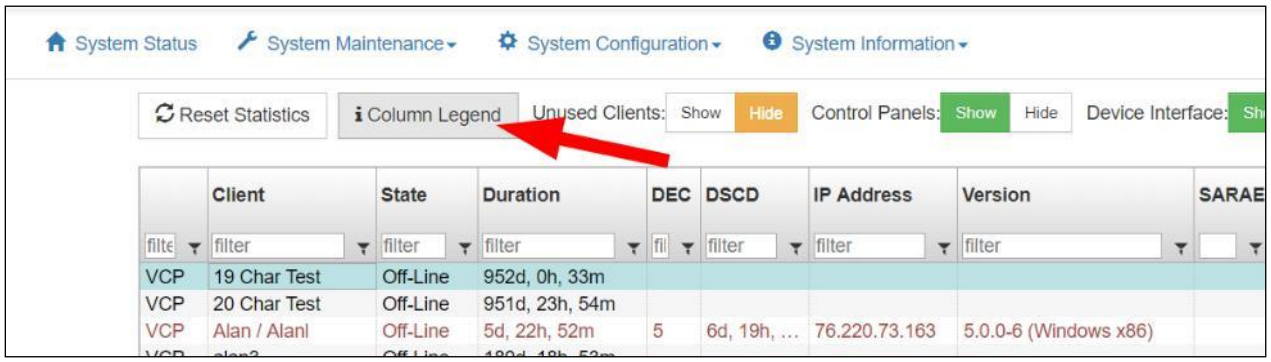


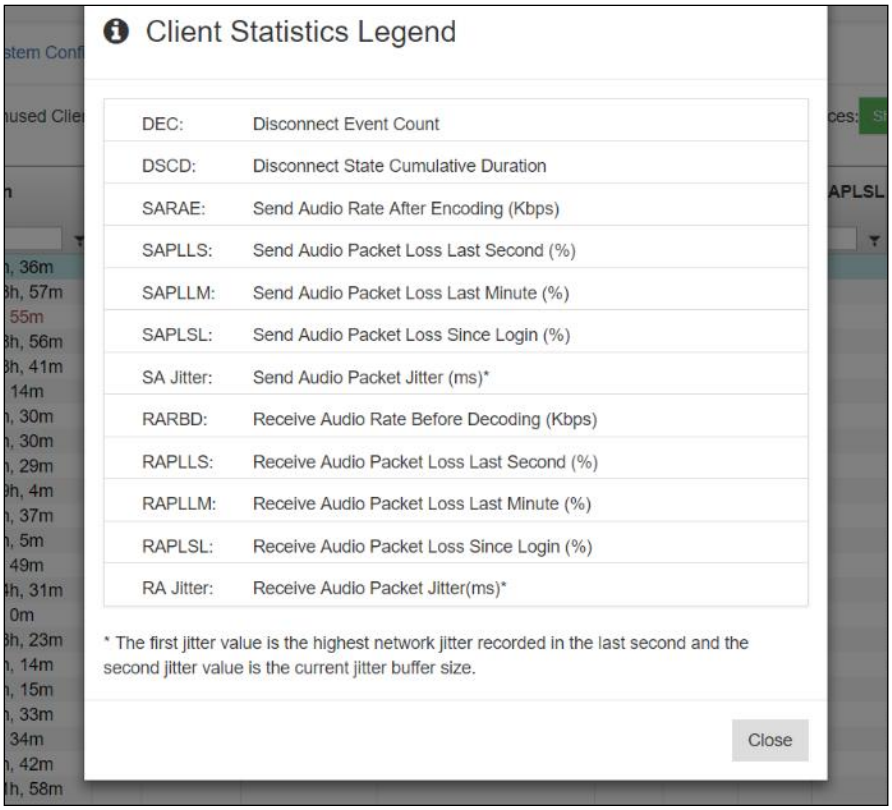


Click the ‘Reset Statistics’ button in the upper, left-hand corner to reset all client statistics or highlight a user or device and then click ‘Reset Statistics’ to reset the statistics of an individual user or device.

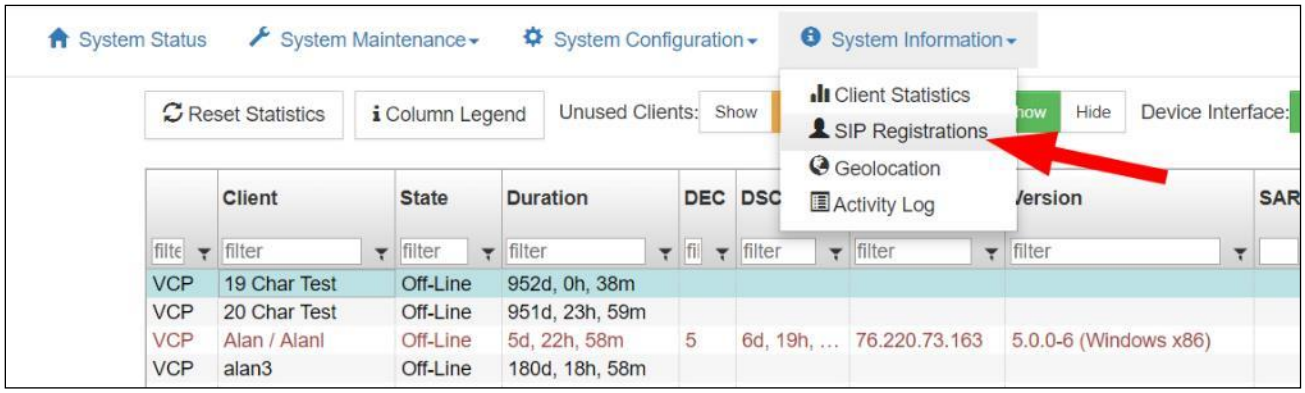
Select ‘Show Unused Clients’ to display users and devices programmed for your system but that are off-line.

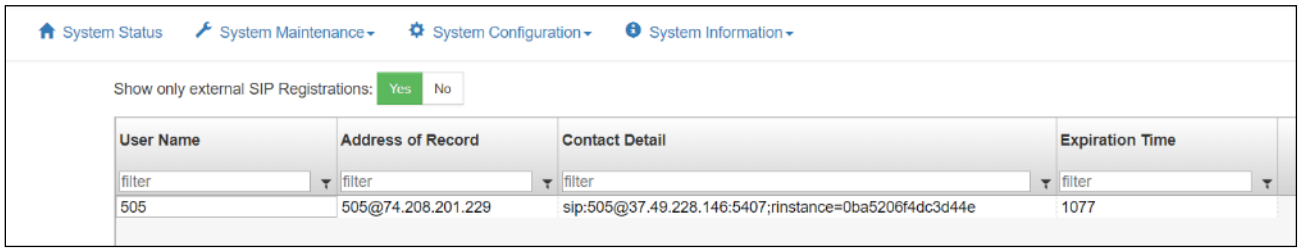
Click the ‘Column Legend’ button in the upper, left-hand side of the ‘Client Statistics’ page to display the legend.



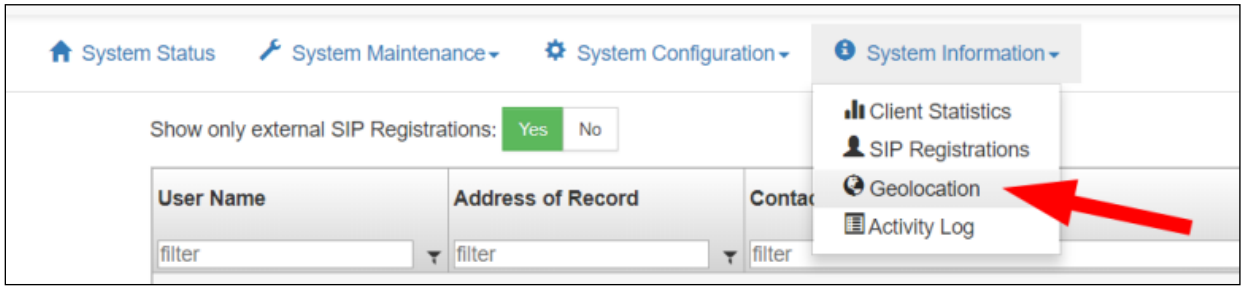


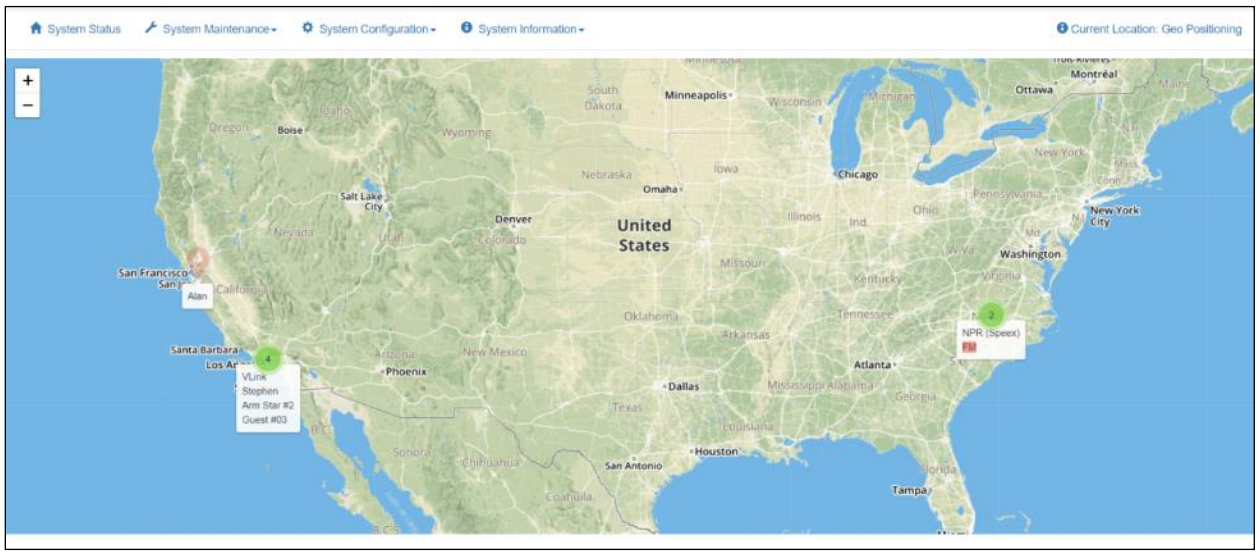
SIP Registrations: Displays all VLINK SIP registrations including user names, address of record, and contact detail



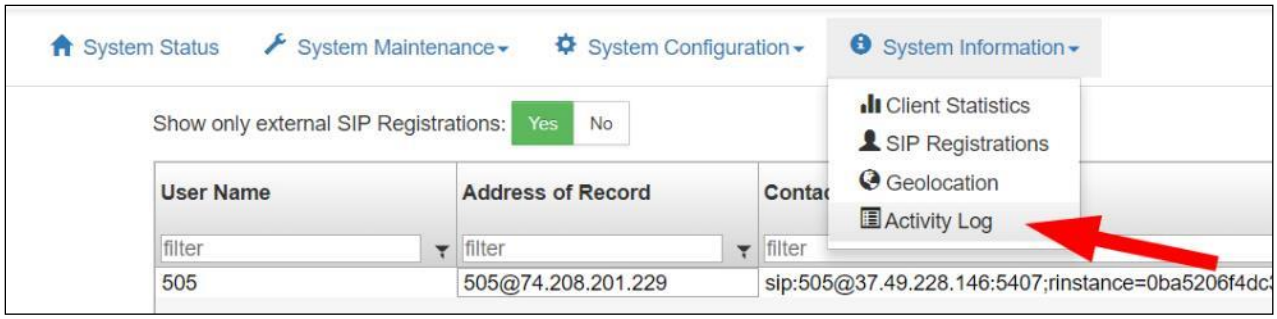


Geolocation: Displays the locations of clients on the system.





Activity Log: During system operation the logging feature displays a time-stamped entry for each client connection/disconnection and key changes


System Status
System Maintenance
System Configuration
System Information
Client Statistics
SIP Registrations
Geolocation
Activity Log
Show only external SIP Registrations:
Yes
No
User Name
Address of Record
filter
505
505@74.208.201.229
sip:505@37.49.228.146:5407;rinstance=0ba5206f4dc

System Status
System Maintenance
System Configuration
System Information
Activity Log
Auto Refresh:
ON
Search:
Date/Time
Message
filter
06/18/18 13:53:16 [SGE]: Listen activated for NPR
06/18/18 13:53:16 [SGE]: Listen activated for MSNBC
06/18/18 13:53:16 [SGE]: Listen deactivated for BBC
06/18/18 13:53:17 [SGE]: Listen activated for BBC
06/18/18 13:53:17 [SGE]: Listen deactivated for BBC
06/18/18 13:53:18 [SGE]: Listen activated for BBC
06/18/18 13:53:20 [SGE]: Listen activated for Fox News
06/18/18 13:53:21 [SGE]: Listen deactivated for Fox News

SUPPORT

Visit our web site at www.intracomsystems.com for general information.

Email us at support@intracomsystems.com for questions not addressed in the sections above or call our technical support hotline at (888) 318-6187.

APPENDIX A: VLINK AND SIP DEVICES

SIP (Session Initiation Protocol) is a widely adopted signaling protocol for Internet conferencing, telephony, presence, events notification, and instant messaging.

The VLINK Virtual Matrix includes an integrated SIP server that allows connection of SIP-enabled, VoIP peripherals directly into the Virtual Matrix from any network connection without the need for a PC. IP phones/phone systems, analog telephone adapters (ATAs), or IP gateways can be seamlessly connected into a network. The VLINK System Administration application can map SIP peripherals to Point-to-Points, Group Calls, and/or Party Lines in any complexity.

Typically SIP devices are used with Telephone IP PBX systems to implement a traditional Telephone system whereby one user dials the number of another user and waits for the recipient to answer. However, as a non-blocking intercom system VLINK is always on and the SIP implementation has been tailored to best compliment this functionality. As such each configured SIP client becomes an extension of a VLINK port by connecting the SIP device to a dedicated virtual SIP client associated with a VLINK port.

Below are a number of common applications in which SIP devices are used with VLINK:

Interoperating ProComm and IP Phone Systems: ProComm Administrators can seamlessly bridge intercom with widely deployed IP Phone Systems that are prevalent in the government and defense markets;

Interfacing Phone Lines: Analog and digital phone lines can be bridged into VLINK from any network connection using SIP-ready VoIP gateways;

Party Line Systems: Users with simple intercommunication requirements that only need to monitor or talk on one or two channels have a wide array of user interface options that do not require a PC. Options include WiFi-enabled IP phones, iPhones (running softphone clients), and desktop speaker phones;

Hoot n' Holler: VLINK is widely deployed as a next generation Hoot n' Holler system replacing expensive, private circuit leased lines, which can have a low quality of service.

In this application most users have the simple requirement of monitoring and talking on a single party line and use an analog telephone attached to an ATA or IP phone, requiring no client-side PCs

Configuring SIP Devices for VLINK

Softphones

Typically softphones will have an account setting or configuration area and a properties tab under which you can point your softphone to a Domain. Find this area and use your VLINK Server IP Address as the Domain, Proxy, and the Registrar address. Unless changed in the VLINK System Administration the standard SIP default port number is 5060. Set the Display Name, User Name, and Password to match the settings programmed in the 'Client Configuration' section of the VLINK System Administration for any given softphone.

Hardphones and ATAs

SIP hardphones and ATAs are configured via a web interface. You need to begin by determining the device IP Address typically by using Internet Explorer to access your router's web interface and view the list of attached devices.

Once you determine your devices IP Address input it into a web browser to access the device's configuration page. Use your VLINK Server IP Address as the Domain, Proxy, and the Registrar address. Unless changed in the VLINK System Administration the standard SIP default port number is 5060. Set the Display Name, User Name, and Password to match the settings programmed in the 'Client Configuration' section of the VLINK System Administration for any given hardphone or ATA.