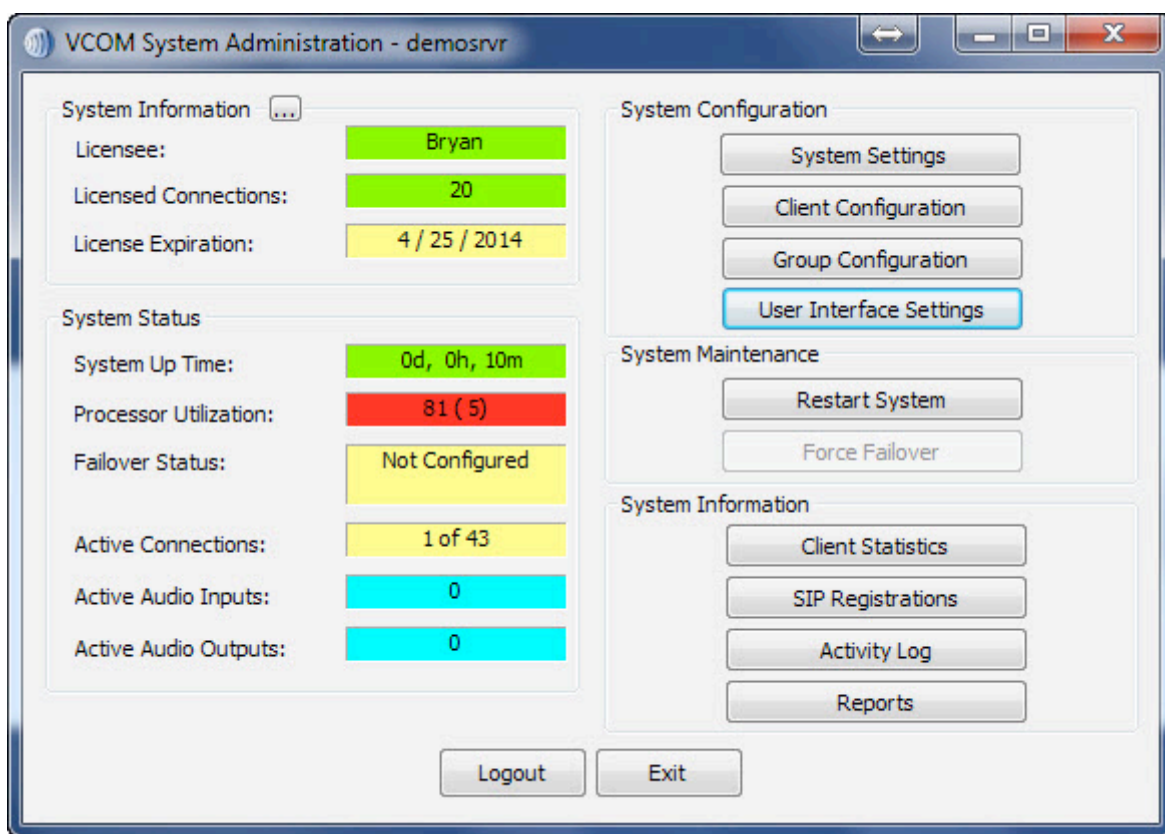# INTRACOM

VCOM System Administration User Guide

U.S. Patent #8660039

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IntraCom Systems, LLC
Ph: (888) 318-6187
support@intracomsystems.com
www.intracomsystems.com

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1. INTRODUCTION

1.1 VCOM System Administration

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

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

1.2 SYSTEM REQUIREMENTS

A. Windows OS

- Hardware Requirements

  o Dedicated: Pentium Celeron 1.0 GHz or equivalent w/ 1 GB RAM

  o Multi-purpose: Pentium 4, 2.0 GHz or equivalent w/ 1 GB RAM

  Lower processing power PCs can be used, however audio quality may be comprised under heavy CPU loads.

- Software Requirement

  o All versions of Windows® XP, Windows® Vista, Windows® 7, including versions for Tablet PCs

- Network Requirement

  o Recommended configuration: 100BaseT connection over private LAN

  o Minimum configuration: 56kbps dial-up connection over public Internet

o Bandwidth Utilization per client:

Audio Sample Rate	Data Rate (Kbps) [ATS=20ms*]	Data Rate (Kbps) [ATS=40ms*]	Data Rate (Kbps) [ATS=60ms*]	Data Rate (Kbps) [ATS=80ms*]	Data Rate (Kbps) [ATS=100ms*]
8 KHz	32	23.6	20.8	19.4	18.56
16 KHz	44.8	36.4	33.6	32.2	31.36
32 KHz	46.8	38.4	35.6	34.2	33.36

*ATS = Audio Time Slice per packet which controls how many 20ms audio frames are transmitted within a single UDP packet. As each UDP packet requires a fixed amount of overhead, the more frames sent at the same time, the less the UDP overhead which conserves network bandwidth. Conversely, the more audio frames sent per transmission, the greater the system latency and the potential audible consequence of a lost packet. The default is 20ms.

- Firewall Requirements

    o Allow TCP connection for data on port 1000 and UDP connection for audio on port 1000

B. Mac OS

- Hardware Requirements

    o Dual Core Intel Processor Macbook, Macbook Pro, or Mac desktop running Windows XP; any Mac running Parallels with suggested 2 GB RAM

    Lower processing power computers can be used, however audio quality may be comprised under heavy CPU loads.

- Software Requirement

    o All versions of Windows® XP, Windows® Vista, Windows® 7, including versions for Tablet PCs

- Network Requirement

    o Recommended configuration: 100BaseT connection over private LAN

    o Minimum configuration: 56kbps dial-up connection over public Internet

o Bandwidth Utilization per client:

Audio Sample Rate	Data Rate (Kbps) [ATS=20ms*]	Data Rate (Kbps) [ATS=40ms*]	Data Rate (Kbps) [ATS=60ms*]	Data Rate (Kbps) [ATS=80ms*]	Data Rate (Kbps) [ATS=100ms*]
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- Firewall Requirements

    o Allow TCP connection for data on port 1000 and UDP connection for audio on port 1000

C. iOS

- Hardware Requirements

    o iPhone, iPad, iPod, 3G and newer

        - It is recommended that you have a minimum 10MB of free storage memory on your device to install the VCOM Control Panel for Mobile.

- Software Requirement

    o iOS rev 3.0 or newer

- Network Requirements

    o Recommended configuration: WiFi - 802.11

    o 3G Data Cellular Connection

        - Please note that data usage costs apply for using VCOM over 3G mobile networks (EDGE, EV-DO and UMTS), so we recommend an unlimited data plan.

- o Bandwidth Utilization per client:

Audio Sample Rate	Data Rate (Kbps) [ATS=20ms*]	Data Rate (Kbps) [ATS=40ms*]	Data Rate (Kbps) [ATS=60ms*]	Data Rate (Kbps) [ATS=80ms*]	Data Rate (Kbps) [ATS=100ms*]
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- Firewall Requirements

Allow TCP connection for data on port 1000 and UDP connection for audio on port 1000

D. Android OS

- Hardware Requirements

    - o Many Android platforms running OS rev 2.2 or newer (note: Kindle Fire HD is NOT suitable for use with VCOM)

        - It is recommended that you have a minimum 10MB of free storage memory on your device to install the VCOM Control Panel for Mobile.

- Software Requirement

    - o Android rev 2.2 or newer, rev 4.1 Jelly Bean or newer recommended

- Network Requirements

    - o Recommended configuration: WiFi - 802.11

    - o 3G Data Cellular Connection

        - Please note that data usage costs apply for using VCOM over 3G mobile networks (EDGE, EV-DO and UMTS), so we recommend an unlimited data plan.

o Bandwidth Utilization per client:

Audio Sample Rate	Data Rate (Kbps) [ATS=20ms*]	Data Rate (Kbps) [ATS=40ms*]	Data Rate (Kbps) [ATS=60ms*]	Data Rate (Kbps) [ATS=80ms*]	Data Rate (Kbps) [ATS=100ms*]
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*ATS = Audio Time Slice per packet which controls how many 20ms audio frames are transmitted within a single UDP packet. As each UDP packet requires a fixed amount of overhead, the more frames sent at the same time, the less the UDP overhead which conserves network bandwidth. Conversely, the more audio frames sent per transmission, the greater the system latency and the potential audible consequence of a lost packet. The default is 20ms.

- Firewall Requirements

Allow TCP connection for data on port 1000 and UDP connection for audio on port 1000

2. INSTALLATION

Locate the VCOM System Administration setup application, typically named 'VCOM_System_Administration_Setup.exe,' that was provided either electronically or on CD.

Run the installer and follow the prompts. You will need to accept IntraCom Systems' License Agreement to install the software.

To open the VCOM System Administration application once installed, click on your 'VCOM System Administration' shortcut icon on your desktop or click on your start menu and select 'All Programs.' Find 'IntraCom' and select 'VCOM System Administration.'

Before logging in the first time you will need to configure the System Administration application to communicate with the Virtual Matrix. Click on the 'Configure' button and configure the IP address of the Virtual Matrix. If you are trying to access the server remotely and your network uses authentication, then you will need to click the check mark box to use Domain Authentication. When completed, click 'apply'.
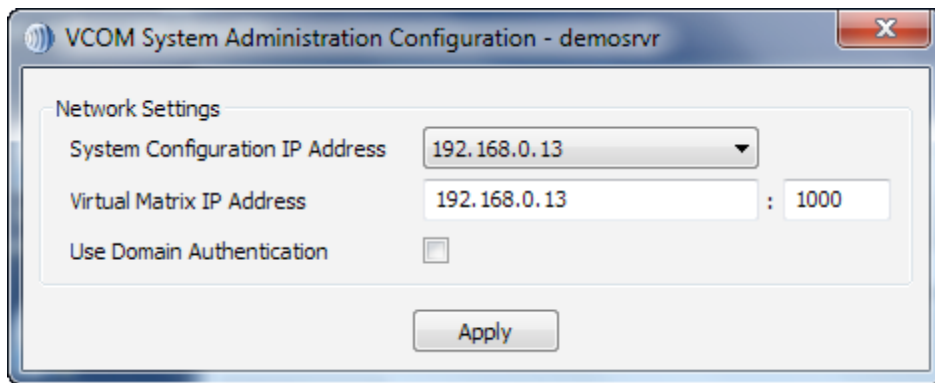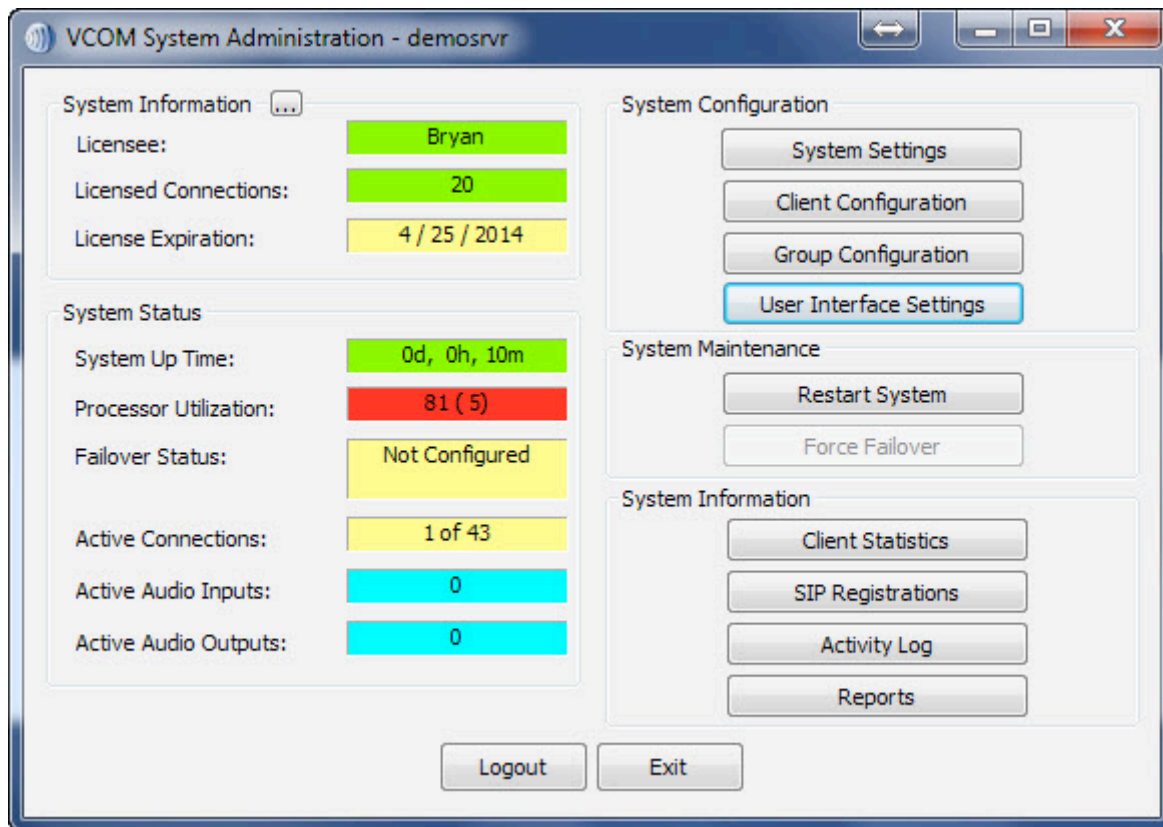
The default master 'Login Name' is 'admin' and there is no default master 'Login Password,' so leave the field blank. Once logged in, credentials can be changed.
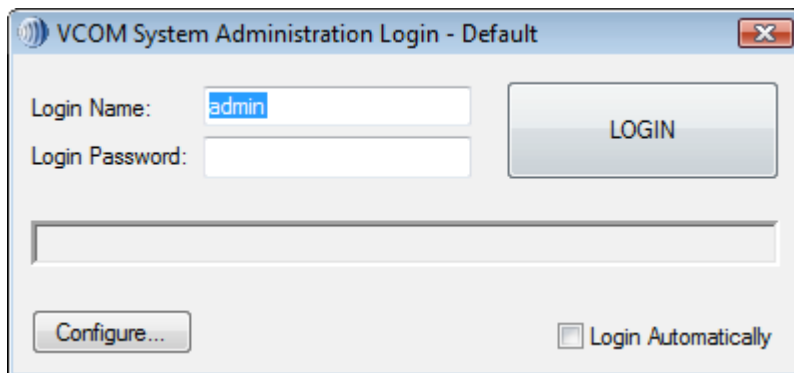
Windows OS screen. Other operating systems have similar information, but may appear slightly different.
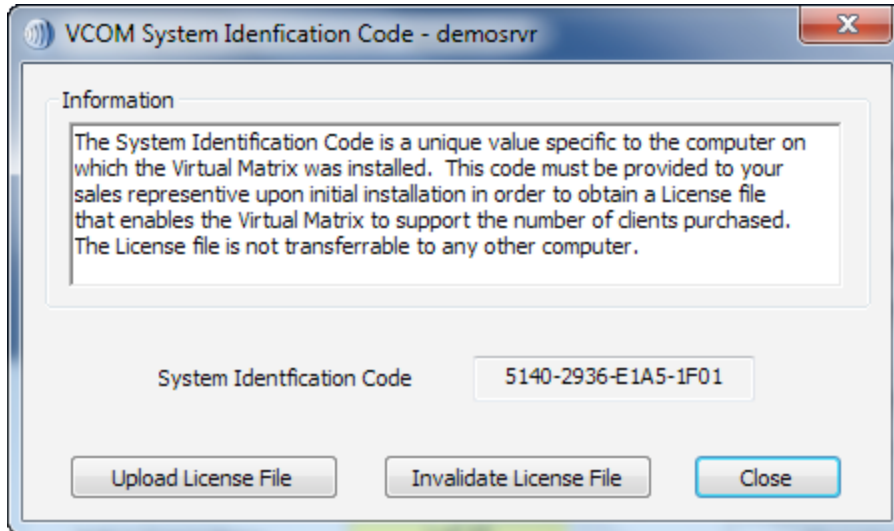
3. Operation

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

3.1 SYSTEM INFORMATION

System Information fields are color coded. Red indicates a problem, yellow a potential problem, and green normal operations.

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.

Click '⋯' to view the '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 File', locate the file, and select 'Open' in the Windows Open File dialog box.

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

3.2 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' area in the lower, left-hand side of the System Administration main window displays general system metrics.

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

- Processor Utilization: Displays CPU utilization of the server or PC hosting the VCOM 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 VCOM Control Panels, VCOM 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.

3.3 SYSTEM CONFIGURATION

System Settings

The 'System Settings' found in the 'System Configuration' area in the upper, right-hand side of the System Administration main window is used to set system-wide parameters.

VCOM System Settings - local

Master System Administrator Login
Login Name: admin
Login Password:

Primary Server Network Settings
Server IP Address (Local Network Interface): 192.168.0.10
Server NAT IP Address:
Server IP Ports for Client Data / Audio: 1000 / 1000
Server IP Ports for SIP Data / RTP Audio Base: 5060 / 0
Server SIP Domain Name:

Secondary (Failover) Server Network Settings
Server IP Address:
Server NAT IP Address:
Server IP Port for VCOM Client Data / Audio: /
Server IP Port for Failover Data: 1001

Audio Settings
Audio Mix Sample Rate: Ultra Wideband (32 KHz)
Audio Output Level Gain (Post-Mix): 0 dB

Voice Activity Indication
Voice Activity Indication Color: Text | Background

Failover Settings
Failover Activation Delay: 0 (seconds)
Automatic Failback: Never. Failback done manually (hh:mm)

Geo Mapping Server Network Settings
Geo Mapping Server IP Address:
Geo Mapping Server IP Port: 8090

OK | Cancel

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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 VCOM Client Data under 'Server IP Port for VCOM 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 VCOM 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 VCOM 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

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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 VCOM 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 VCOM Client Data under 'Server IP Port for VCOM Client Data / Audio'

- Enter the Server IP Port for Client Audio under 'Server IP Port for VCOM 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 VCOM 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.

Client Configuration

The 'Client Configuration' found in the 'System Configuration' area is used to set system-wide parameters.
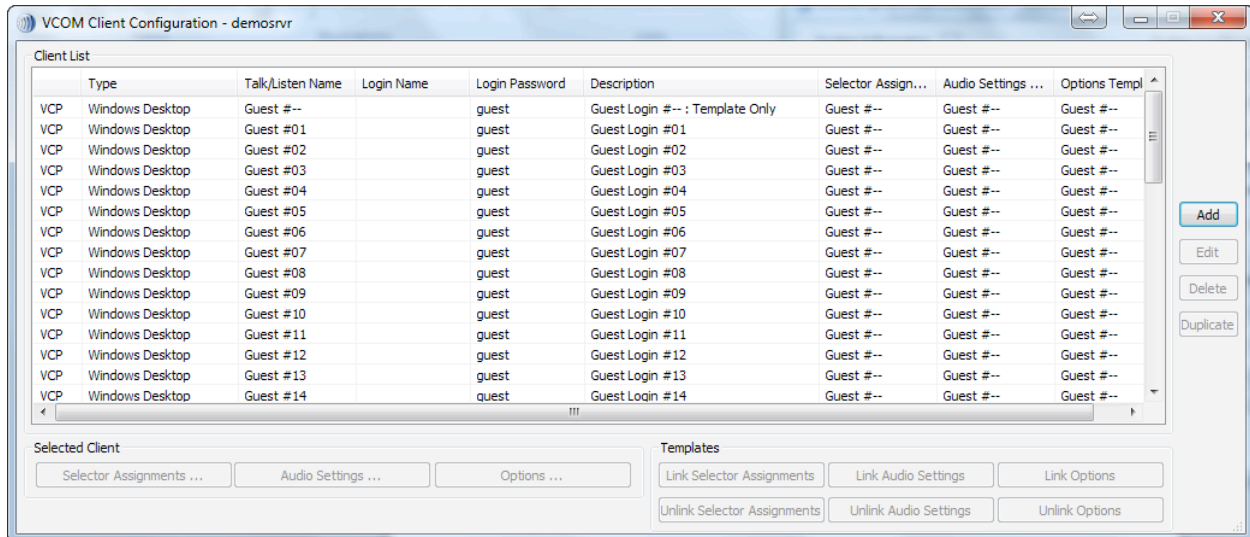
The upper section of the 'Client Configuration' window displays all configured users and devices, log in names, passwords, selector labels, client type, and if the given channel is set as a party line.

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.

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

Under 'Add' or 'Edit' you can set or change the following settings:

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

    o '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.

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

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

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

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

- o 'SIP Device: Intersystem/PBX Registered Trunk': enables a connection between PBX/SIP Servers and VCOM using Registration of clients.

- o 'SIP Device: Four Wire Interface': enables connection of a SIP based Four Wire Interface to transport audio into VCOM.

- 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: This is used when a Client is outside the LAN, **and** your LAN uses authentication to provide secure connections to remote users. This check mark box will also need to be checked on the client's login configuration.

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 VCOM Control Panel that is configured to hide the off-line selectors. It is generally used for VCOM Device Interface clients that should, under normal operation, never go off-line thus allowing VCOM 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.

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- 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.

Selector Assignments:
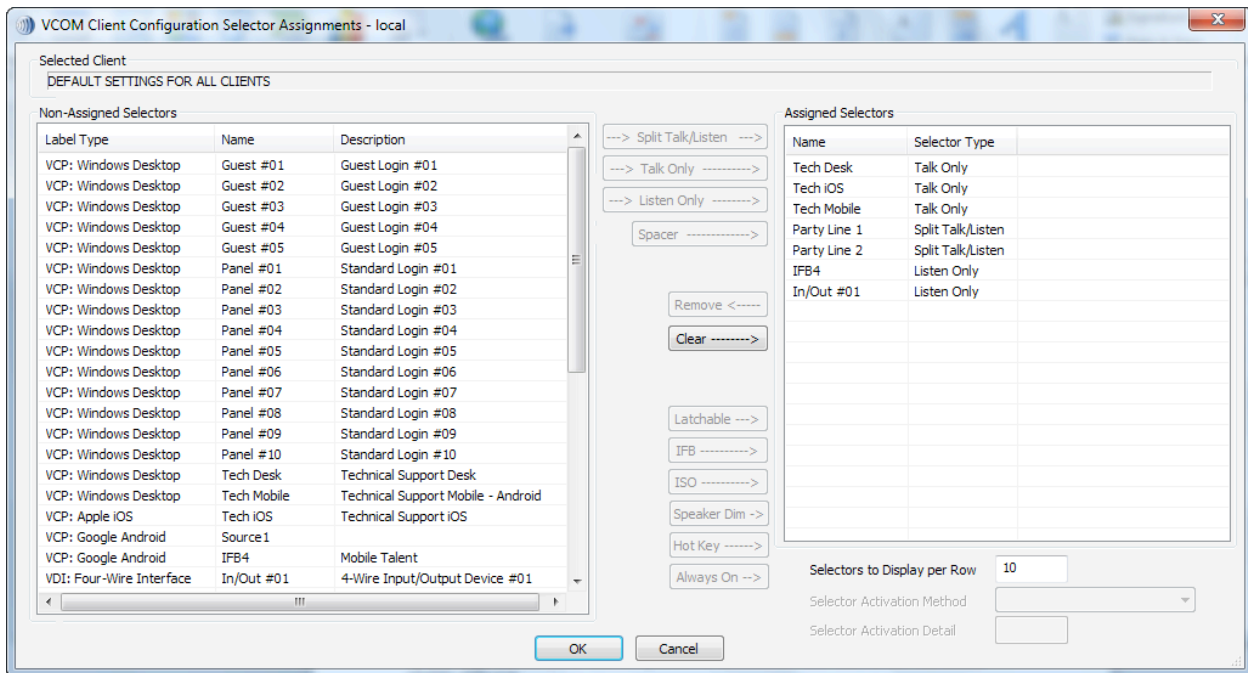
Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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.

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- 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 VCOM Control Panel.

- 'Selectors to Display per Row' dictates the number of selectors to be displayed per row on a VCOM Control Panel.

- '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:

  o 'On This Client Connect' – This default option results in the selectors be activated as long as the client is connected.

  o '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.

  o '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.

  o '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.

  o '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.

Audio Settings:
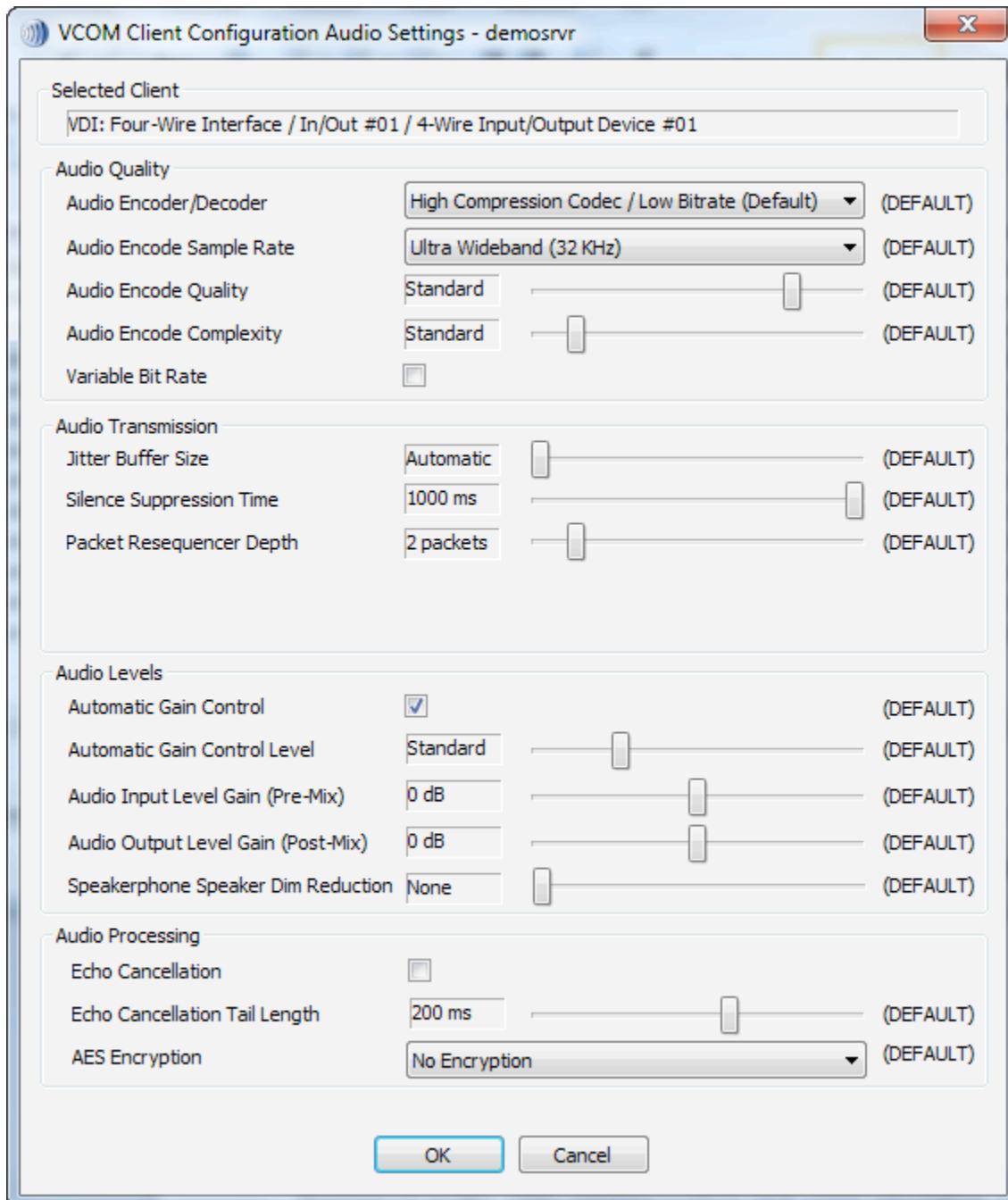
Windows OS screen. Other operating systems have similar information, but may appear slightly different.

- Audio Encoder/Decoder: This setting allows selection of a different encoder/decoder. For VCOM Control Panels and VCOM 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 VCOM 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 VCOM 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.

- 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.

- Audio Capture Buffer Size: The option controls the size of the audio capture buffer used by the VCOM Control Panel and VCOM Device Interface when using the Microsoft WDM drivers. This default value is

20 ms. With some slower computers and/or handhelds the audio capture buffer size may need to be increased to prevent audio overrun issues. However, it should be noted that increasing this value also increases the audio latency.

- Audio Time Slice Per Packet: Controls how many 20ms audio frames are transmitted within a single UDP packet. As each UDP packet represents a fixed amount of overhead, the more frames sent at the same time the less UDP overhead, which conserves network bandwidth. Conversely, the more frames sent per transmission, the greater the system latency and audible consequence of lost packets, *e.g.,* 20ms of lost frames is generally hardly audible, whereas 40ms typically is.

- Jitter Buffer Minimum 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: 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.

- 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.

- 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.

Client Options

VCOM Client Configuration Options - demosrvr			✕

Selected Client
VCP: Windows Desktop / Panel #01 / Standard Login #01

Control Panel Options
Allow Disable Client Login / Connection	☐	(DEFAULT)
Hide disabled selectors	☐	(DEFAULT)
Hide selector legends	☐	(DEFAULT)
Voice Activity Indication	☑	(DEFAULT)
Split Selector Center Zone	☑	
Ring Tone Disable	☐	
Automatic Page Switch On Call Notification	☐	

Control Panel Function Buttons
Enable Party Line Assignment Button	☑
Enable Selector Volume Buttons	☑
Enable Dial Pad Button	☑
Enable Selector Assignment Button	☑
Enable Selector Relocation Button	☑
Enable Reveal Inactive Selectors Button	☑
Enable Launch Geo Mapping Button	☑

Control Panel for iOS Options
Restart in Background if Previously Active	☐	(DEFAULT)
Operate as a Beltpack	☐	
Disable Integrated Mic and Speaker	☐	

Control Panel / Device Interface Options
| Voice Activity Detection Time In Ms | 100 | (DEFAULT) |

Geo Mapping
Geo Mapping Disable	☐
Geo Mapping Latitude (Fixed Position)	
Geo Mapping Longitude (Fixed Position)	

[OK]   [Cancel]

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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.

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.

SIP Options

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

- 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 VCOM 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 VCOM 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 VCOM 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 VCOM 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 VCOM 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 VCOM 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 VCOM 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.

- 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.

- 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.

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.
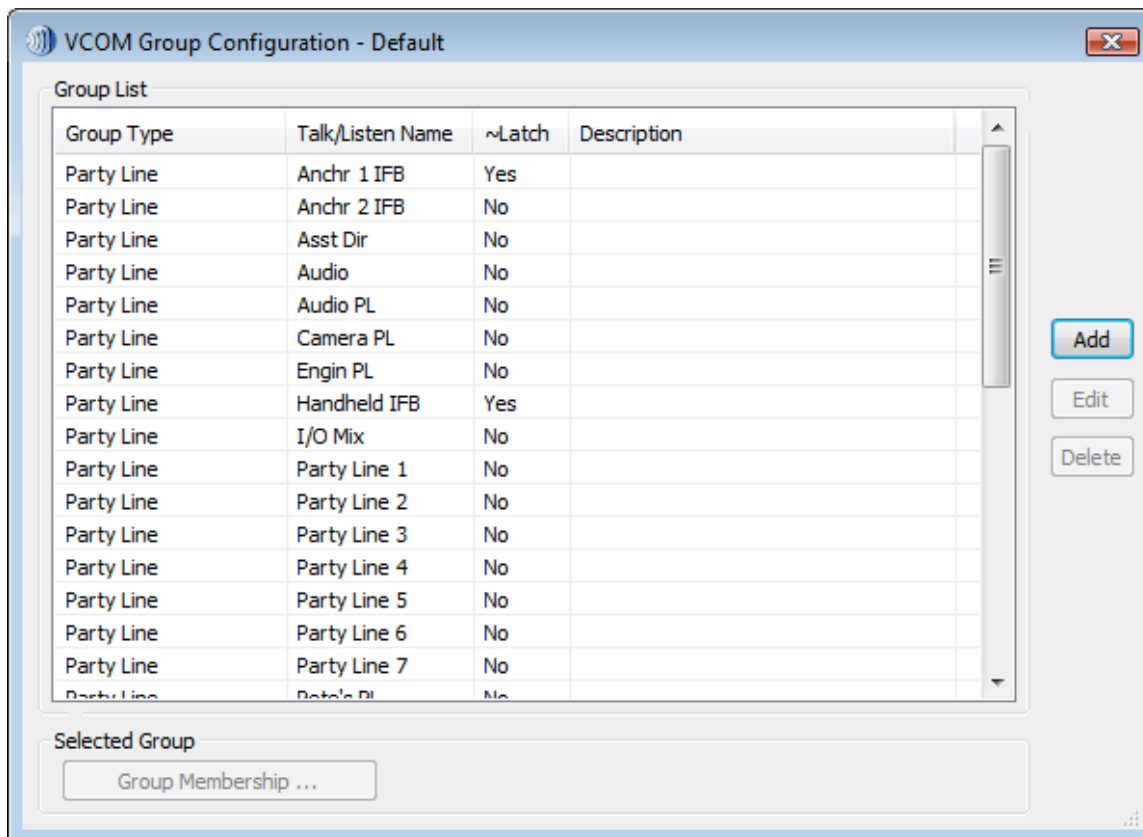
Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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.

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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.
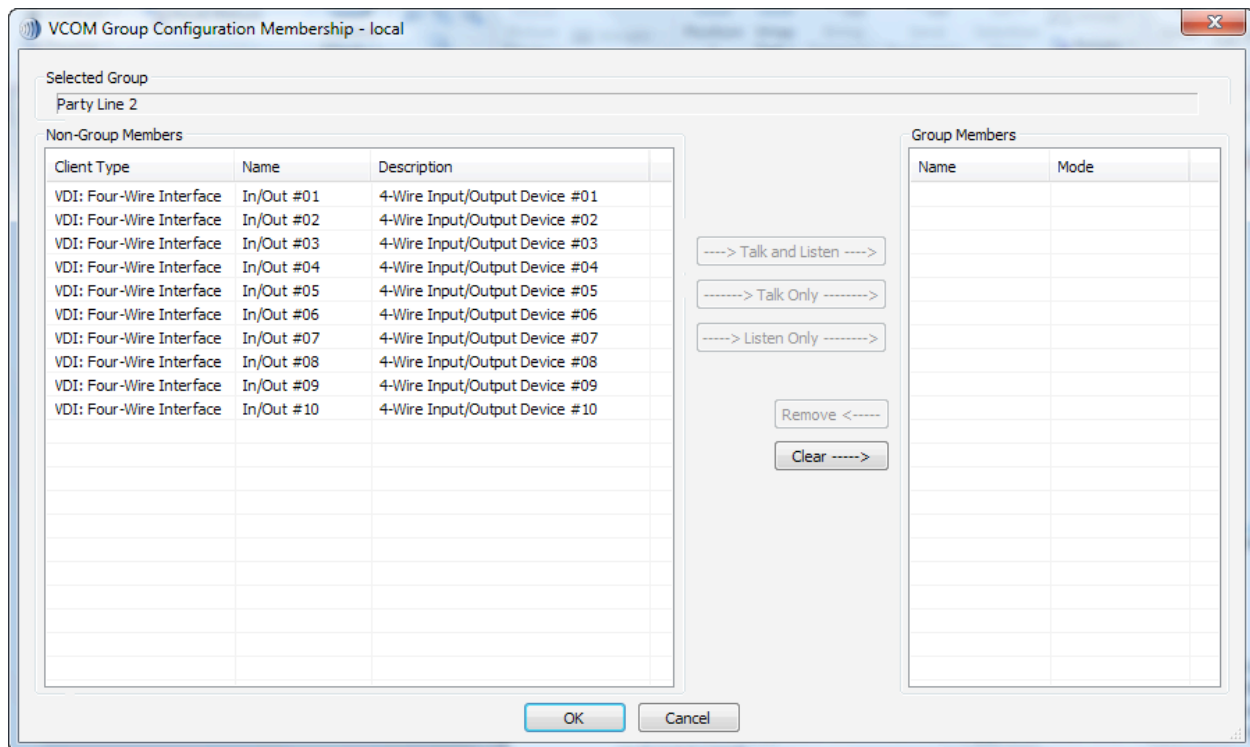
Windows OS screen. Other operating systems have similar information, but may appear slightly different.

User Interface Settings:

This allows customizations of the VCP GUI such as the Colors(Background and Text), Selector Spacing, and Graphics that can be uploaded, like a business logo.
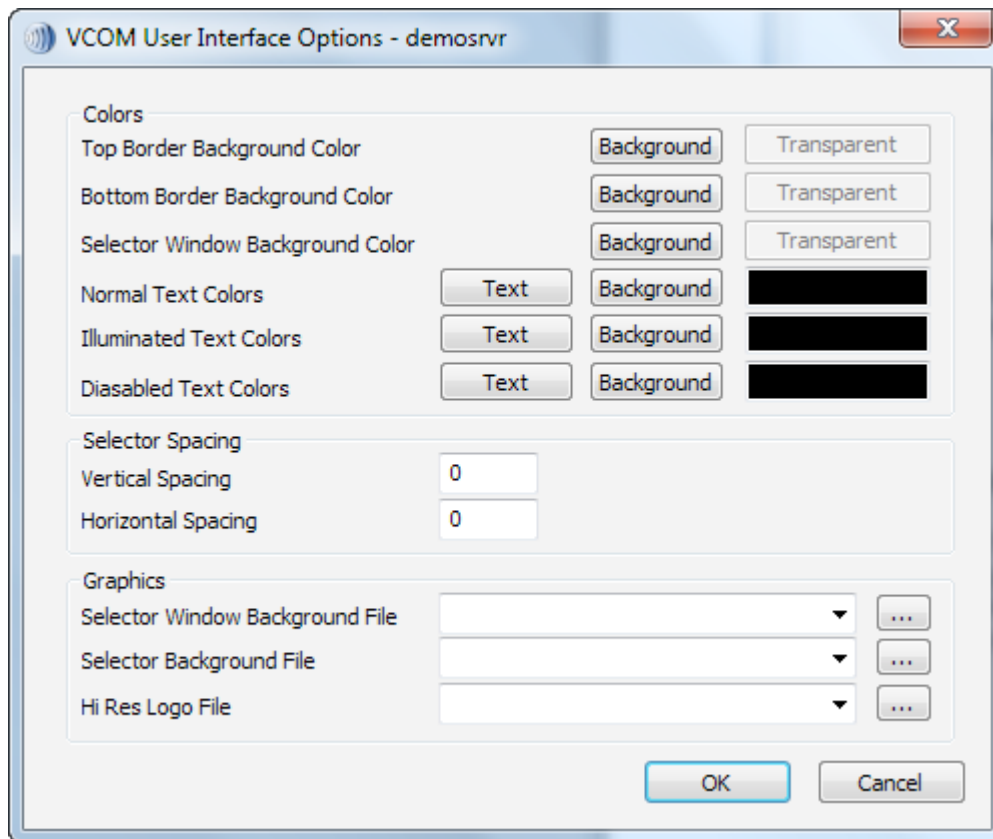
3.4 <u>SYSTEM MAINTENANCE</u>

- 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.
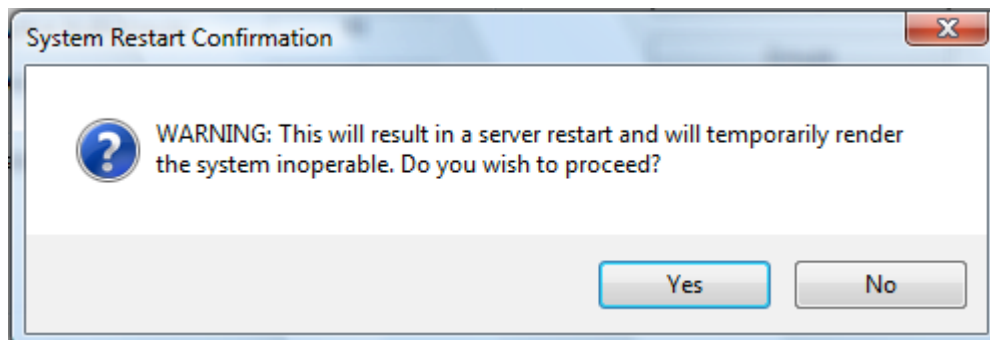
Windows OS screen. Other operating systems have similar information, but may appear slightly different.

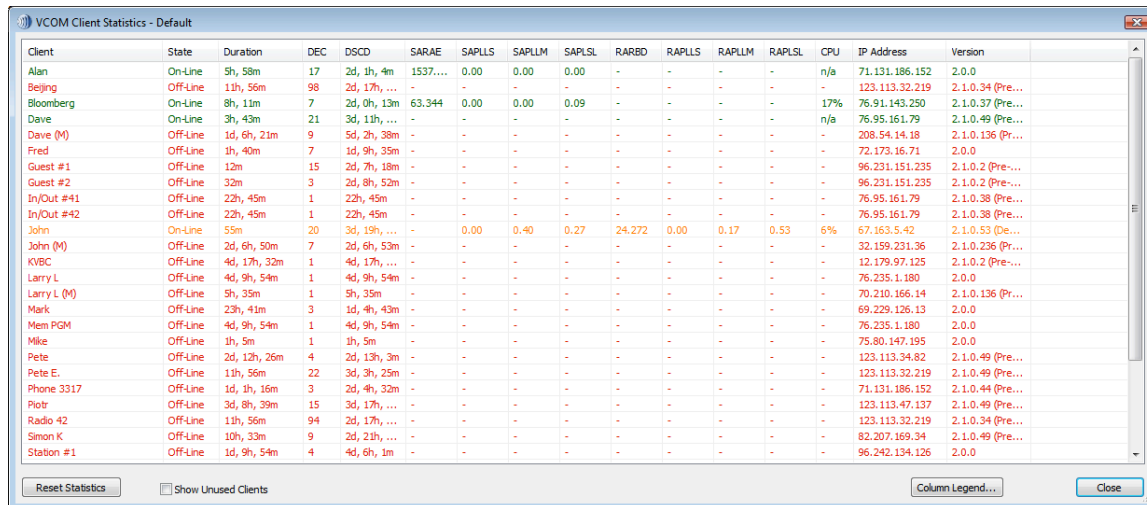- Force Failover: Forces the Primary to relinquish control to the Secondary Server.

3.5 SYSTEM STATISTICS:

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

Client	State	Duration	DEC	DSCD	SARAE	SAPLLS	SAPLLM	SAPLSL	RARBD	RAPLLS	RAPLLM	RAPLSL	CPU	IP Address	Version
Alan	On-Line	5h, 58m	17	2d, 1h, 4m	1537....	0.00	0.00	0.00	-	-	-	-	n/a	71.131.186.152	2.0.0
Beijing	Off-Line	11h, 56m	98	2d, 17h, ...	-	-	-	-	-	-	-	-	-	123.113.32.219	2.1.0.34 (Pre...)
Bloomberg	On-Line	8h, 11m	7	2d, 0h, 13m	63.344	0.00	0.00	0.09	-	-	-	-	17%	76.91.143.250	2.1.0.37 (Pre...)
Dave	On-Line	3h, 43m	21	3d, 11h, ...	-	-	-	-	-	-	-	-	n/a	76.95.161.79	2.1.0.49 (Pre...)
Dave (M)	Off-Line	1d, 6h, 21m	9	5d, 2h, 38m	-	-	-	-	-	-	-	-	-	208.54.14.18	2.1.0.136 (Pr...)
Fred	Off-Line	1h, 40m	7	1d, 9h, 35m	-	-	-	-	-	-	-	-	-	72.173.16.71	2.0.0
Guest #1	Off-Line	12m	15	2d, 7h, 18m	-	-	-	-	-	-	-	-	-	96.231.151.235	2.1.0.2 (Pre-...)
Guest #2	Off-Line	32m	3	2d, 8h, 52m	-	-	-	-	-	-	-	-	-	96.231.151.235	2.1.0.2 (Pre-...)
In/Out #41	Off-Line	22h, 45m	1	22h, 45m	-	-	-	-	-	-	-	-	-	76.95.161.79	2.1.0.38 (Pre...)
In/Out #42	Off-Line	22h, 45m	1	22h, 45m	-	-	-	-	-	-	-	-	-	76.95.161.79	2.1.0.38 (Pre...)
John	On-Line	55m	20	3d, 19h, ...	-	0.00	0.40	0.27	24.272	0.00	0.17	0.53	6%	67.163.5.42	2.1.0.53 (De...)
John (M)	Off-Line	2d, 6h, 50m	7	2d, 6h, 53m	-	-	-	-	-	-	-	-	-	32.159.231.36	2.1.0.236 (Pr...)
KVBC	Off-Line	4d, 17h, 32m	1	4d, 17h, ...	-	-	-	-	-	-	-	-	-	12.179.97.125	2.1.0.2 (Pre-...)
Larry L	Off-Line	4d, 9h, 54m	1	4d, 9h, 54m	-	-	-	-	-	-	-	-	-	76.235.1.180	2.0.0
Larry L (M)	Off-Line	5h, 35m	1	5h, 35m	-	-	-	-	-	-	-	-	-	70.210.166.14	2.1.0.136 (Pr...)
Mark	Off-Line	23h, 41m	3	1d, 4h, 43m	-	-	-	-	-	-	-	-	-	69.229.126.13	2.0.0
Mem PGM	Off-Line	4d, 9h, 54m	1	4d, 9h, 54m	-	-	-	-	-	-	-	-	-	76.235.1.180	2.0.0
Mike	Off-Line	1h, 5m	1	1h, 5m	-	-	-	-	-	-	-	-	-	75.80.147.195	2.0.0
Pete	Off-Line	2d, 12h, 26m	4	2d, 13h, 3m	-	-	-	-	-	-	-	-	-	123.113.34.82	2.1.0.49 (Pre...)
Pete E.	Off-Line	11h, 56m	22	3d, 3h, 25m	-	-	-	-	-	-	-	-	-	123.113.32.219	2.1.0.49 (Pre...)
Phone 3317	Off-Line	1d, 1h, 16m	3	2d, 4h, 32m	-	-	-	-	-	-	-	-	-	71.131.186.152	2.1.0.44 (Pre...)
Piotr	Off-Line	3d, 8h, 39m	15	3d, 17h, ...	-	-	-	-	-	-	-	-	-	123.113.47.137	2.1.0.49 (Pre...)
Radio 42	Off-Line	11h, 56m	94	2d, 17h, ...	-	-	-	-	-	-	-	-	-	123.113.32.219	2.1.0.34 (Pre...)
Simon K	Off-Line	10h, 33m	9	2d, 21h, ...	-	-	-	-	-	-	-	-	-	82.207.169.34	2.1.0.49 (Pre...)
Station #1	Off-Line	1d, 9h, 54m	4	4d, 6h, 1m	-	-	-	-	-	-	-	-	-	96.242.134.126	2.0.0

Reset Statistics ☐ Show Unused Clients   Column Legend...   Close

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

Click the 'Reset Statistics' button in the lower, 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 lower, right-hand side of the 'Client Statistics' window to display the legend.

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

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

Windows OS screen. Other operating systems have similar information, but may appear slightly different.

Reports:

This allows you to view/print Party Line Memberships and Selector Assignments to record configurations.
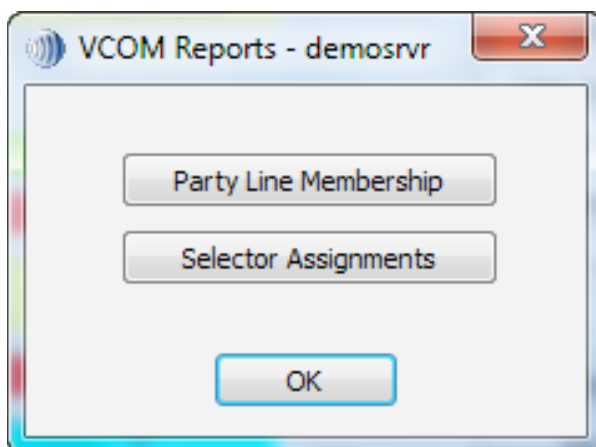
4. SUPPORT

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
VCOM and SIP

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

The VCOM 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 VCOM 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 VCOM 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 VCOM port by connecting the SIP device to a dedicated virtual SIP client associated with a VCOM port.

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

- 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 VCOM 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: VCOM 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 VCOM

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 VCOM Server IP Address as the Domain, Proxy, and the Registrar address. Unless changed in the VCOM 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 VCOM 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 VCOM Server IP Address as the Domain, Proxy, and the Registrar address. Unless changed in the VCOM 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 VCOM System Administration for any given hardphone or ATA.