Innovating the Future of Global Communications

VLINK Server SIP Setup Guide

Bosch Security Systems

www.rtsintercoms.com

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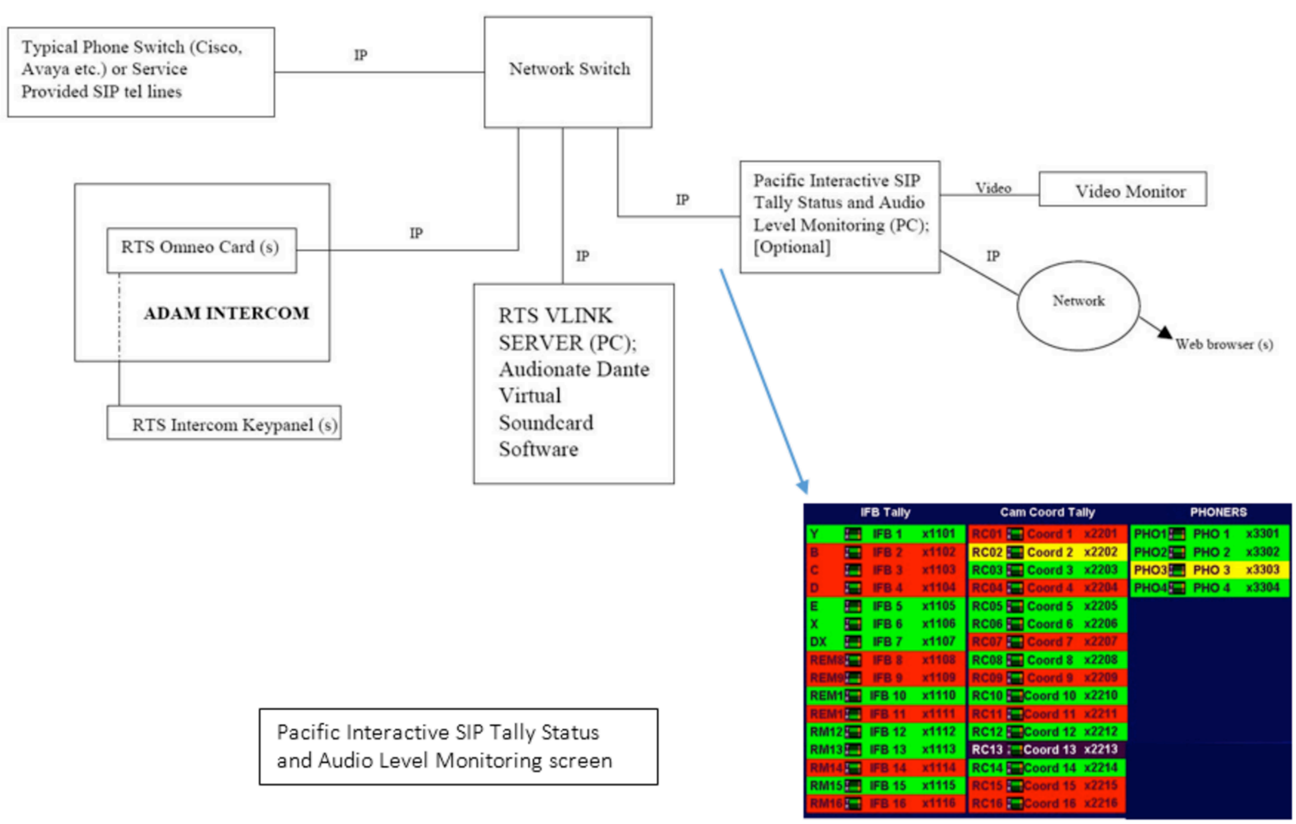
1. INTRODUCTION

VLINK provides seamless integration between RTS ADAM intercoms and many compatible external communications methods. This document provides the basic outline of configurations required to use VLINK to connect ADAM to SIP phone systems and provide intelligent control and status of phone lines.

Please refer to your phone system administrator or SIP phone service provider for configuration information needed to properly connect SIP lines from your VLINK Server, and which may be required on the phone system side for proper interconnect.

Below is a block diagram of the system described in this document. A typical phone switch or SIP lines from a phone service provider are connected to a data network. Also connected to this common network is an RTS VLINK Server and RTS Omneo Dante audio card(s) in an ADAM intercom. RTS key panel(s) can either be connected via analog, or via network using RVON VOIP or Omneo Dante audio to the RTS intercom.
Optionally, a PC running a software package designed for line status descriptions, tally, and audio level metering, including a browser-based display, may be connected to the network. Learn more about the optional tally display at www.pacint.com/broadcast .

RTS SIP INTERFACE SOLUTION (Dante) BLOCK DIAGRAM



2. INITIAL PLANNING

Phone Switch SIP Line Planning

The first step in a successful setup is to establish the number of phone lines to be configured and determine the details of the phone switch or SIP service provider being used. VLINK will generally interface with any typical phone switch or service provider.

In each case there will be an **IP address** for the SIP line services, and most likely **user name** and **password** access information for each line being configured.

Before moving forward, please ensure this preliminary information is on hand for entry into the configuration screens described in this document.

IP Network Considerations

A critical element in successful planning is the IP network connectivity required for the system components to link to each other. Whether the SIP lines are provided by an external SIP service provider, or internal phone switch, the VLINK PC must have direct IP connectivity with at minimum TCP/UDP Port 5060 open between the VLINK Server PC and the source of the SIP lines as well as public internet access.

Additionally, the RTS ADAM Omneo card(s) and master controller cards, AZEDIT configuration PC, and optional Pacific Interactive TIF Tally display must have IP connectivity between each other and the Server.

An important note about Dante audio is that it requires DIRECT IP connectivity on the same local subnet. For this reason, the VLINK Server PC and the RTS Omneo cards should be plugged directly into the same network switch for best results. Other components can be plugged in anywhere on a network, so long as IP connectivity between them is available. If network connections between components other than the VLINK Server and Omneo card (s) are routed and/or protected by firewalls, certain TCP and UDP ports must be configured as passable between components. Please refer to corresponding network configuration sections of documents pertaining to AZEDIT and ADAM network setup.

VLINK Server port licensing

Also before proceeding please ensure that minimally your VLINK Server is licensed for TWO ports per SIP line being interfaced. The first port will connect to the SIP phone switch or service provider, and the second port will interface that line into the RTS ADAM intercom. Additional VLINK port licenses are preferred to allow for local Server control panels used for testing, control, and diagnostics.

Optional Pacific Interactive TIF Tally Display

While VLINK and ADAM configuration can proceed and display capabilities added at a later date if desired, the display is a great way to confirm and operationally monitor the status of all SIP lines calls. See www.pacint.com/broadcast and contact Pacific Interactive for pricing and further details.

Audio Connectivity between VLINK Server and RTS ADAM intercom

There are a variety of possible audio connection methods between the VLINK Server and RTS ADAM intercom, including Dante, Analog and MADI. MADI interface from the VLINK Server to an ADAM intercom requires a MADI PC interface card: http://www.rme-audio.de/en/products/hdspe_madi.php . This document details using Audinate's DANTE Virtual Soundcard (DVS) on the VLINK Server PC. Using the Dante audio standard, the connection to an ADAM intercom can then either be via analog or RTS' Omneo Dante interface card. To use analog, a conversion box such as Focusrite's Rednet 1 or 2 can be used: https://us.focusrite.com/ethernet-audio-interfaces/rednet-1; https://us.focusrite.com/ethernet-audio-interfaces/rednet-2 However, it is recommended using RTS' Omneo Dante interface card for the ADAM intercom as described in this document as setup will be simpler with superior results. Using Dante audio requires purchasing an Audinate Dante Virtual Soundcard license for the VLINK Server PC (current cost USD $29). Using one or more RTS Dante Omneo cards in the ADAM intercom requires the installation of Omneo card(s) in the ADAM intercom. Note the Omneo cards can be purchased or upgraded with 16, 32, 48 or 64 licensed channels per card. Contact your RTS sales representative for more details.

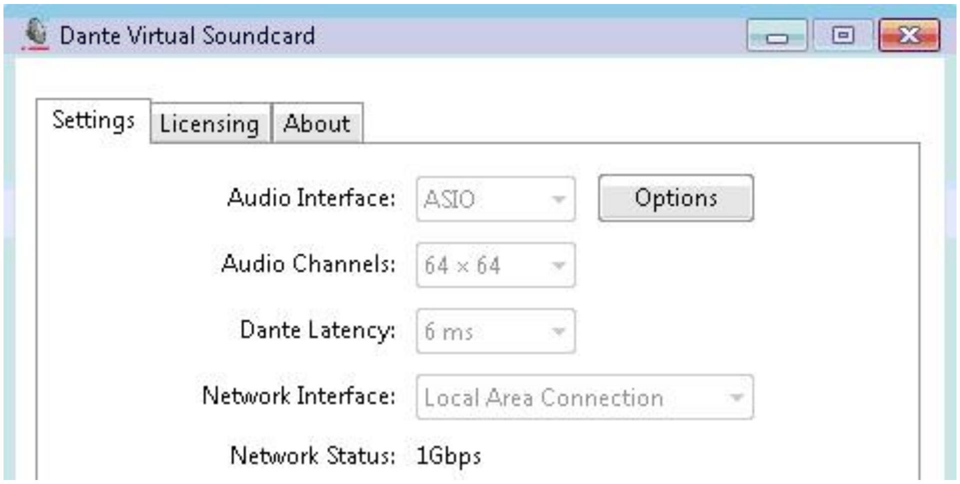
Audinate Dante Virtual Soundcard (DVS)

This document details audio connectivity between the VLINK Server and RTS intercom utilizing Audionate's Dante audio protocol. The first step in preparation for this connectivity is to download and install two software packages onto the VLINK Server PC.

From the VLINK Server PC, go to: https://www.audinate.com/user/register and register with Audinate. Registration is free. Note the user information, including e-mail provided will be used for access and to define the licensee and to install and activate the software.

After registering, go to: https://www.audinate.com/node/123 and download and install the Dante configuration tool, which is available at no charge. This tool will be used to "map" the audio channels between the VLINK Server and the RTS Omneo card.
Next, go to: https://www.audinate.com/products/software/dante-virtual-soundcard and purchase a Dante Virtual Soundcard (DVS) license. The full price for unlimited use is USD $29.00.
Next install the soundcard software, run it and activate using the license key provided.

Once activated, select setup parameters as shown below:
Audio Interface = ASIO
Audio Channel = 64 x 64
Dante Latency = 6ms
Network Interface = (choose the appropriate Ethernet connection the PC uses)

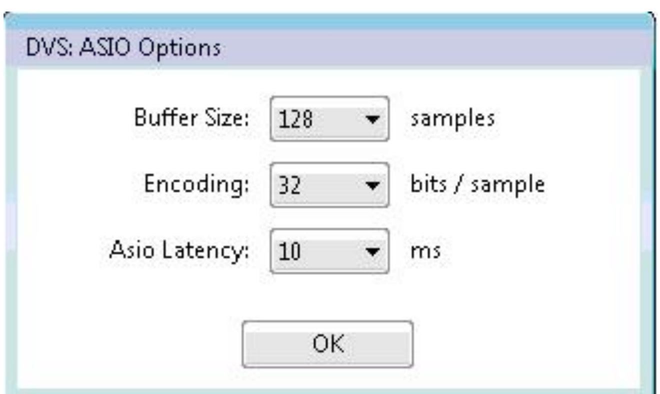


Next, click the options button and configure the fields as shown:

Buffer size = 128 samples

Encoding = 32 bits / sample

Asio Latency = 10 ms



The Dante Virtual Soundcard on your VLINK Server is now ready for use.

RTS intercom port planning

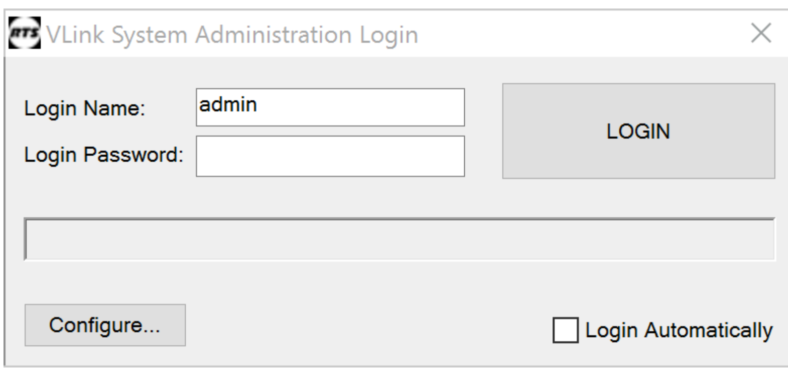
Based on the number of lines being interfaced, and the number of channels on the RTS OMNEO card, it is important at an early stage to clearly identify which phone line will connect to which intercom port. For example, if a 16 channel Omneo card is used, and it is installed as intercom ports 97-112, and the phone numbers being interfaced are 999-999-1001 thru 999-999-1016, you should determine and note which line will be on which RTS ADAM port. For this example, 999-999-1001 might connect to intercom port 97, 999-999-1002 to port 98 thru 999-999-1016 connecting to intercom port 112.

Also note that VLINK SIP Server features require ADAM Master Controller software version x.x.x or later, and AZEDIT version x.x.x or later. Contact your RTS support representative should you require this software or assistance with upgrades.

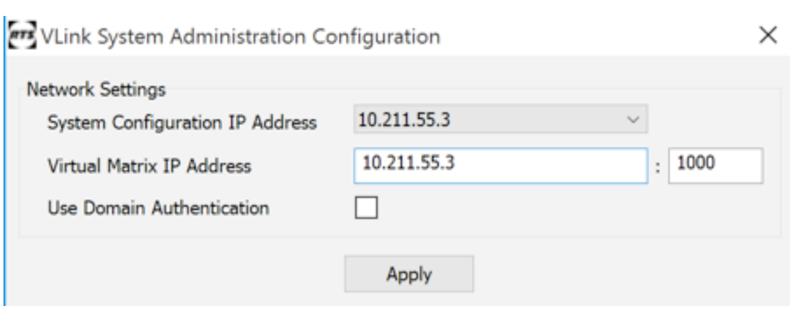
3. VLINK SERVER CONFIG: VLINK SERVER TO SIP / PHONE SWITCH

General VLINK System Configuration

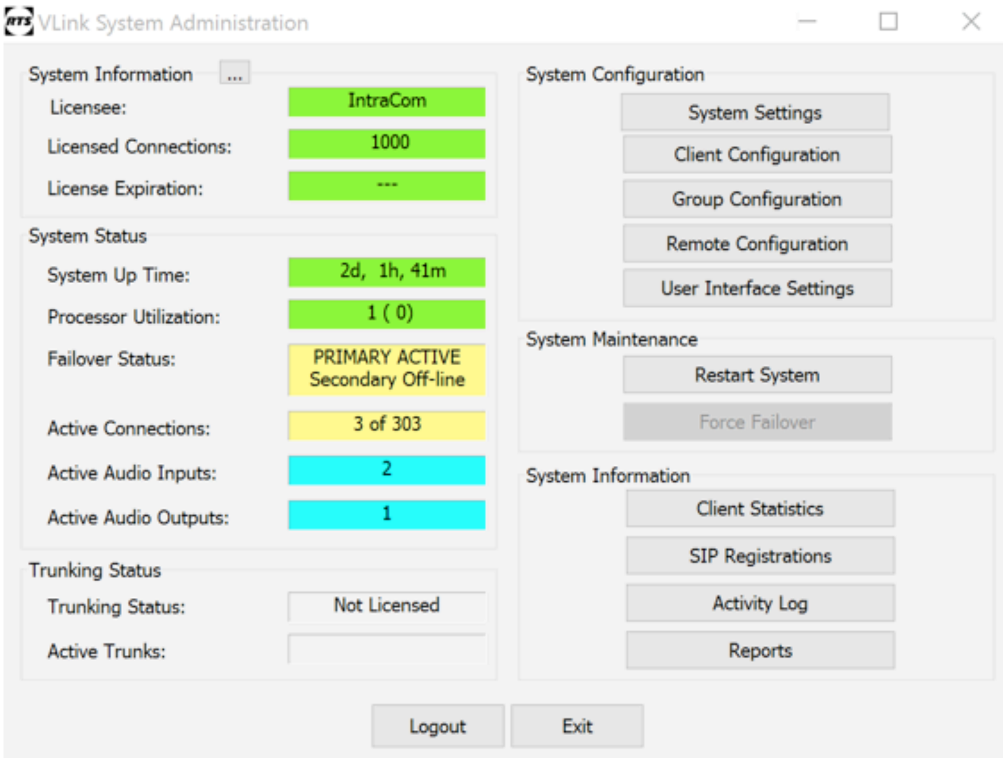
Open your VLink System Administration application. At the Login screen enter your "Login Name" (default: "admin") and "Login Password" (default: no password)



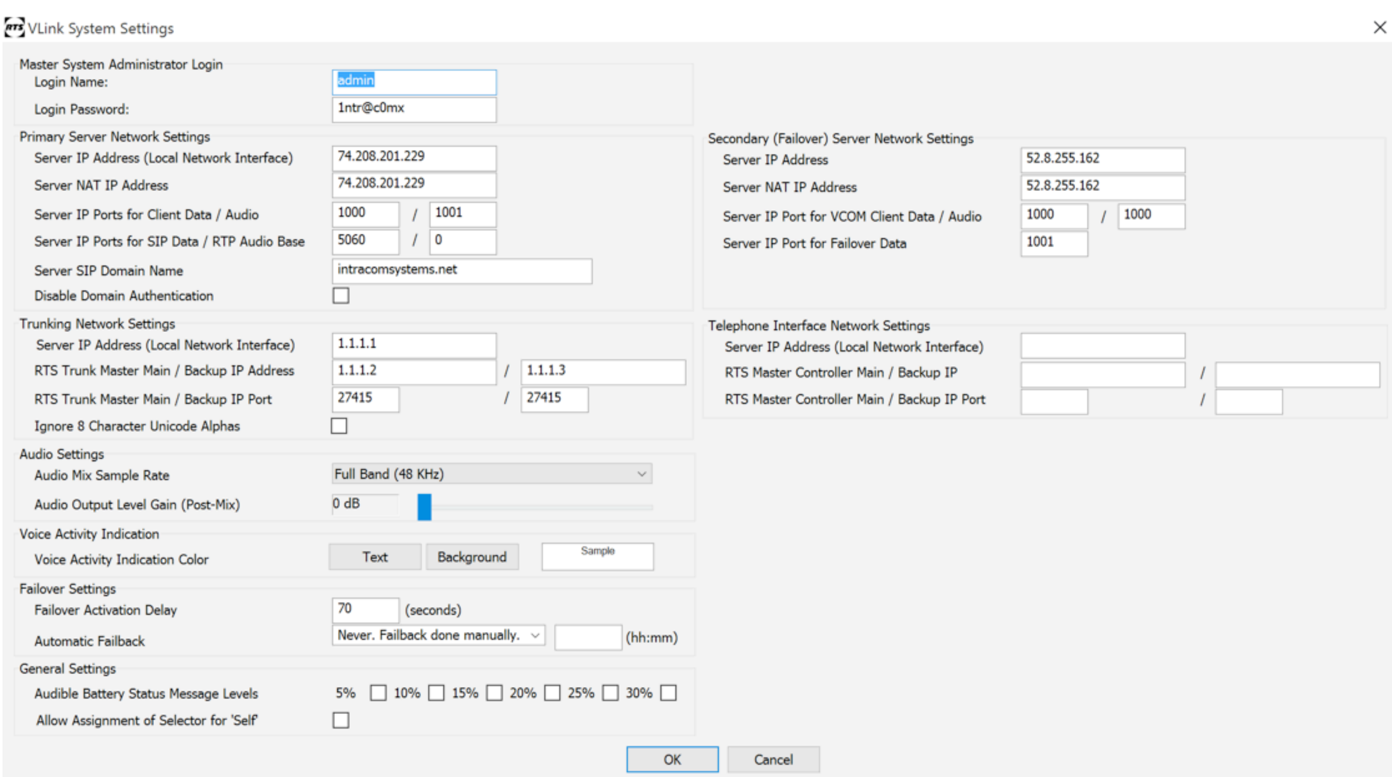
If this is your first log-in click "Configure…". Enter the IP address of your VLINK Server immeditaley followed by :1000 (no spaces). For detailed instructions on using the VLink System Administration application, and confirguring your VLink system in general, refer to the VLink System Administration Guide: http://www.rtsintercoms.com/us/rts/file?i=98673&lg=eng



Next, set up the main system settings for SIP and the connection to the RTS ADAM intercom. Click on the “System Settings” button from the main System Administration window.



Fill in the following fields:
“Sever IP Ports for SIP Data / RTP Audio Base” should be left at default or filled in as 5060 / 0 under most circumstances.

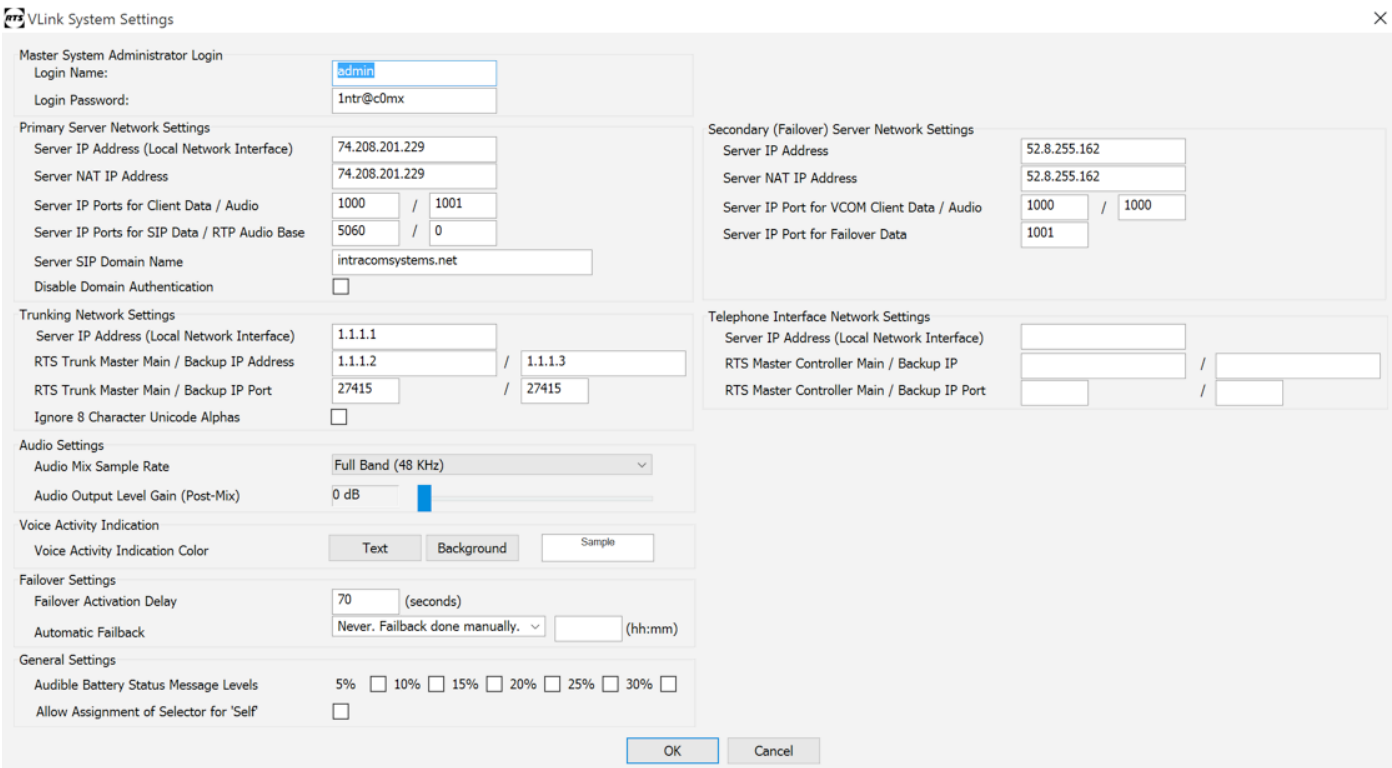


RTS Telephone Interface Network Settings

"Server IP address (Local Network Interface)" should be the IP address of this VLINK Server under most circumstances.

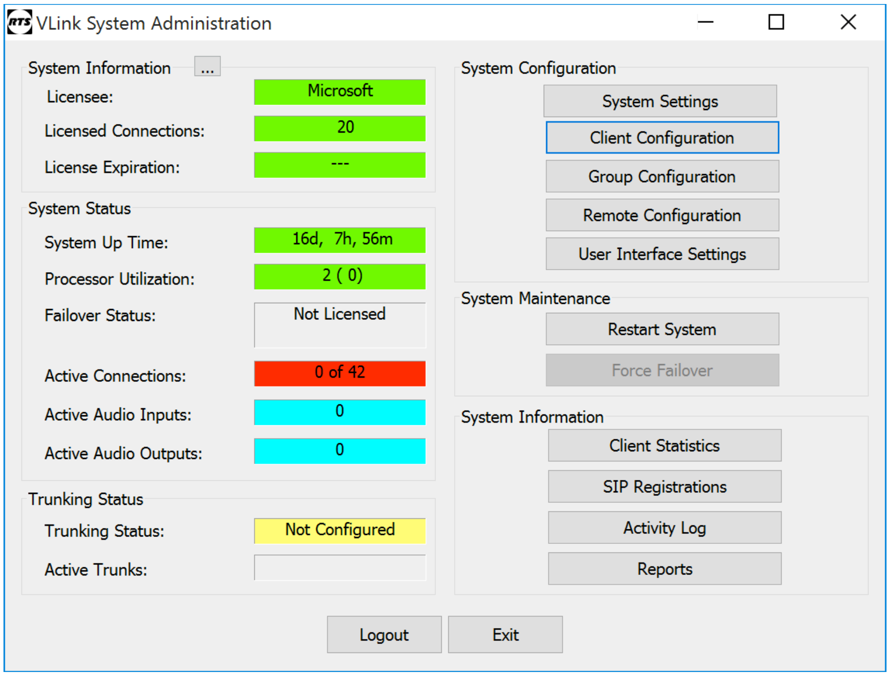
"RTS Master Controller Main / Backup IP" should be the IP address of the ADAM main master controller followed by the IP address of the ADAM backup master controller.

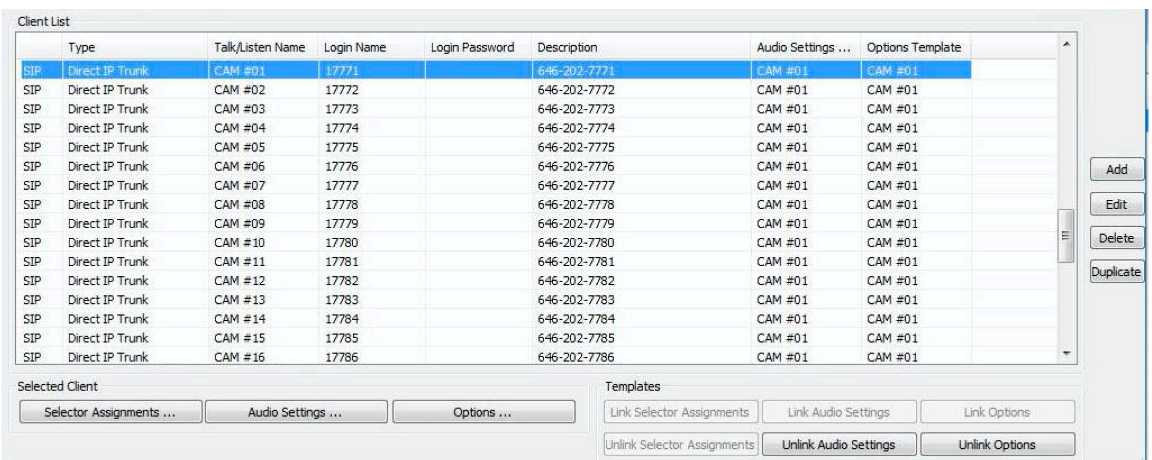
"RTS Master Controller Main / Backup IP Port" should be 27415 / 27415 under most circumstances.



Adding a SIP Phone Line into the Server

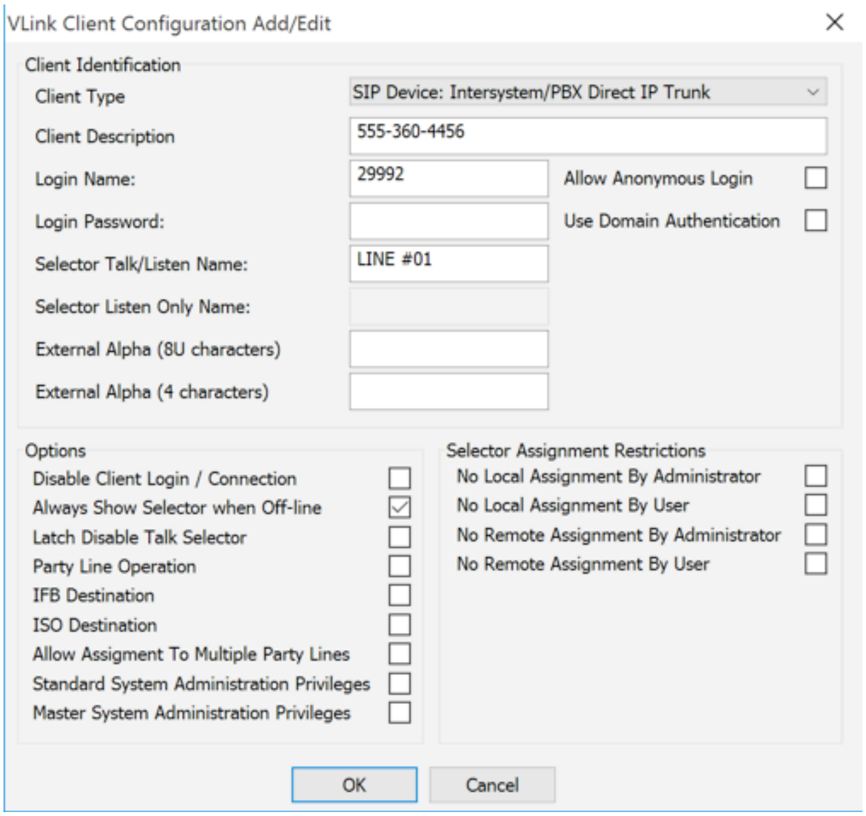
To add a SIP Phone line connection from your phone switch or service provider to the Server click “Client Configuration” from the main System Adminisration Window. From the Client List window click “Add”.



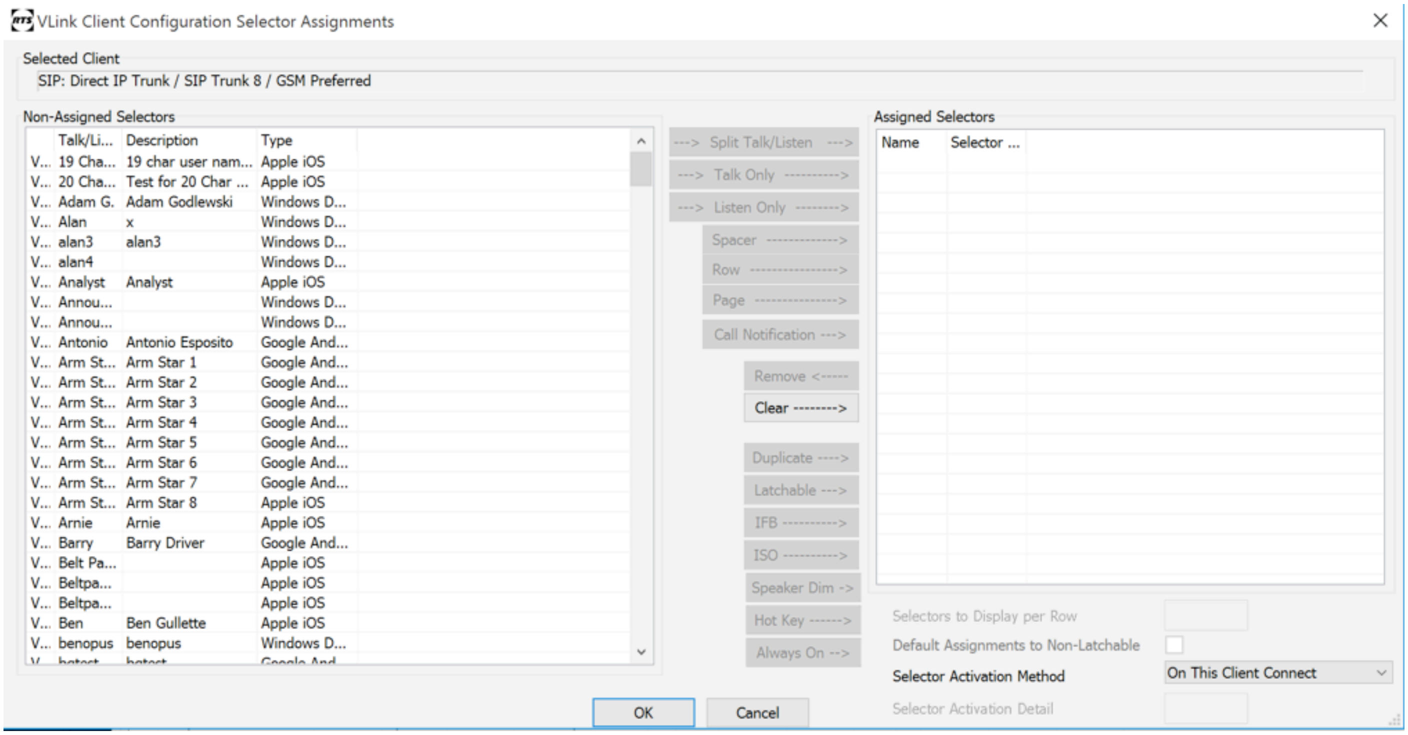


From the Client Configuration Add/Edit window, under “Client Type”, select “SIP Device: Intersystem/PBX Direct IP Trunk”. **In the “Client Description” section enter the phone number.**
Input a “Login Name” and a “Login Password”.
Important Note: The Login name and password should be that provided by your SIP phone line provider for that particular line.
Also, Input a “Selector Talk/Listen Name, which will appear on a VLink Control Panel selector, and on the optional Pacific Interactive TIF Tally Display.
Click “OK”.

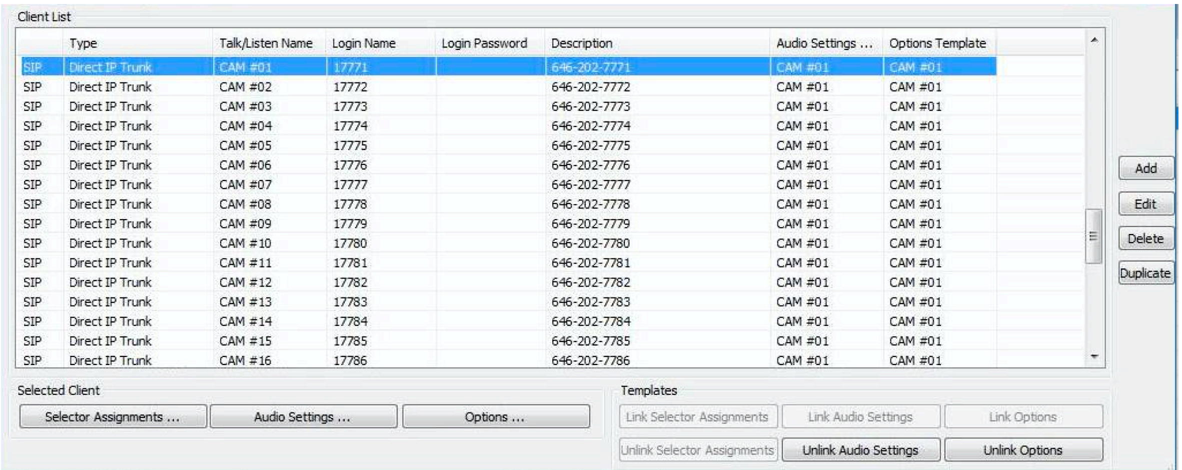
Important note: The "Client Description" field will be the one that is used and displayed within AZEDIT to link the VLINK Server SIP line and the ADAM intercom ports, so it is highly recommended that the actual phone number is used for the Client Description.
The Selector Talk/listen Name will be the one that is displayed on the optional Pacific Interactive Status Display System, so it is highly recommended if possible to use the last 4-digits of the phone number for the Selector Talk/Listen name.



Next, from the Client List Window highlight this client and click the Selector Assignment Button at the bottom. Click this same client from the left window, click the >Listen Only Button and next to Selector Activation Method in the lower right, click the pulldown and select “On This Client Connect”.
Click “OK”.



Make sure your added entry in the Client List window is still highlighted and click the "OPTIONS" button at the bottom:



Setting Phone Line Control Methods for each Phone Line

There are Serveral user defined options that can be chosen for each phone line.

- Inbound Session Activation allows you to choose the method used to answer an incoming call to the Server.
- Outbound Session Activation allows you to choose the method used to go "off-hook" on a phone line and dial an outbound call.

The choices for these two fields are:
Disabled – For Inbound: Disables any incoming calls; For Outbound: Disables any outgoing calls - for this phone line in / out of the Server
On Call Receive (Auto Answer) – Auto Answers an incoming call into the Server for this phone line*
NOTE: If this option is chosen, please fill in the field ***"Auto-Answer time in ms"*** to determine the length of time after ringing begins for the line to be auto-answered by the Server (1000 ms = 1 second)
On Talk Selector Activation – Requires an RTS intercom panel with this phone line's key to manually answer the call.

- Inbound Session deactivation allows you to choose the method used to hang up a call which is made into the Server.
- Outbound Session Deactivation allows you to choose the method used to hang up a call which is originally dialed outbound from an RTS panel.

The choices for these two fields are:
Disabled – For Inbound: Once answered, calls will not be hung up by the Server; For Outbound: For calls originated via an RTS intercom panel, calls will not be hung up by the Server for this phone line in/out of the Server.
On Forced Disconnect – Requires a manual Hang-Up for this phone line to be performed from an RTS panel
On Talk Selector Deactivation – Will hang up the line when an RTS panel talk key for this phone line is turned off.

Configurable Phone Line Sounds

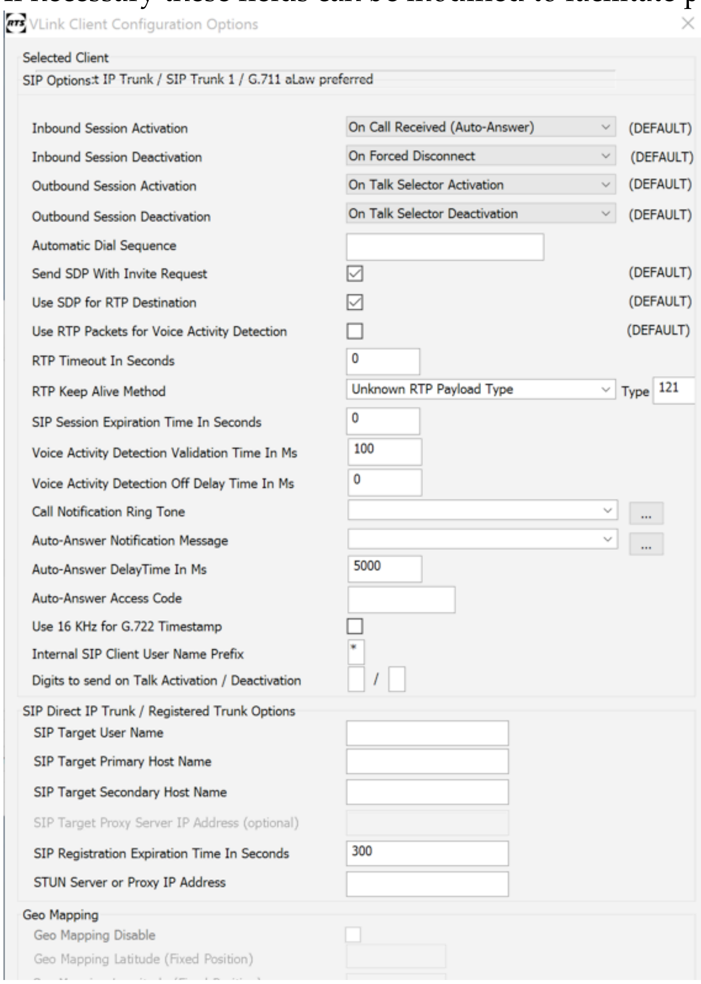
There are two system or user provided types of configurable phone line sounds which are in a .wav sound file format*.
Call Notification Ringtone – This .wav file is played out and heard on the intercom system whenever a phone line is ringing
Auto-Answer Notification Sound – This .wav file is played out and heard by the caller whenever a phone call is answered. This sound is heard whether a call is auto-answered or manually answered by an RTS intercom panel.
* Customers can add their own .wav files which will then appear in the pull down lists. The files must be .wav and placed in the directory Program Files/Intracom/Virtual Matrix/Sounds.

SIP Target Host

In the SIP Target Host field enter the IP address or domain name for the phone switch or SIP line service provider. This field is critical for proper operation.

Other Server Settings for SIP Lines

Initially, choose the default values for all other fields on the Options Page for each SIP Phone line.
If necessary these fields can be modified to facilitate proper connection to some phone systems.



After adding a SIP line via the EDIT tab, and completing the Selector Assignment and Options Setup, go thru the same setup procedure for each SIP line connecting to the system,

4. VLINK SERVER 4-WIRE DANTE INTERFACE TO OMNEO

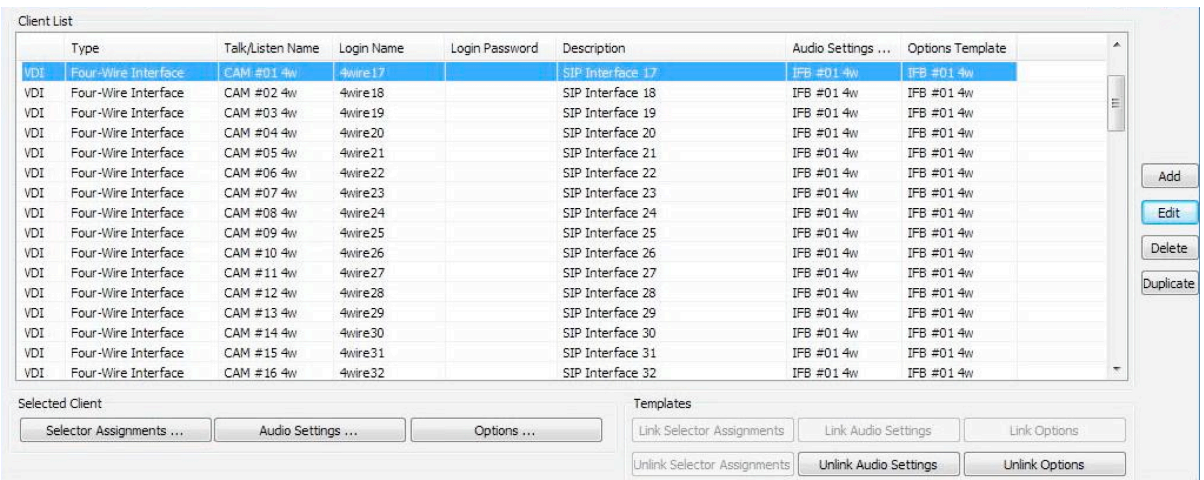
4-Wire Interface Setup

Next, the VLINK 4-wire Interfaces which will connect each phone line to the ADAM intercom will be setup.

To connect the phone line thru the VLINK Server and provide audio to the RTS ADAM intercom, your will first create a 4-wire interface for each phone line's audio which will create the internal audio connections between the Server and PC audio channels (in this document using Dante Virtual Soundcard* software). There will be one separate 4-wire for each phone line that needs to be connected to the RTS intercom.

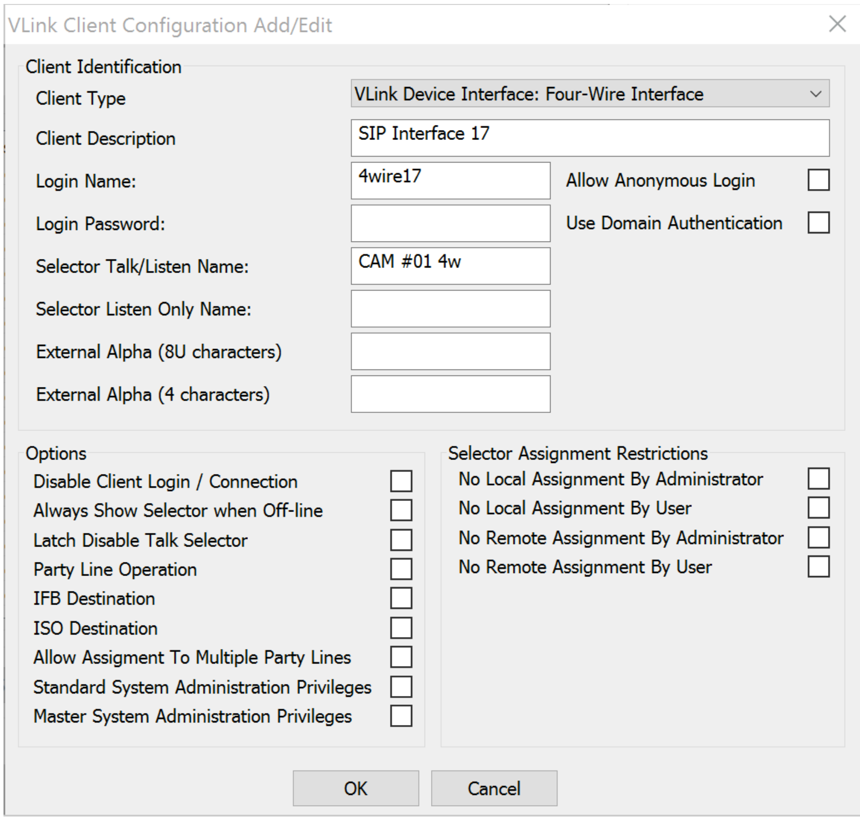
* Note other alternative PC audio interfaces such as hardware sound card(s) are also possible. Contact your RTS support engineer if other alternatives are desired.

To add a 4-wire interface click “Client Configuration” from the main System Adminisration Window. From the Client Configuration window click “Add”.



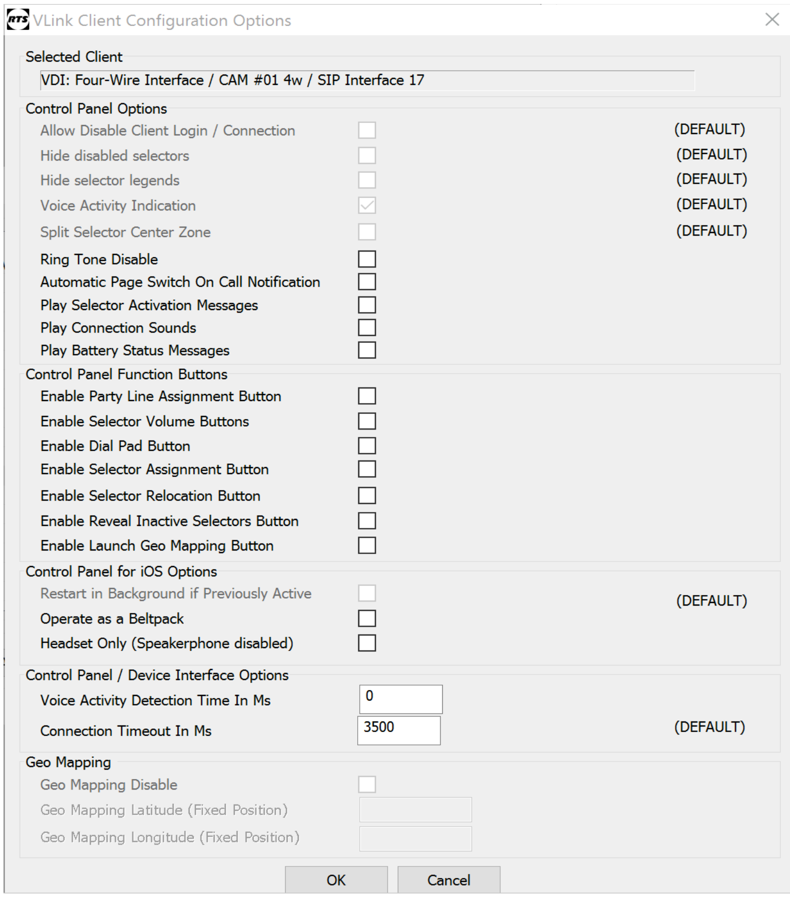
From the Client Configuration Add/Edit window, under “Client Type”, select “VLink Device Interface: Four-Wire Interface”. Input a “Client Description”, “Login Name”, “Login Password” (can be left blank), and a “Selector Talk/Listen Name”.

The login name and password for each audio line interface can be any name of your choosing. To simplify setup it is recommended that the login names and talk/listen names for these 4-wires are designated as 4W#1, 4W#2 etc. thru the number of phone line connections in the system. The password can be left blank. The Client Description can be any name of your choosing. For clarity we recommend using “4-Wire Interface xx”, where xx is the numerical count for the associated phone line in the system.

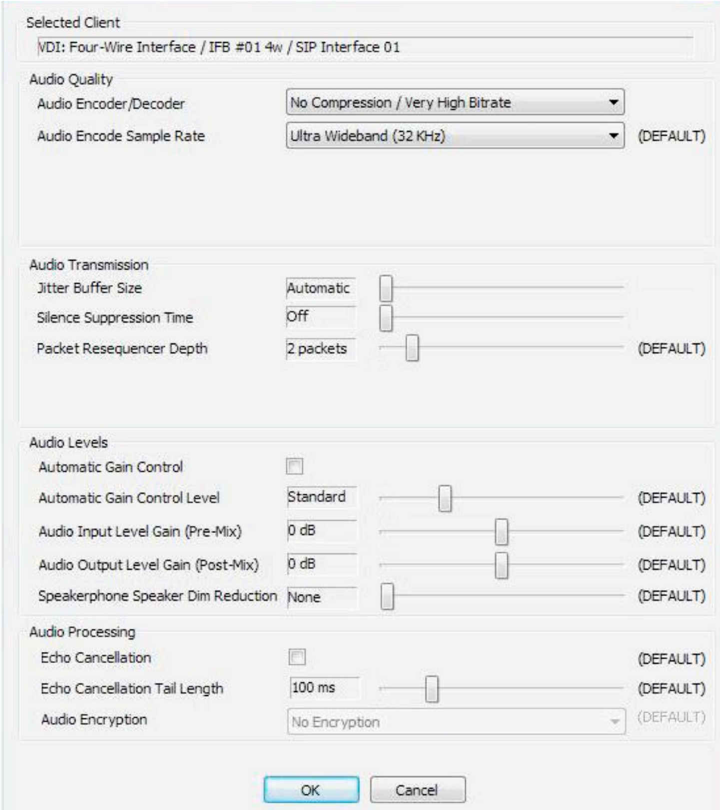


Click “OK”. Find your newly added entry in the Client List window, click to highlight it, and click “Options…”

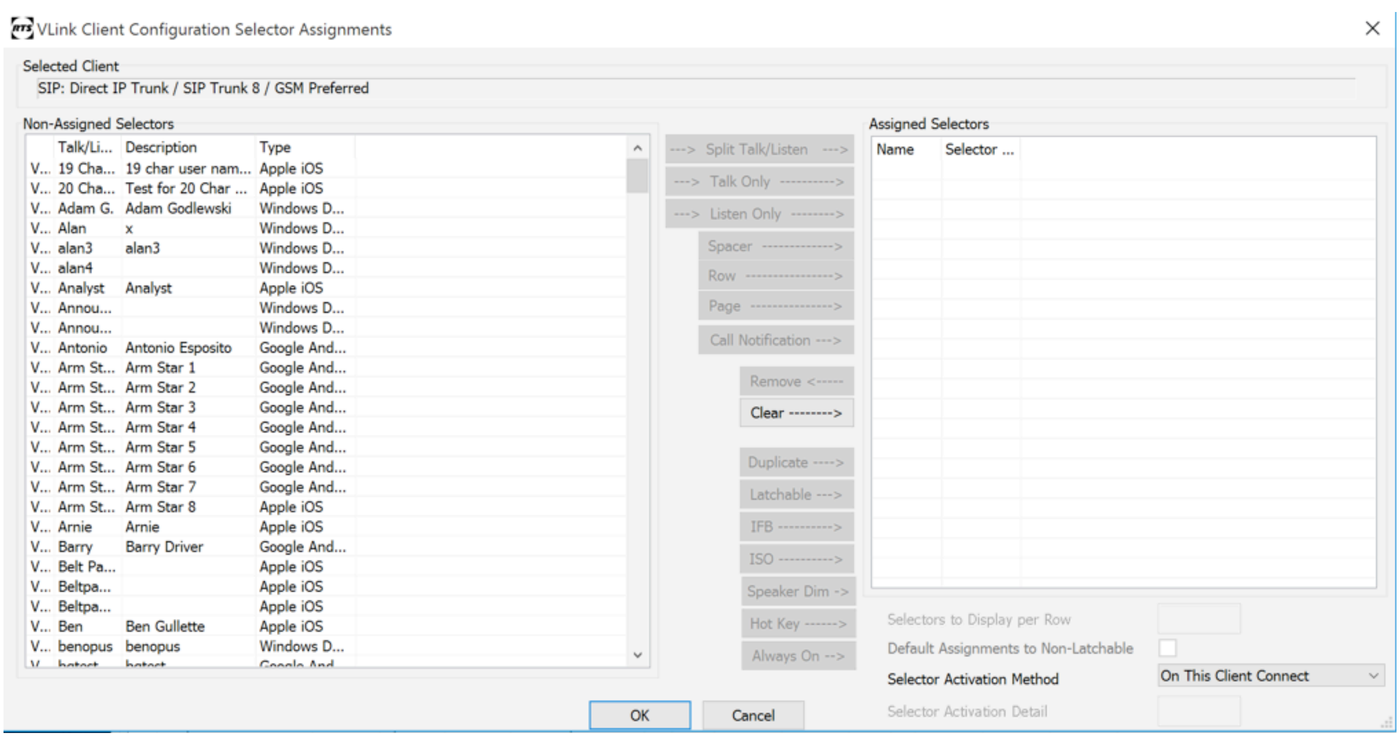
In the Client Configuration Options window, set “Voice Activity Detection Time In Ms to “0” and click “OK”.



Make sure your added entry in the Client List window is still highlighted and click "Audio Settings…" Check that the settings match the screen below and click "OK".



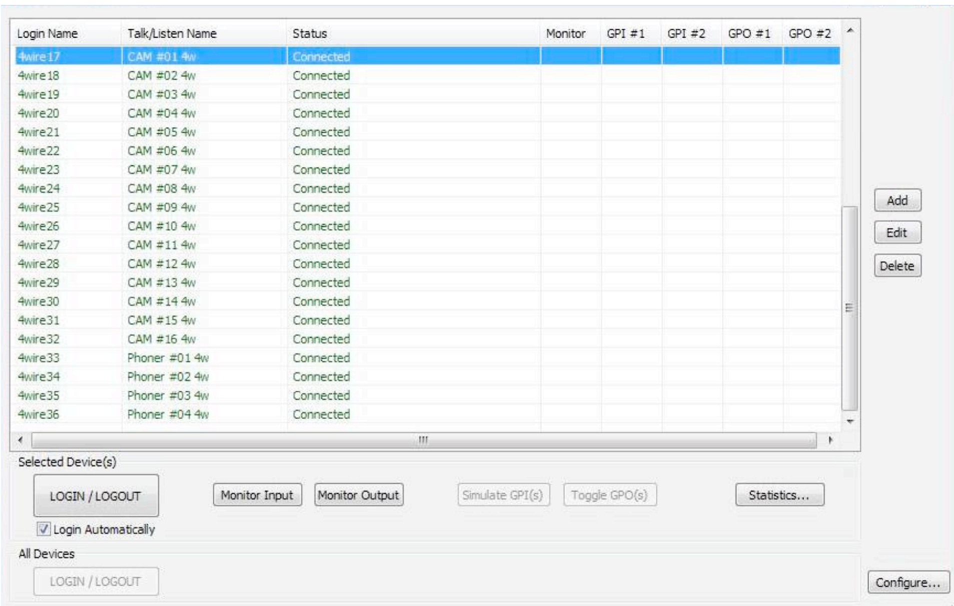
Make sure your added entry in the Client List window is still highlighted and click “Selector Assignments…”
Choose this same 4-wire interface from the list on the left, and click the >Listen Only button.
On the bottom right, next to Selector Activation Method, click the pull down and select “On This Client Connect”. Click OK.



Perform all of the above steps again and add a 4-wire Interface for each phone line.

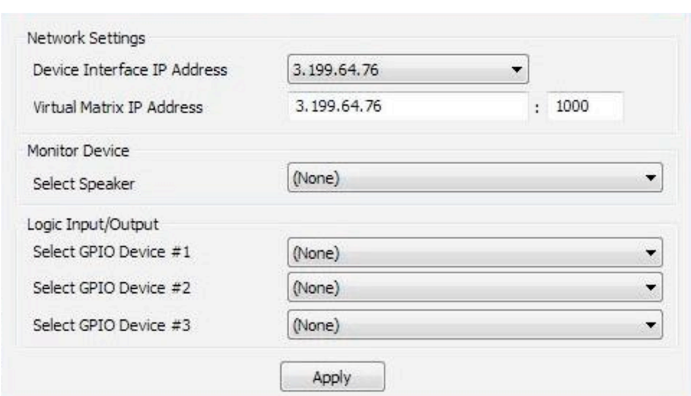
Internal Server Dante Virtual Soundcard Channel Connections

After adding one 4-wire for each phone line in the system, open the VLINK Device Interface Application. This application would have been installed as part of the original VLINK Server install. If needed for detailed instructions on how to install and use the VLINK Device Interface application refer to the VLINK Device Interface application User Guide which can be found here:
http://www.rtsintercoms.com/us/rts/product/RTS_Vlink_Device_Interface/1502?fam%5B%5D=162



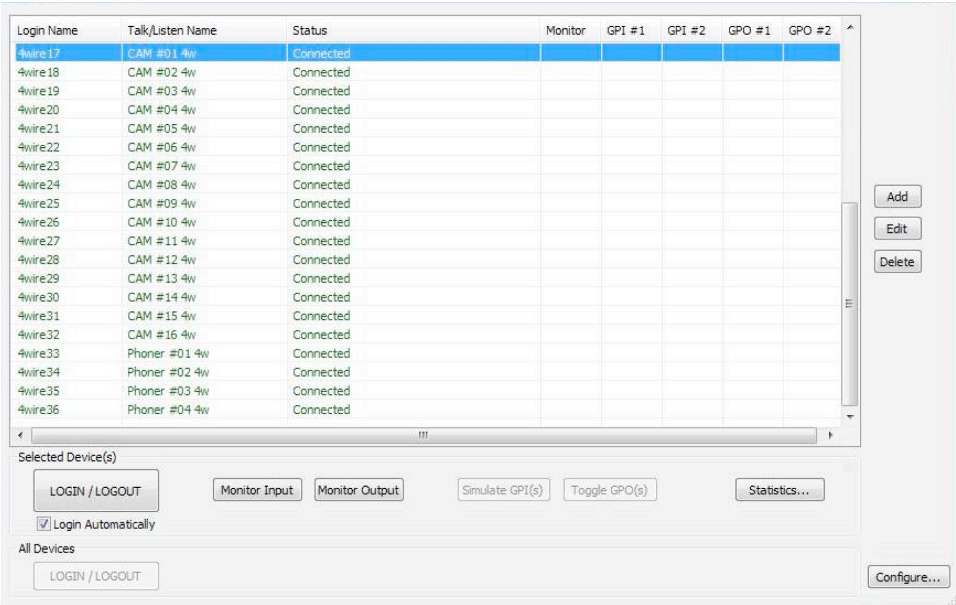
The internal audio connections between each phone line's 4-wire interface and the Dante Virtual Soundcard audio channels will now be done.

From the main window select "Configure…"



Input your VLINK Server IP address in the Virtual Matrix IP Address field and click "Apply".

Next, from the main window click “Add”.

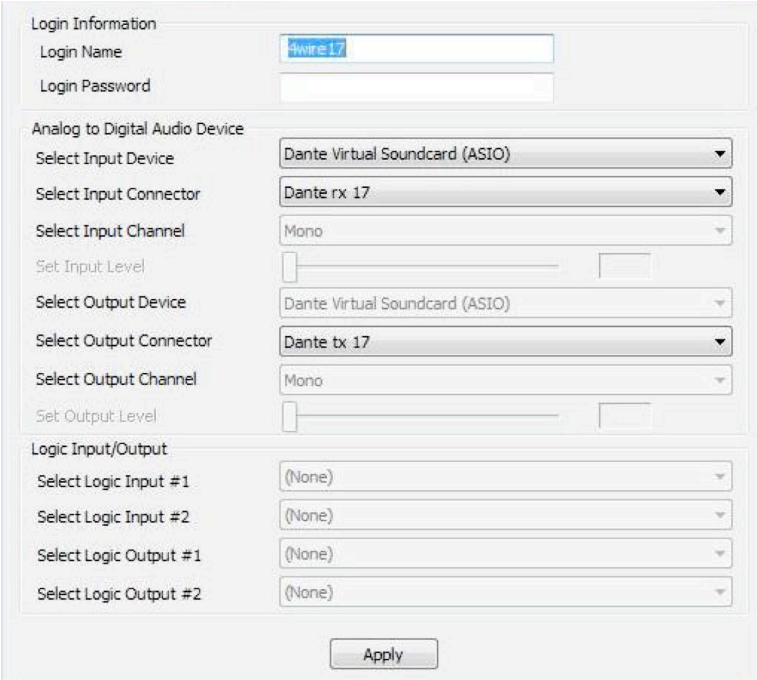


Fil in **IDENTICAL** “Login Name” and “Login Password” that had just been used when setting up each 4-wire interface connection. It is critical that the exact same login names and passords are used (these are case sensitive) as were previously configured for the 4-wire interface connections.

Then Under “Select Input Device”, select “Dante Virtual Soundcard (ASIO). Under “Select Input Connector”, select “Dante rx xx, where xx is the sequential phone line number & 4-wire interface number. For example, if the first phone line, which will be associated with 4-wire 4W#1 is being added, under “Select Input Connector” use the pulldown to select Dante rx 01.

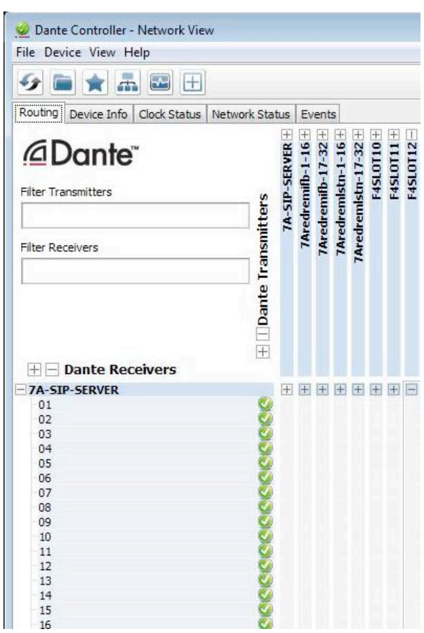
Make a similar selection for “Select Output Connector” select. In this example use the pulldown to select “Dante tx 01”. Click “Apply”.

Add one Device Interface entry for each phone line / 4-wire interface.

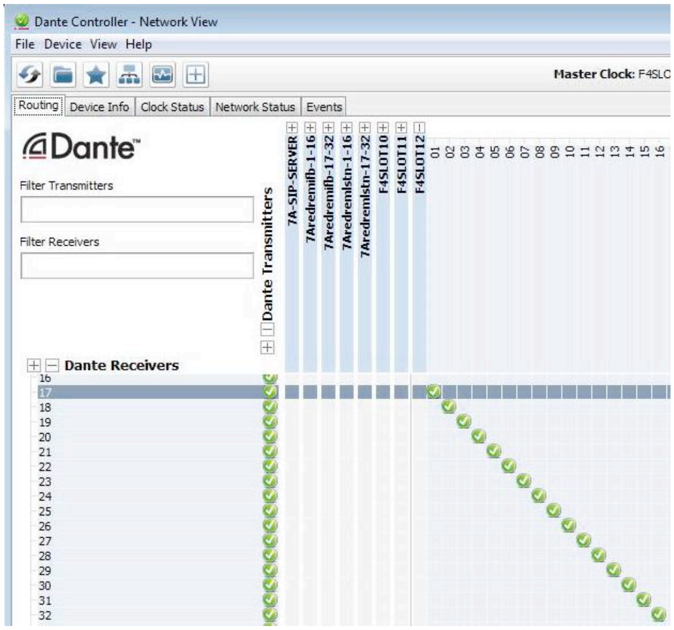


5. SETUP OF AUDIO CHANNELS BETWEEN VLINK SERVER AND OMNEO CARD(s)

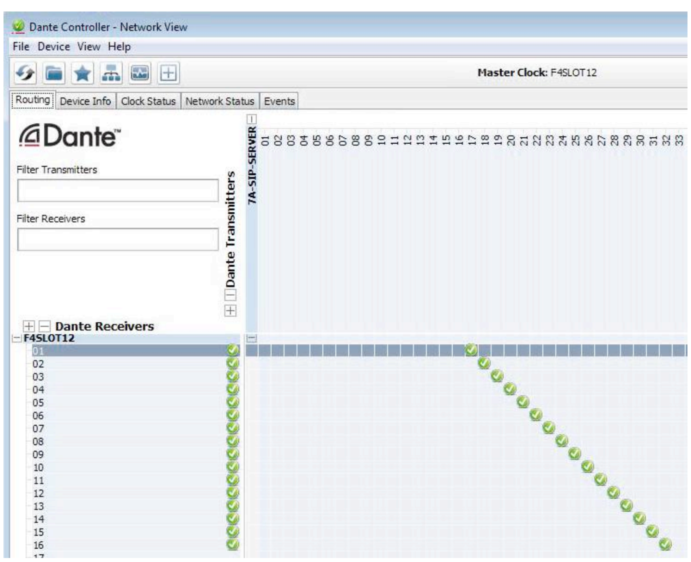
Once the SIP phone line has been configured to connect to the VLINK Server, and internally the Server has been configured to send audio from an installed Dante Virtual Soundcard, the next step in configuration is to map audio channels from the VLINK Server's virtual Soundcard to the associated ADAM Omneo channels. Note it is important to pre-design which phone lines will connect to which intercom port, as this mapping relationship will also need to be done later for phone line control.
The screen shots below show a one-for-one mapping of phone lines in the Server to intercom ports via Omneo in the intercom using the Dante Controller software.



Click the + next to the VLINK Server PC name on the left.
Click the + next to the RTS Omneo Card name on the top.
Click at the crosshairs to create an audio connection (indicated by a green circle) in a diagonal so each VLINK Server channel is mapped in one direction to the associated Omneo card channel.
Click the – next to the VLINK Sever PC name on the left.
Click the – next to the RTS Omeno Card name at the top.



Next, click the + next to the RTS Omneo Card name on the left.
Click the + next to the VLINK Server PC name on the top.
Click at the crosshairs to create an audio connection (indicated by a green circle) in a diagonal so each Omneo card channel channel is mapped in the other direction to the associated VLINK Server channel.
Click the – next to the VLINK Sever PC name on the left.
Click the – next to the RTS Omeno Card name at the top.

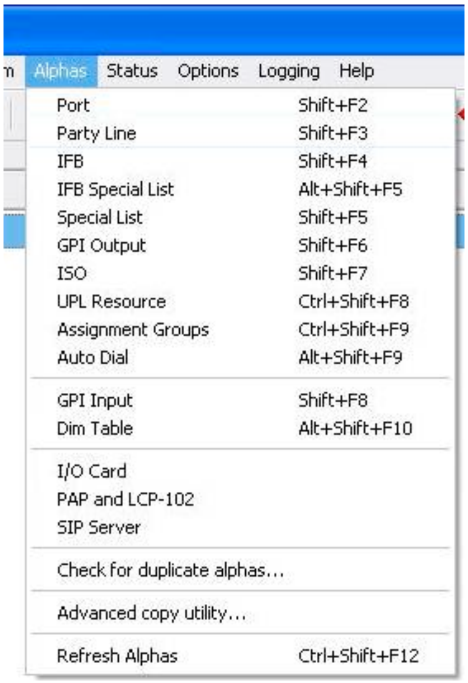


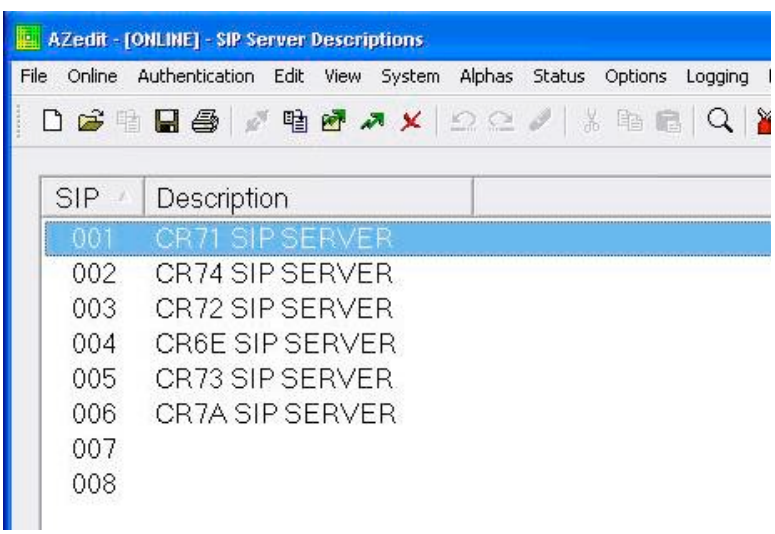
Exit the Dante Controller software program.

6. ADAM/AZEDIT VLINK SERVER SETUP

To connect a VLINK Server to be used for SIP phone line connection to an ADAM intercom, follow the steps below:

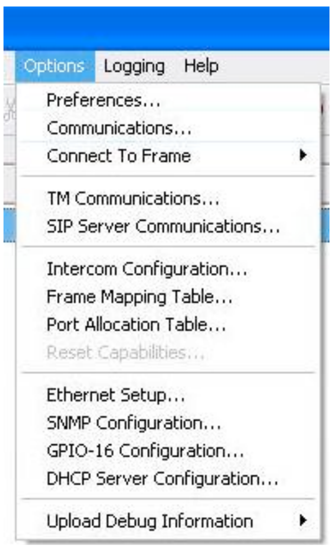
The RTS ADAM intercoms support up to eight redundant and separate SIP Servers. In AZEDIT, name the Server you are currently configuring. Click Alphas / SIP Server. Double Click on the SIP Server being configured.

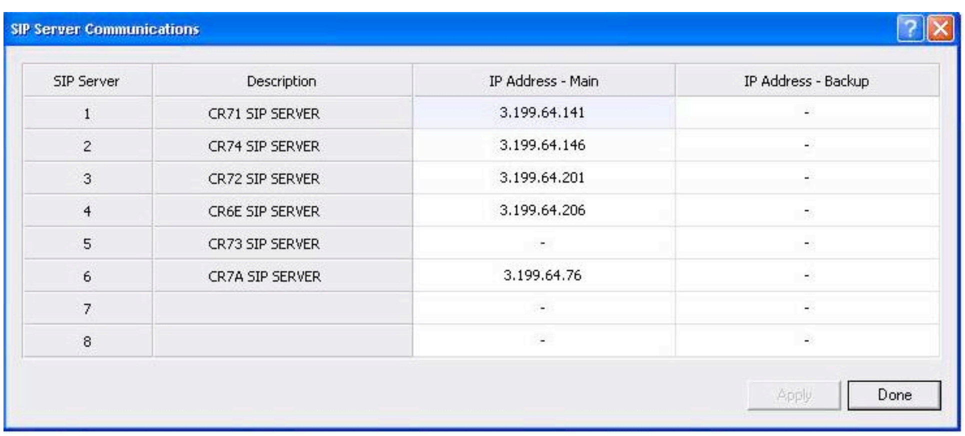




Enter the SIP Server Description in the EDIT box and click DONE.

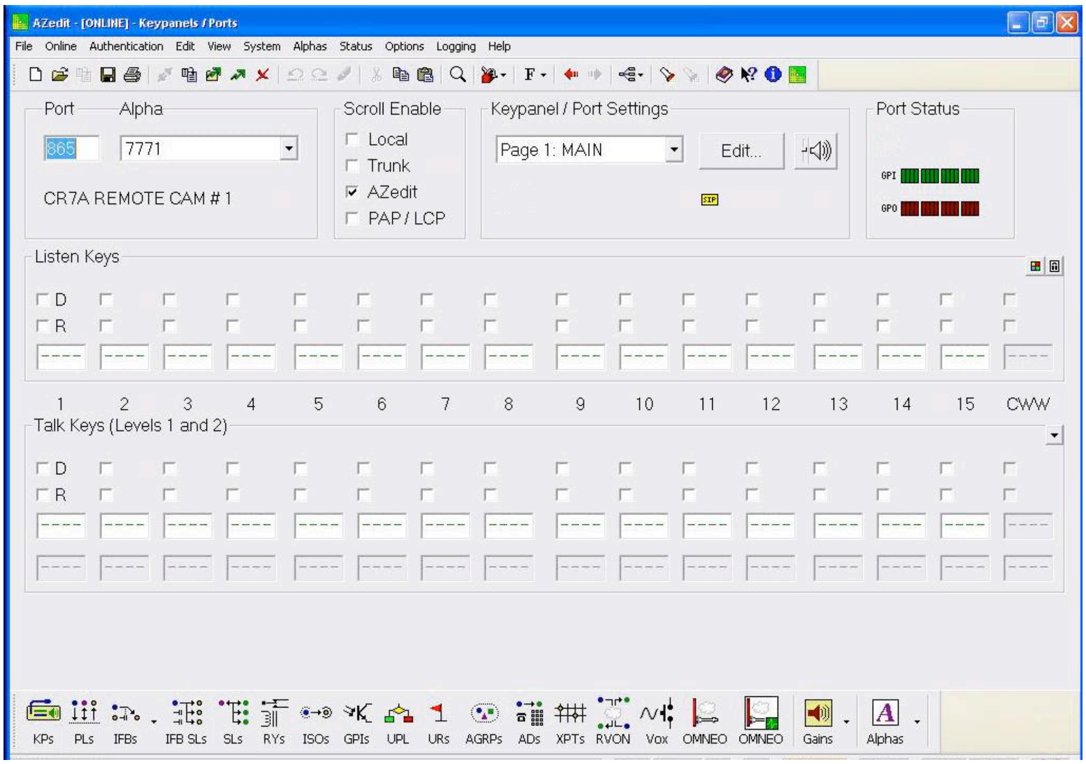
Under Options/SIP Server communications, enter the IP address of the SIP Server (and redundant Server if applicable) as shown below.



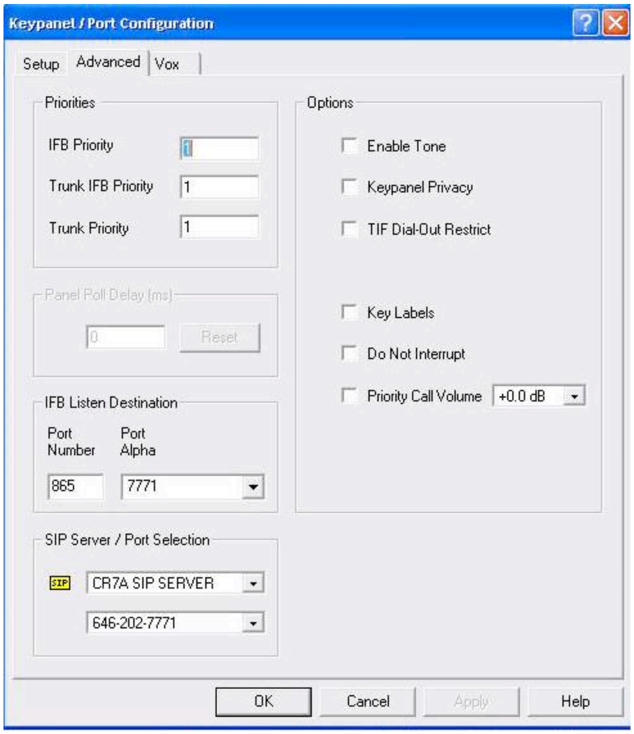


7. INTERCOM PORT TO SIP LINE ASSOCIATION

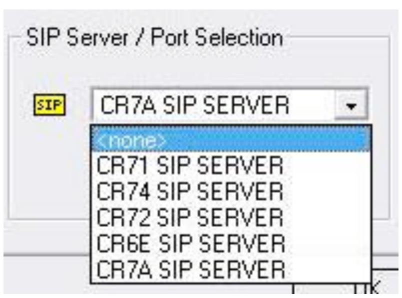
Next you must tell the intercom which intercom port is connected via audio to which VLINK device interface channel / SIP Phone Line. This must be consistent with how you have designed the phone line connection to the Server 4-wire, and the internal Dante Virtual Soundcard channel to the Omneo intercom port channel.
When completed the intercom port keypanel setup page for each phone line's intercom port will have a yellow "SIP" indicator box under the Edit button.
From the keypanel setup screen for the first SIP phone line port, Click the EDIT button.



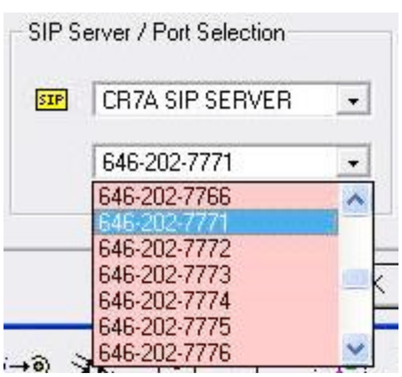
Click the Advanced Tab:



Under the SIP / Server / Port Selection, Click the first pulldown and select the VLINK Server for the phone lines being configured



Click the second pulldown and select the VLINK phone line to be associated with this RTS intercom port.



Click “OK”.

Do this for each intercom port that has been connected to each of the VLINK Server SIP phone lines.

8: TESTING THE COMPLETED SYSTEM AFTER SETUP

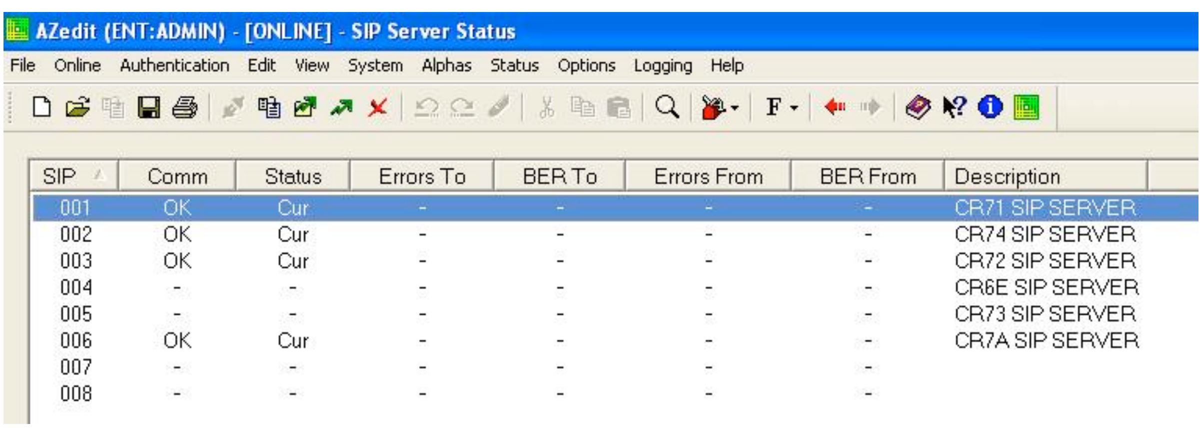
To test the completed system, first make sure all systems are turned on and connected to the IP network. Make sure the VLINK Device Interface Application is running on the VLINK Server. Additionally, make sure the Audinate Dante Virtual Soundcard Application is running (NOT the Dante Controller Program, which does not need to be running).

Begin testing by opening a command (DOS) prompt on the VLINK Server PC. One-by-one ping the IP addresses for the following and make sure a reply is received: IP address of phone switch or SIP service provider network, optional TIF Tally Display PC, RTS ADAM Primary master controller card, RTS ADAM back-up master controller card, RTS Dante Omneo card. If any of the devices are not pingable, check the IP network configurations and connections.

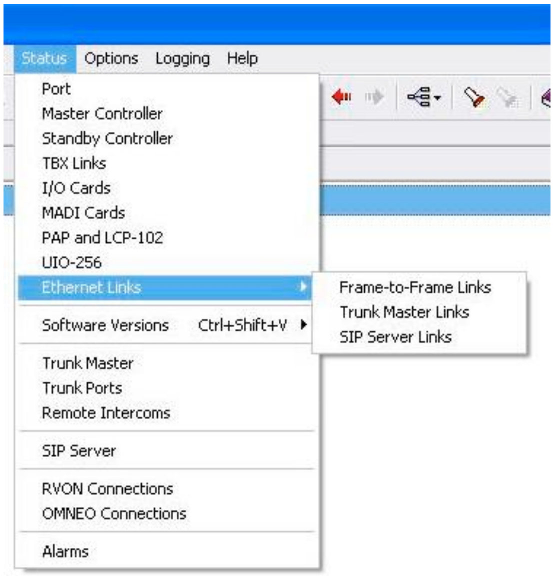
On the VLINK Server PC, using the System Administration program, click client statistics. Make sure the SIP line clients are shown as online. If not, check the configured SIP Provider / Phone switch settings and SIP login information for each line.

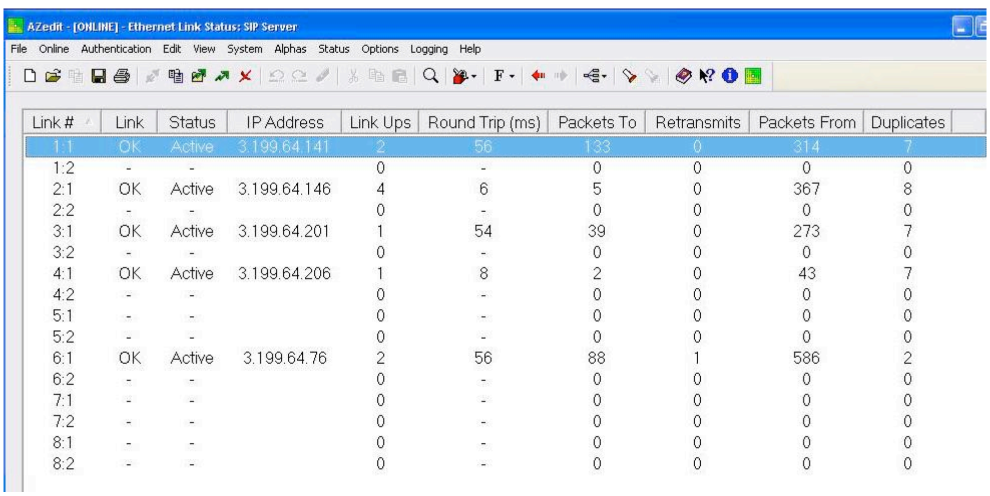
Also on the VLINK Server PC, maximize the Device Interface program which is running. Make sure all of the device interfaces are shown as Connected. If not, click configure and check the IP address in both fields is the IP address of the VLINK Server PC. If necessary, click EDIT for each of the device interfaces and check the Dante Virtual Soundcard channel mapping.
WARNING: NEVER EXIT OR CLOSE THE VLINK DEVICE INTERFACE PROGRAM. Always just minimize it when you are done looking at status.

Check the SIP Server status in AZEDIT. When the VLINK SIP SERVER is correctly connected for SIP Line Control, you can see the status under Status/SIP Server as shown below:



You can see more detailed status by going to Status/Ethernet Links/SIP Server Links:





Link #	Link	Status	IP Address	Link Ups	Round Trip (ms)	Packets To	Retransmits	Packets From	Duplicates
1:1	OK	Active	3.199.64.141	2	56	133	0	314	7
1:2	-	-		0	-	0	0	0	0
2:1	OK	Active	3.199.64.146	4	6	5	0	367	8
2:2	-	-		0	-	0	0	0	0
3:1	OK	Active	3.199.64.201	1	54	39	0	273	7
3:2	-	-		0	-	0	0	0	0
4:1	OK	Active	3.199.64.206	1	8	2	0	43	7
4:2	-	-		0	-	0	0	0	0
5:1	-	-		0	-	0	0	0	0
5:2	-	-		0	-	0	0	0	0
6:1	OK	Active	3.199.64.76	2	56	88	1	586	2
6:2	-	-		0	-	0	0	0	0
7:1	-	-		0	-	0	0	0	0
7:2	-	-		0	-	0	0	0	0
8:1	-	-		0	-	0	0	0	0
8:2	-	-		0	-	0	0	0	0

If status is not shown correctly, check the IPaddress listed for the VLINK SIP Server and check the network connection between the VLINK Server and the ADAM Master Controller again.

Place key assignments on any RTS panel for the ports used for the SIP phone lines.
Dial into each line from an external phone and check the functionality of auto-answer, manual answer by keypanel, and audio in both directions.

Further operational information concerning RTS ADAM control of telephone line interfaces, including those which come from a VLINK Server is detailed below.

The VLINK SIP phone lines are operated from the intercom keypanels, and from the dial pad on the telephone at the remote end of the line. Any keypanel with a keypad may use the VLINK Server SIP phone lines. All that is necessary is to program a key to talk to the VLINK Server SIP phone line, as if it were a keypanel. The alpha numeric display or tally LED for that key then provides information about the phone line. A solid display or non-illuminated LED indicates a line which is not in use. A slow flash indicates a line which is in use (off-hook). A rapidly flashing display or LED indicates a line which is ringing. In addition, the alpha numeric display shows digits as they are dialed, and the LED flashes for each digit.

NOTE: Displayed tallies are different if the *Don't Generate Tallies for TIF or Trunk Use* option has been selected in

Options|Intercom Configurations|Options.

To use a VLINK Server SIP phone line, either to answer a call, or to call out, you first need to program a key to talk to the VLINK Server SIP phone line. This is accomplished in the same manner as programming a key to talk to a keypanel. To program a key by port number, enter
NUM-nnn_PGM-t, where NUM is the number 1 key, nnn is the port number of the TIF-4000 you want to use, and t is any talk key. You also need to use the listen key, so it should be assigned as either AF (auto-follow) or AL (auto-listen).

> **NOTE:** The VLINK Server SIP phone lines only respond to commands which are sent via a point-to-point key assignment. If you wish to use the VLINK Server SIP phone line primarily on a PL, you must add a point-to-point assignment as the L2 talk assignment on the talk keys which are going to either answer the line, or dial out on the line.

Dialing a Call

To **dial a call on a VLINK Server SIP phone line using a or KP-32**, do the following:

1. Turn **on the listen key** for the line you wish to dial on.

This allows you to hear dial tone, and your DTMF dialing tones.

2. Enter dial mode by entering **PHONE-PGM-T**.

PHONE is the 4 button on the keypad. PGM is on the keypad, and T is the talk key which is programmed to talk to the TIF-4000 you are dialing on. Leave the talk key in the latched position as you dial the number.

3. Dial the **number**.

As you enter each digit, it appears in the alpha display above the key you are dialing on. If the listen key is latched, you hear each DTMF tone as it is generated.

4. When you have completed dialing, momentarily turn **off** the talk key to exit dial mode.

The alpha numeric display reverts to normal, and you may use the key and keypad in the normal manner.

NOTE: Digits 0-9 generate the DTMF digits 0-9. PGM generates the #, and CLR generate * (# and * are displayed
for these keys).

It is necessary to press CLR twice if you wish to generate an *, as a single CLR is used to trigger the speed dial and redial features.

To **dial a call on a VLINK Server SIP phone line using a KP-12 or KP-32**, do the following:
1. Tap the **phone key** to begin your call.
This places the keypanel in dial mode: the CALL indicator turns on, and the MAN DIAL (manual dial) displays in the call waiting window. You should also hear the dial tone.
NOTE: You can hang up the phone line at this time by simply tapping the phone key again.
2. Tap **SEL** (select) to select MAN DIAL.
The twelve intercom keys can now be used to dial a telephone number. Each key corresponds to the number printed next to it on the front of the panel. If the keypanel has alphanumeric displays, the key numbers are displayed above each key.
3. Begin **dialing** the number by tapping the appropriate keys.
After you dial the first digit, END DIAL appears in the call waiting window.
4. When you have completed the dialing, tap **SEL** to select END DIAL.
This returns the keypanel to normal operating mode. If the called party answers, proceed with your conversation.

Hanging Up

The VLINK Server SIP line control detects the call at the far end has hung up under most circumstances. It detects the hang up by either loop interrupt, battery reversal, or the presence of a dial tone or busy signal. Some telephone systems do not provide any of the above, so it is necessary to force a hang up. In addition, if the call was placed to an auto-answer device, it is necessary to force a hang up when the call is complete.
> Enter **PHONE-CLR-t**, where PHONE is the 4 on the keypad, CLR is the CLR button, and t is the talk key which is programmed to talk to the VLINK Server SIP phone line which you want to hang up. This disconnects the line for which you struck the talk key.
NOTE: If talk is in the on position, you must turn off the key, then momentarily turn it on again to indicate which line you wish to disconnect. If the line is in dialing mode, then you must first exit dialing mode by turning off the key, then use PHONE-CLR-t to hang up.