Innovating the Future of Global Communications

VLINK SIP Setup Guide

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Warranty and Service Information

For warranty and service information, refer to the appropriate web site below:

RTS Intercoms .... www.rtsintercoms.com/warranty
- RTS Digital
- RTSTW
- AudioCom
- RadioCom
- Intercom Headsets

Customer Support

Technical questions should be directed to:

Customer Service Department
Bosch Security Systems, Inc.
www.telex.com

Technical Questions EMEA

Bosch Security Systems Technical Support EMEA
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THE LIGHTNING FLASH AND ARROWHEAD WITHIN THE TRIANGLE IS A WARNING SIGN ALERTING YOU OF “DANGEROUS VOLTAGE” INSIDE THE PRODUCT.	CAUTION RISK OF ELECTRIC SHOCK DO NOT OPEN CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER. NO USER-SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.	THE EXCLAMATION POINT WITHIN THE TRIANGLE IS A WARNING SIGN ALERTING YOU OF IMPORTANT INSTRUCTIONS ACCOMPANYING THE PRODUCT.
SEE MARKING ON BOTTOM/BACK OF PRODUCT.		

WARNING: APPARATUS SHALL NOT BE EXPOSED TO DRIPPING OR SPLASHING AND NO OBJECTS FILLED WITH LIQUIDS, SUCH AS VASES, SHALL BE PLACED ON THE APPARATUS.

WARNING: THE MAIN POWER PLUG MUST REMAIN READILY OPERABLE.

CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, GROUNDING OF THE CENTER PIN OF THIS PLUG MUST BE MAINTAINED.

WARNING: TO REDUCE THE RISK OF FIRE OR ELECTRIC SHOCK, DO NOT EXPOSE THIS APPARATUS TO RAIN OR MOISTURE.

WARNING: TO PREVENT INJURY, THIS APPARATUS MUST BE SECURELY ATTACHED TO THE FLOOR/WALL/RACK IN ACCORDANCE WITH THE INSTALLATION INSTRUCTIONS.

∿	This product is AC only.
CE	

Important Safety Instructions

1. Read these instructions.
2. Keep these instructions.
3. Heed all warnings.
4. Follow all instructions.
5. Do not use this apparatus near water.
6. Clean only with dry cloth.
7. Do not block any ventilation openings. Install in accordance with the manufacturer's instructions.
8. Do not install near any heat sources such as radiators, heat registers, stoves, or other apparatus (including amplifiers) that produce heat.
9. Do not defeat the safety purpose of the polarized or grounding-type plug. A polarized plug has two blades with one wider than the other. A grounding type plug has two blades and a third grounding prong. The wide blade or the third prong are provided for your safety. If the provided plug does not fit into your outlet, consult an electrician for replacement of the obsolete outlet.
10. Protect the power cord from being walked on or pinched particularly at plugs, convenience receptacles, and the point where they exit from the apparatus.
11. Only use attachments/accessories specified by the manufacturer.
12. Use only with the cart, stand, tripod, bracket, or table specified by the manufacturer, or sold with the apparatus. When a cart is used, use caution when moving the cart/apparatus combination to avoid injury from tip-over.
13. Unplug this apparatus during lightning storms or when unused for long periods of time.
14. Refer all servicing to qualified service personnel. Servicing is required when the apparatus has been damaged in any way, such as power-supply cord or plug is damaged, liquid has been spilled or objects have fallen into the apparatus, the apparatus has been exposed to rain or moisture, does not operate normally, or has been dropped.

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CHAPTER 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 SIP Server PC, and which may be required on the phone system side for proper interconnect.

A typical phone switch or SIP lines from a phone service provider are connected to a data network (see Figure 1). Also connected to this common network is an RTS VLINK SIP Server PC and RTS Omneo card(s) in an ADAM intercom. RTS key panel(s) can either be connected via analog, or via network using RVON VOIP or Omneo 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.

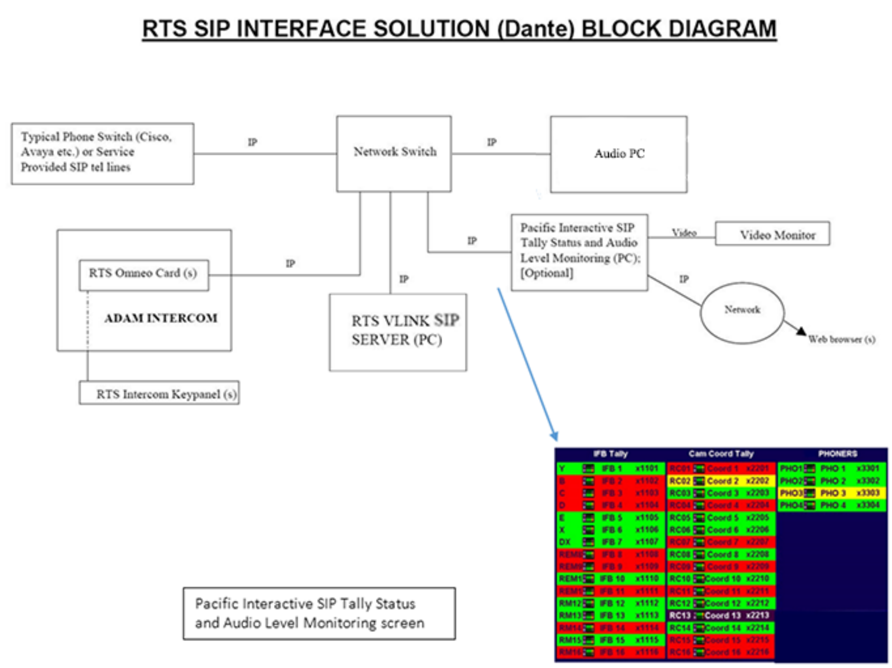


FIGURE 1. RTS SIP Interface Solution (Dante) Block Diagram

Initial Planning

SIP Setup Planning:

Ensure the following information is readily available for the configuration process.

SIP Line Considerations:

- Establish the number of phone lines to be configured
- 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.

IMPORTANT: An IP address is needed for each SIP line service, and most likely user name and password access information for each line being configured.

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 SIP PC must have:

- Direct IP connectivity
- At minimum TCP/UDP Port 5060 open between the VLINK Server PC and the source of the SIP lines
- Public internet access

The Audio Device Interface PC must have:

- Direct IP connectivity
- At minimum TCP/UDP Port 5060 open between the VLINK Server PC and the source of the SIP lines
- Public internet access
- Dante Virtual Soundcard

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.

IMPORTANT: Dante audio requires DIRECT IP connectivity on the same local subnet. For this reason, the VLINK Server PC, the Audio Device Interface PC, and the RTS Omneo cards should be plugged directly into the same network switch for best results.

VLINK Server Port Licensing

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, Audio Device Interface PC, 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.

IMPORTANT: Verify your VLINK Server is licensed for a minimum of 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 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 line calls. 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 SIP PC and RTS ADAM intercom, including Analog, Dante, and MADI.

This document details using Audinate's DANTE Virtual Soundcard (DVS) on the Audio Device Interface PC. Using the Dante audio standard, the connection to an ADAM intercom can then either be via analog or OMI (Omneo) card. To use analog, a conversion box such as Focusrite's Rednet 1 or 2 can be used. However, using RTS's Omneo interface card for the ADAM intercom as described in this document is recommended as setup will be simpler with superior results.

NOTE: Using Dante audio requires purchasing an Audinate Dante Virtual Soundcard license for the VLINK SIP PC.

NOTE: Utilizing a MADI interface from the VLINK Server to an ADAM intercom requires a MADI PC interface card, which can be purchased from RME.

NOTE: OMI cards can be purchased or upgraded with 16, 32, 48, or 64 licensed channels per card. Contact your RTS sales representative for more details.

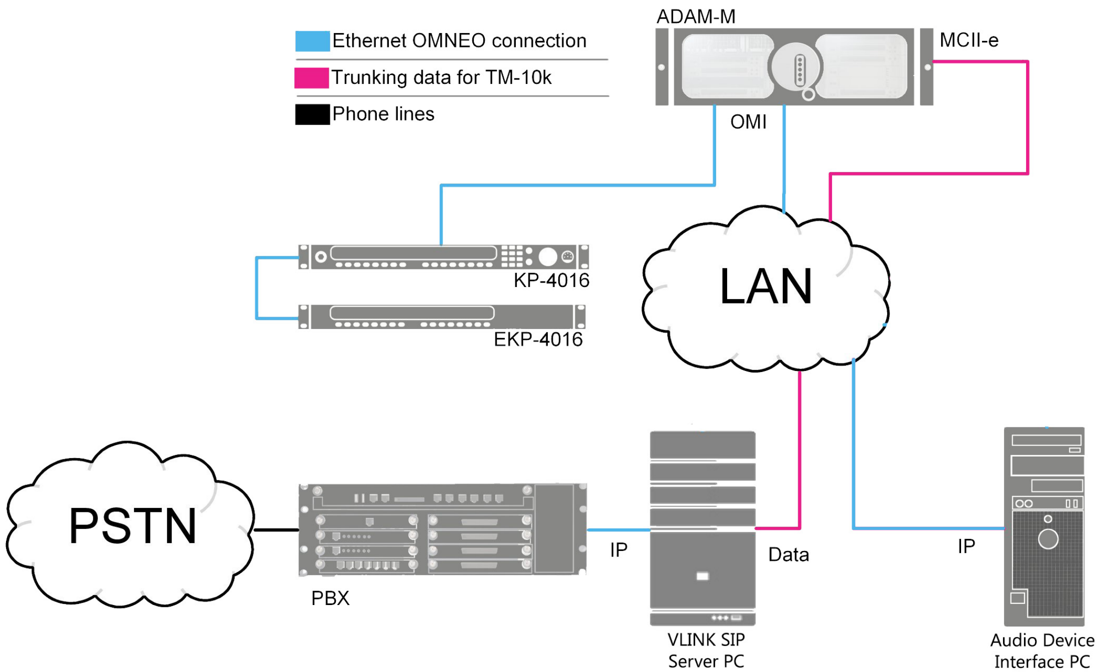


FIGURE 2. SIP Server, audio is sent over Dante/OMNEO

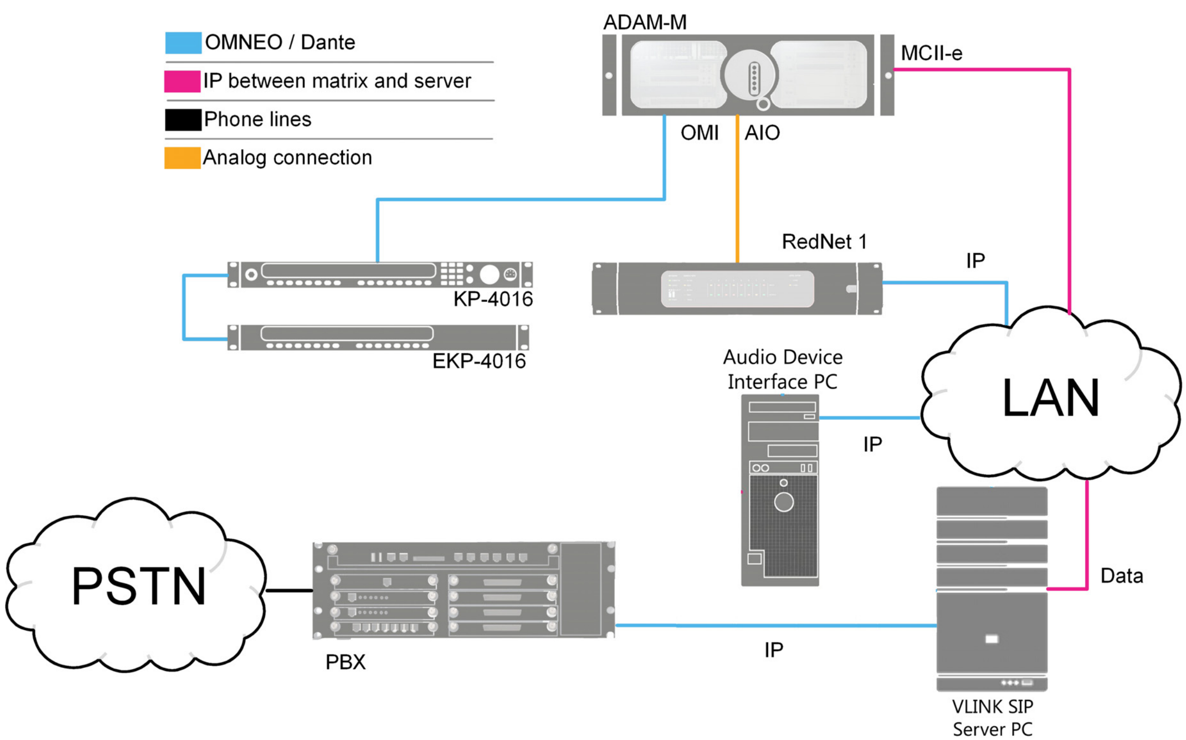


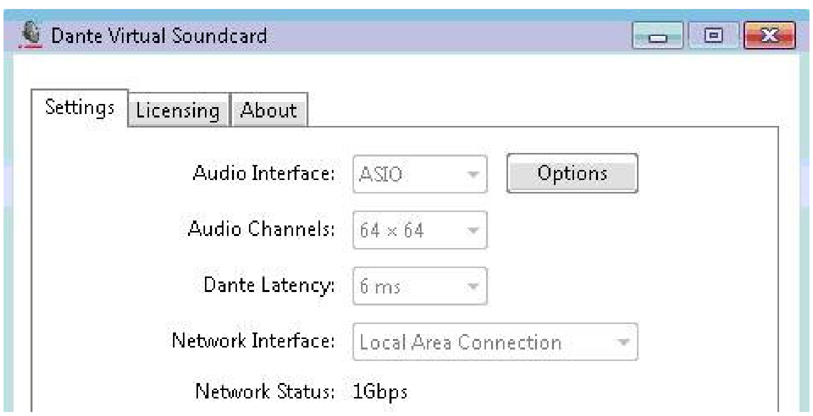
FIGURE 3. SIP Server via analog, using RedNet 1 box

Audinate Dante Virtual Soundcard (DVS)

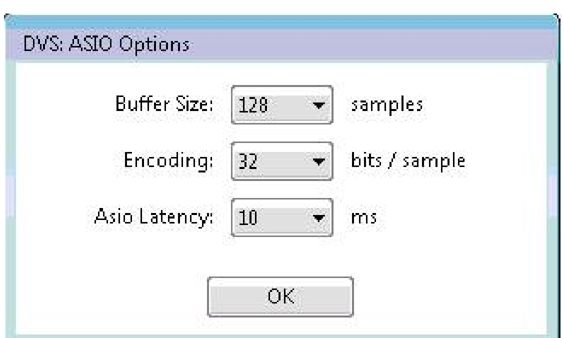
This document details audio connectivity between the OMI and Audio Device Interface PC utilizing Audinate's Dante audio protocol.

To **prepare for audio connectivity**, do the following:

1. Download and install **two software packages** onto the Audio Device Interface PC.
2. From the Audio Device Interface PC, go to **Audinate's website, and register the software package**.
3. Note the **user information, including e-mail provided**.
   This is used for access, to define the licensee, and to install and activate the software.
4. After registering, navigate to the relevant page of Audinate's website and download and install the **Dante configuration tool**.
   This tool is used to map the audio channels between the Audio Device Interface PC and the OMI card.
5. From Audinate's website, purchase a **Dante Virtual Soundcard (DVS) license**.
6. Using the license key provided, install and run the **soundcard software**.
7. Once activated, select setup parameters as shown:
   - Audio Interface = ASIO
   - Audio Channel = 64 x 64
   - Dante Latency = 6ms
   - Network Interface = Choose the appropriate Ethernet connection the PC uses



8. Click the **options button** and configure the **fields** as shown:
   - Buffer size = 128 samples
   - Encoding = 32 bits / sample
   - Asio Latency = 10 ms



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 to clearly identify which phone line connects to which intercom port. For example, if a 16 channel Omneo card is used, and it is installed as intercom ports 1-16, and the phone numbers being interfaced are 555-555-1001 through 555-555-1016, you should determine and note each line's port assignment. For this example, 555-555-1001 might connect to intercom port 1, 555-555-1002 to port 2, through 555-555-1016 connecting to intercom port 16.

NOTE: For more information about planning port assignments, see the OMI Technical Manual on the RTS Intercoms website.

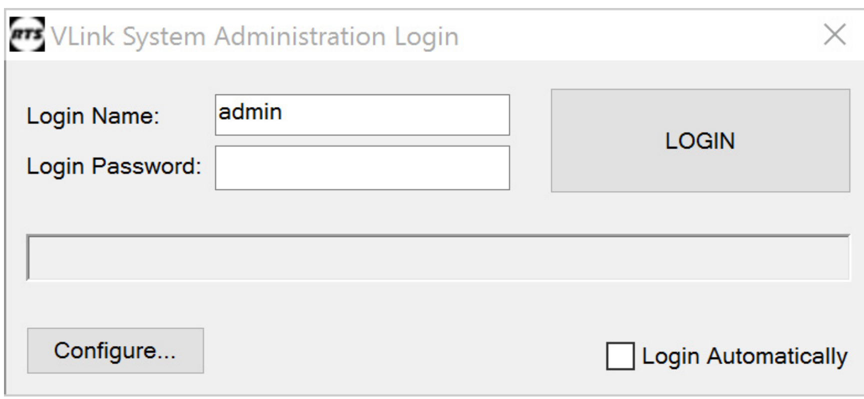
Also note that VLINK single SIP Server features require ADAM Master Controller software version V3.1.0 or later and AZEDIT version V4.7.0 or later. VLINK multiple SIP Server features require ADAM Master Controller software version V3.3.0 or later and AZEDIT version V5.1.0 or later. Contact your RTS support representative should you require this software or assistance with upgrades.

CHAPTER 3

VLINK Server Configuration: VLINK Server to SIP / Phone Switch

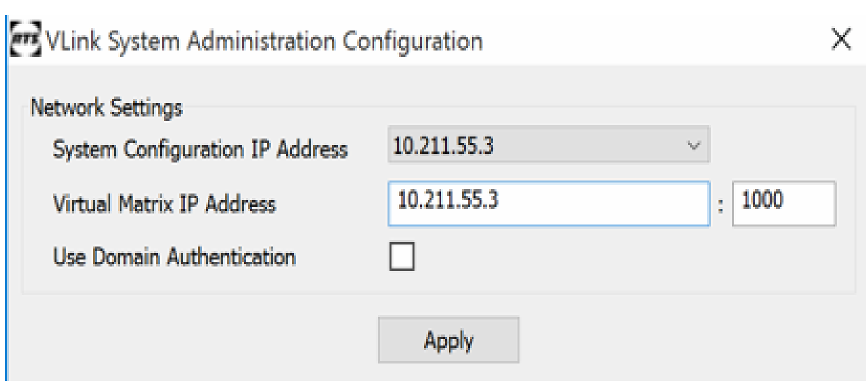
General VLINK System Configuration:

1. Open the **VLINK System Administration application**.
2. At the Login screen, enter a **Login Name** (default: admin) and **Login Password** (default: no password).



3. If this is the initial log-in, click the **Configure... button**.
4. Enter the **IP address of the VLINK SIP Server PC, immediately followed by :1000 (no spaces)**.

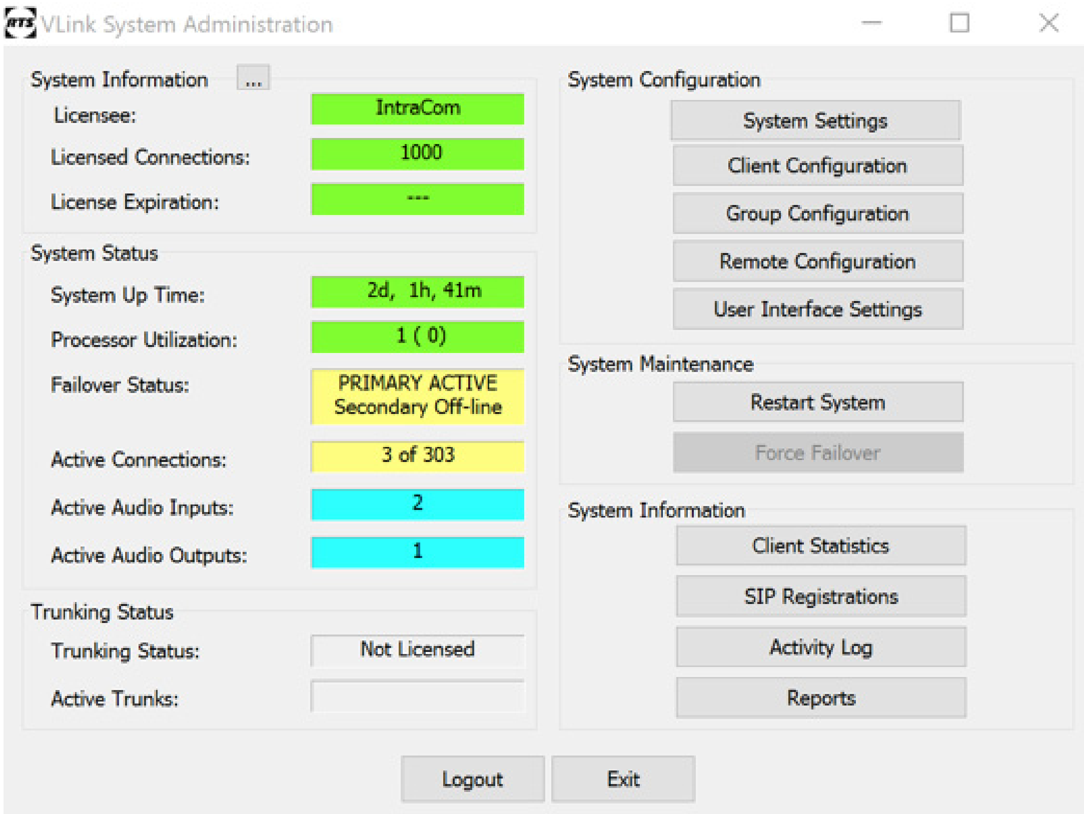
NOTE: For detailed instructions on using the VLINK System Administration application, and configuring a VLINK system in general, refer to the VLINK System Administration Guide: www.rtsintercoms.com



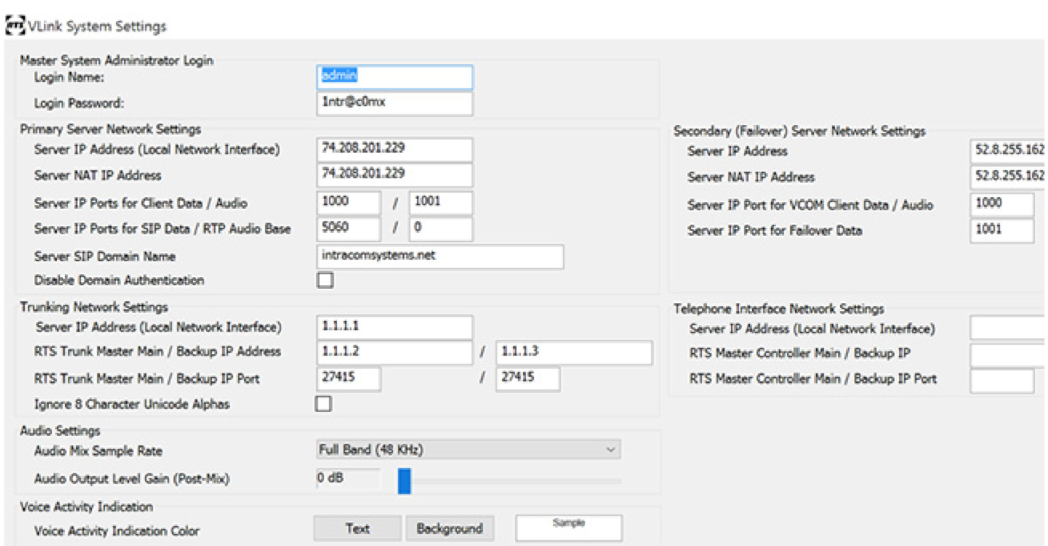
Configure Main System Settings for SIP

To **configure the main system settings for SIP**, do the following:

1. From the main System Administration window, click the **System Settings button**.
   The VLink System Settings window appears.

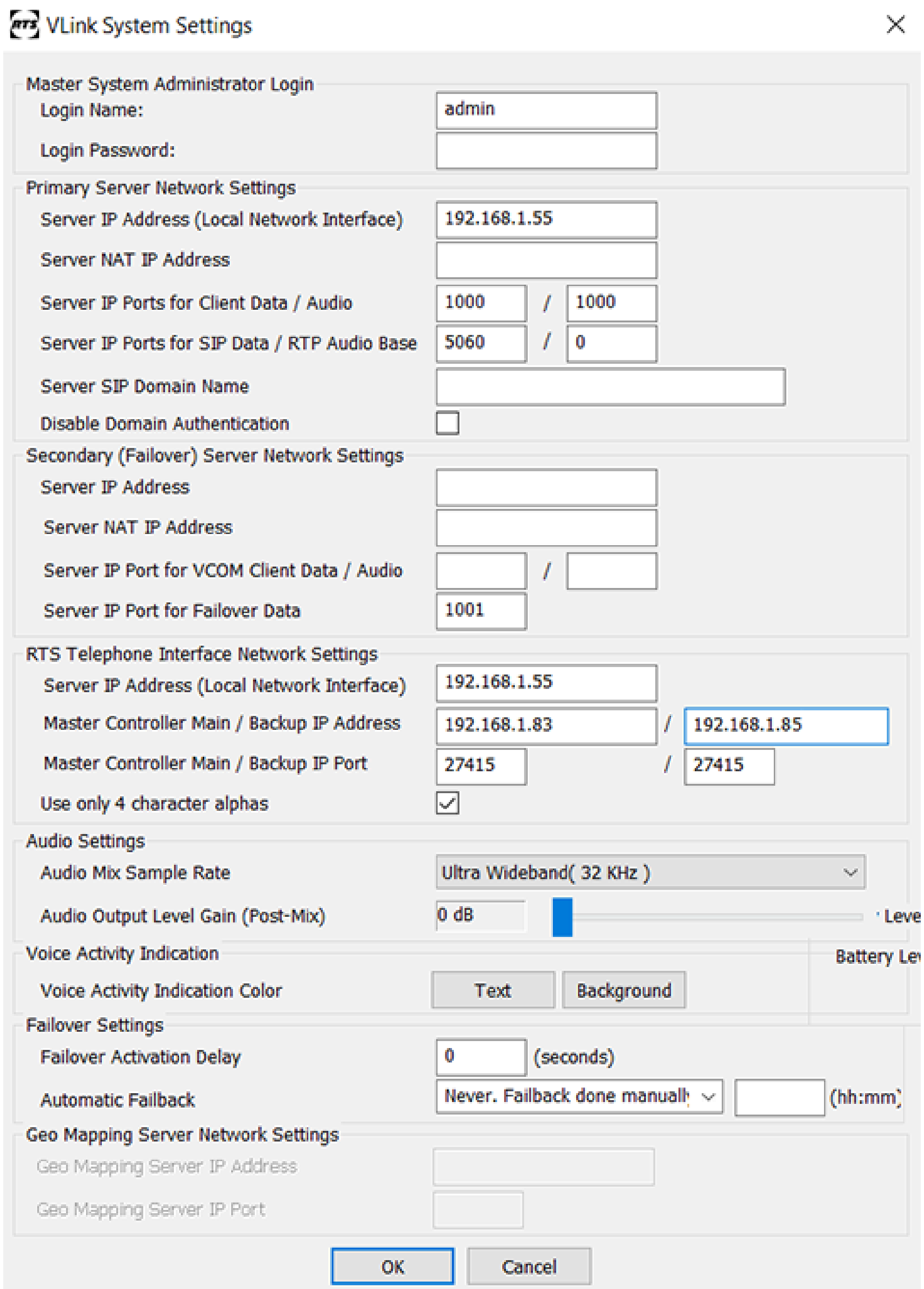


2. Verify the Server IP Ports for SIP Data/RTP Audio Base fields are set as **5060 / 0** (default settings).
   OR
   In the Server IP Ports for SIP Data/RTP Audio Base fields, enter **5060 / 0**.



Under RTS Telephone Interface Network Settings Group Box

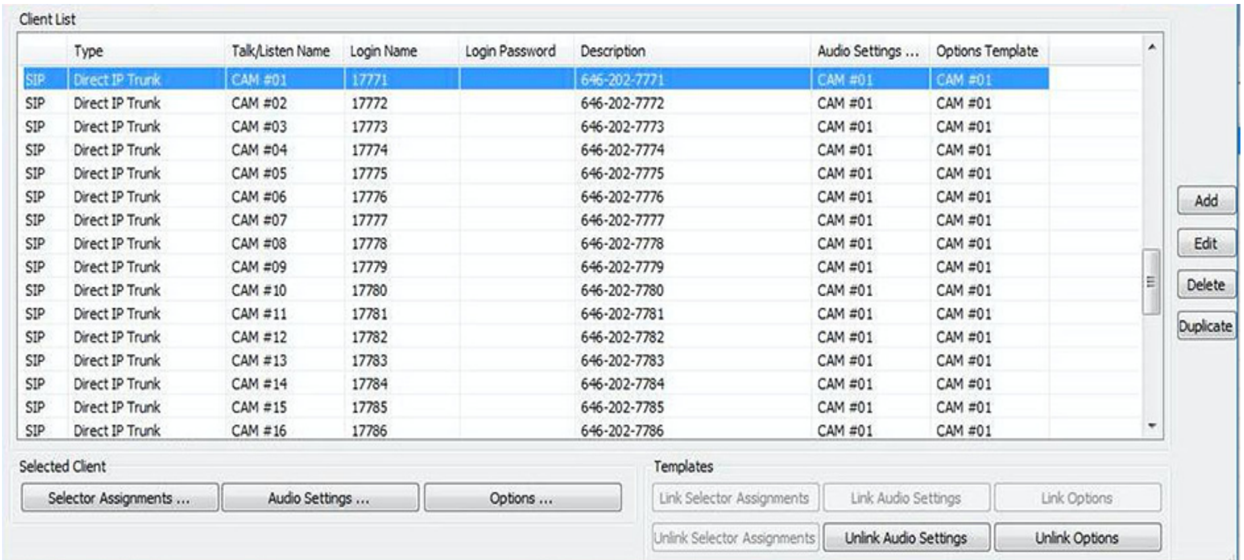
3. In the Server IP Address (Local Network Interface) field, enter the **IP address of the VLINK Server**.
4. In the Master Controller Main/Back IP fields, enter the **IP address of the ADAM main master controller** followed by the **IP address of the ADAM backup master controller**.
5. In the Master Controller Main/Backup IP Port fields, enter **27415 / 27415**.



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**, do the following:

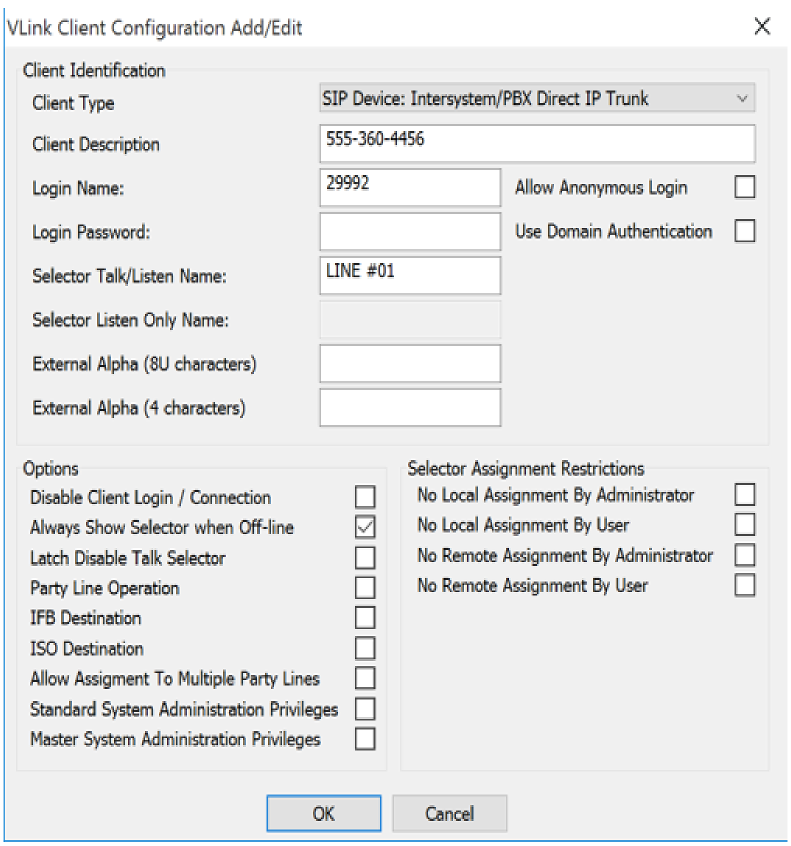
1. From the main System Administration, click the **Client Configuration button**.
   The VLink Client Configuration window appears.



Client List

Type		Talk/Listen Name	Login Name	Login Password	Description	Audio Settings ...	Options Template
SIP	Direct IP Trunk	CAM #01	17771		646-202-7771	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #02	17772		646-202-7772	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #03	17773		646-202-7773	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #04	17774		646-202-7774	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #05	17775		646-202-7775	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #06	17776		646-202-7776	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #07	17777		646-202-7777	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #08	17778		646-202-7778	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #09	17779		646-202-7779	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #10	17780		646-202-7780	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #11	17781		646-202-7781	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #12	17782		646-202-7782	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #13	17783		646-202-7783	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #14	17784		646-202-7784	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #15	17785		646-202-7785	CAM #01	CAM #01
SIP	Direct IP Trunk	CAM #16	17786		646-202-7786	CAM #01	CAM #01

2. From the Client List group box, click the **Add button**.
   The VLink Client Configuration Add/Edit window appears.



3. From the Client Type drop down menu, select **SIP Device: Intersystem/PBX Direct IP Trunk**.

4. In the Client Description field, enter the **phone number**.

IMPORTANT: The Client Description field 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.

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5. In the Login Name field, enter a **Login Name**.
6. In the Login Password field, enter a **Login Password**.

IMPORTANT: Use the login name and password provided by your SIP phone line provider. A unique login name and password is needed for each SIP phone line.

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7. In the Selector Talk/Listen field, enter a **unique name**, which appears on a Control Panel selector, and on the optional Pacific Interactive TIF Tally Display.

IMPORTANT: The Selector Talk/Listen Name is displayed on the optional Pacific Interactive Status Display System, so it is highly recommended to use the last 4 digits of the phone number for the Selector Talk/Listen name, if possible.

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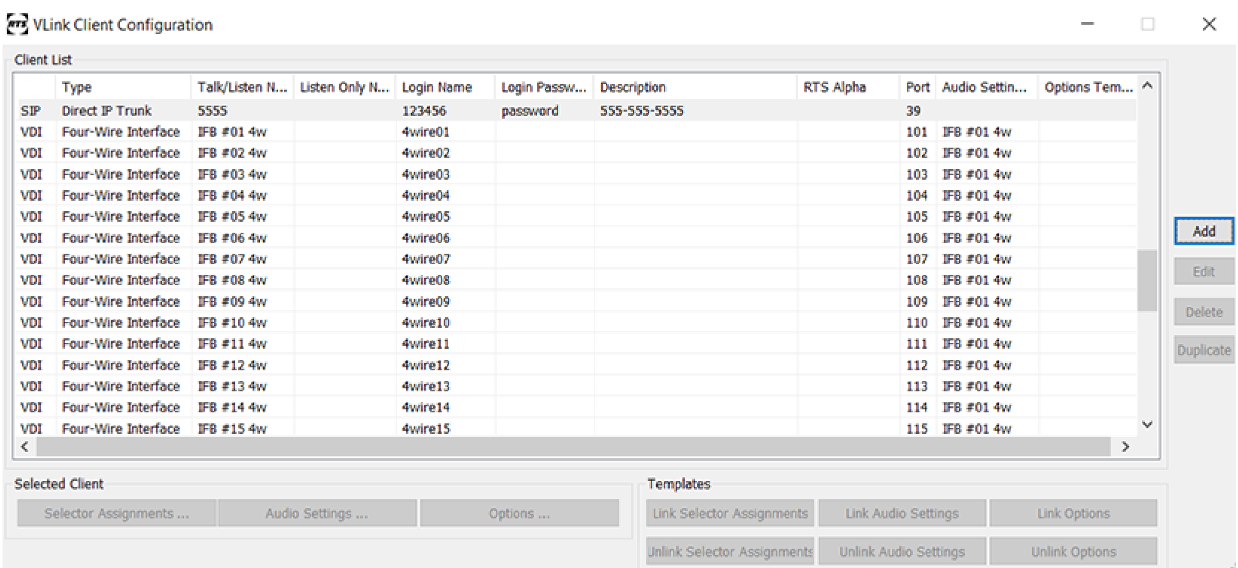
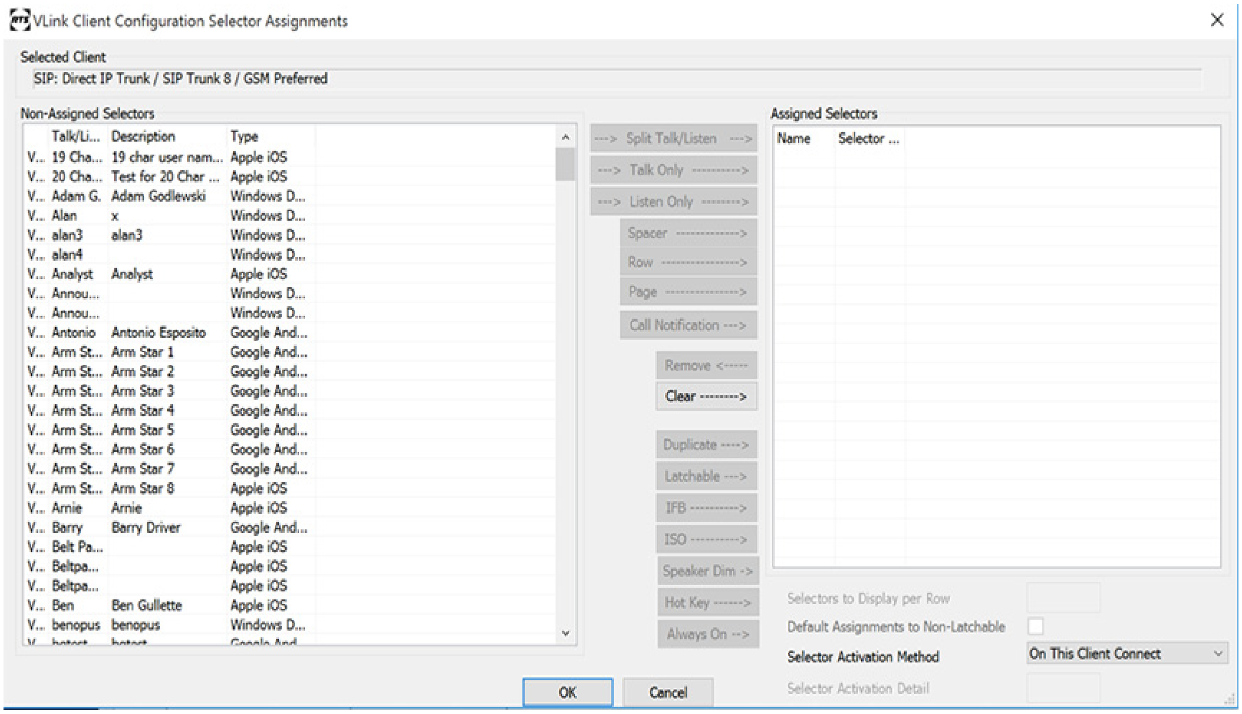
8. Select the **Always Show Selector when Off-line check box**.
9. Click the **OK button**.
   The VLink Client Configuration Add/Edit window closes. The added client is now visible in the Client Configuration window.



10. From the Client List Window, select the **client**.
    Selected row is highlighted in blue and the buttons at the bottom of the window are now active.

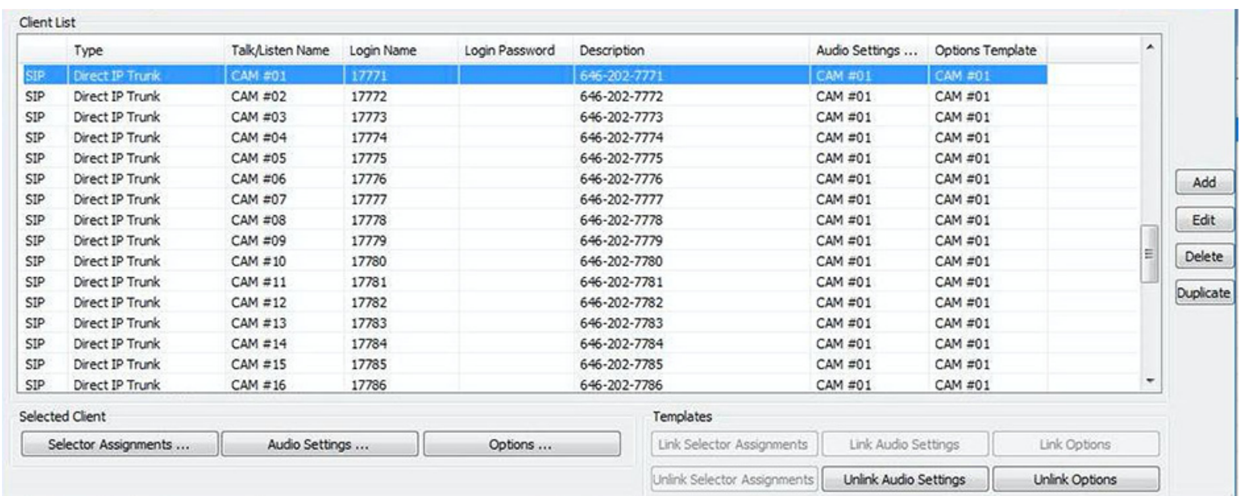
11. Click the **Selector Assignments button**.
The Client Configuration Selector Assignments window appears. The Selected Client field is populated.



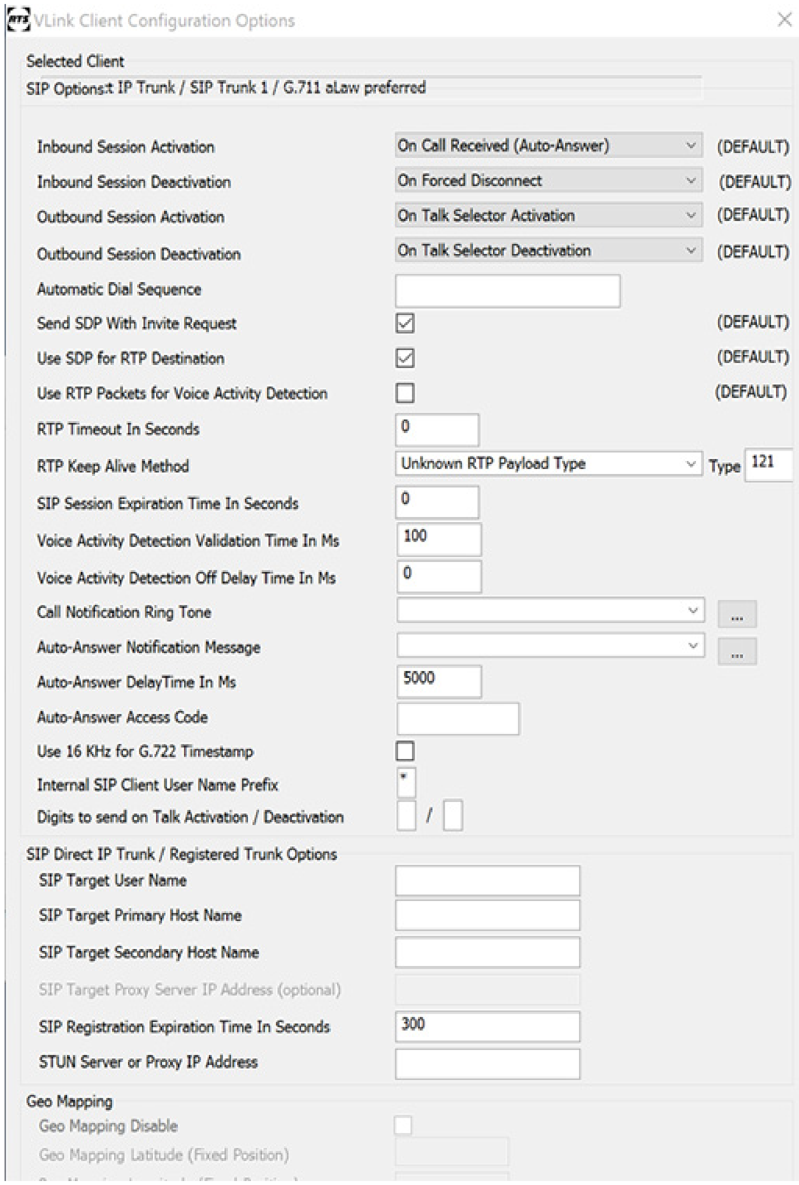
12. Verify the Selector Activation Method drop down menu displays the **On This Client Connect option**.

13. Click the **OK button**.
The Client Configuration Selector Assignments window closes.

14. In the Client List window, verify the newly added client is still highlighted.



15. Click the **OPTIONS button**.
The Client Configuration Options window appears. The Selected Client field is populated.



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

IMPORTANT:	Each additional SIP line must be created separately, using the Adding a SIP Phone Line instructions.

Customizing the Client Configuration Options

If necessary, there are several user defined options that can be set up for each phone line.These fields are found in the VLINK Client Configuration Options window.

Session Configurations

Inbound Session Activation Drop Down Menu

The Inbound Session Activation drop down menu is used to select the method used to answer an incoming call to the Server.

Available options are:

Disabled	Disables any incoming 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 Drop Down Menu

The **Inbound Session Deactivation** drop down menu is used to select the method used to hang up a call which is made to the Server.

Available options 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.

Outbound Session Activation Drop Down Menu

The **Outbound Session Activation** drop down menu is used to select the method used to go "off-hook" on a phone line and dial an outbound call.

Available options are:

Disabled	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.

Outbound Session Deactivation Drop Down Menu

The **Outbound Session Deactivation** drop down menu is used to select the method used to hang up a call which is originally dialed outbound from an RTS panel.

Available options 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 Request	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.

Phone Line Sound Configurations

There are two system or user provided types of configurable phone line sounds which are in a .wav sound file format*.

Call Notification Ringtone Drop Down Menu

The **Call Notification Ringtone** drop down menu is used to select the .wav file broadcast on the intercom system whenever a phone line is ringing.

To **configure the call notification ringtone**, do the following:

> From the Call Notification Ringtone drop down menu, select the desired **.wav file**.
OR
Click the **browse button** to navigate to select a .wav file from a network folder.

NOTE: 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.

Auto-Answer Notification Message Drop Down Menu

The **Auto-Answer Notification Message** drop down menu is used to select the .wav file played to 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.

To **configure the auto-answer notification sound**, do the following:

> From the Auto-Answer Notification message drop down menu, select the desired **.wav file**.
OR
Click the **browse button** to navigate to select a .wav file from a network folder.

NOTE: 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.

CHAPTER 4

VLINK Server 4-Wire Dante Interface to Omneo

4-Wire Interface Setup

-Create a 4-Wire Interface for Phone Line Audio

-Configuring the 4-Wire Client Options

-Configuring the Client Audio Settings

To **connect the phone line** through the Audio Device Interface PC and **provide audio** to the RTS ADAM intercom, do the following:

Create a 4-wire Interface for Phone Line Audio

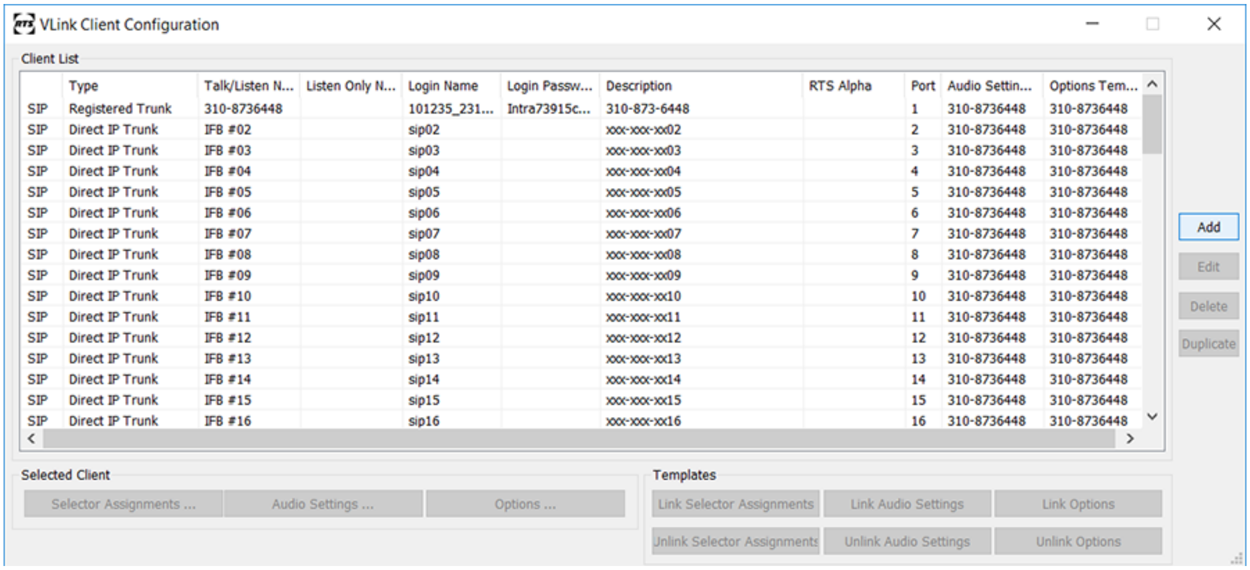
When creating the **internal audio connections** between the Server and PC audio channels (in this document using Dante Virtual Soundcard* software), there is one separate 4-wire for each phone line that needs to be connected to the RTS intercom.

NOTE: 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**, do the following:

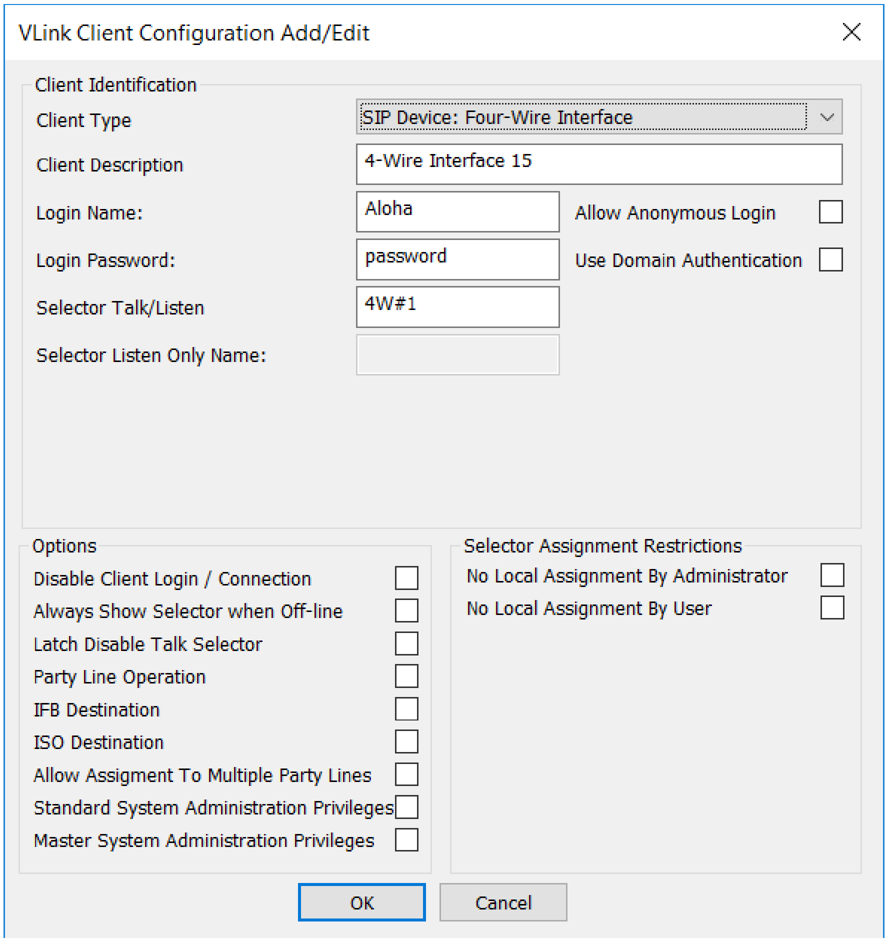
1. From the main System Administration window, click the **Client Configuration button**.
The Client Configuration window appears.

2. From the Client Configuration window, click the **Add button**.
The Client Configuration Add/Edit window appears.

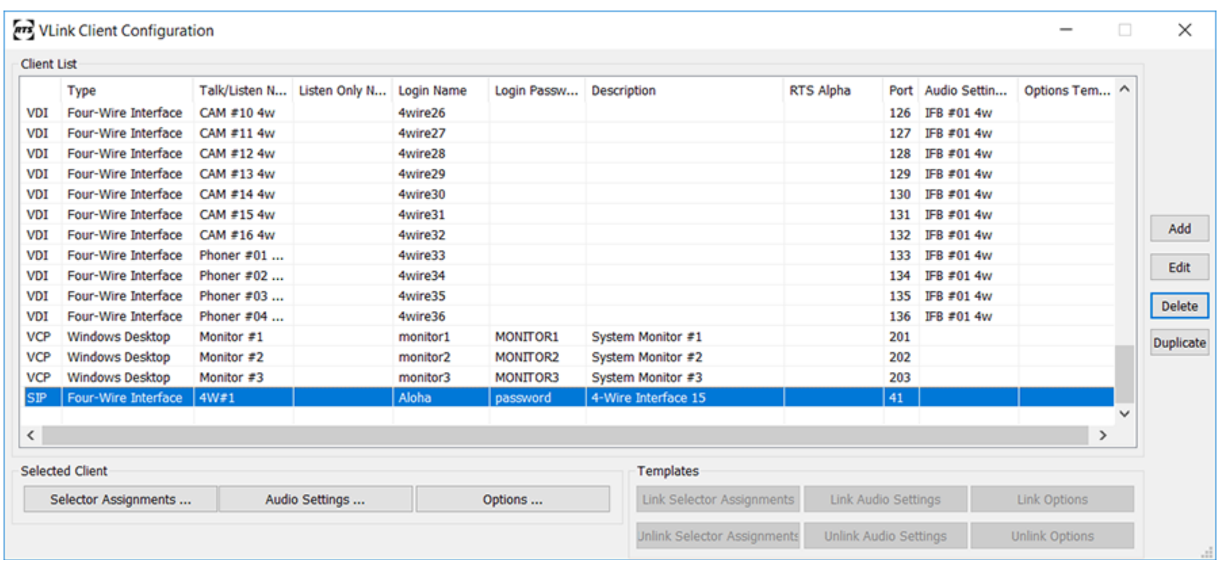


3. From the Client Type drop down menu, select **VLink Device Interface: Four-Wire Interface**.
4. In the Client Description field, enter a **client description**.
5. In the Login Name field, enter a **login name**.
6. Deselect the **Use Domain Authentication check box**.
The Login Password field becomes active.
7. In the Login Password field, enter a **login password**.
8. In the Selector Talk/Listen Name field, enter a **selector talk/listen name**.

IMPORTANT: 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. through 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.



9. Click the **OK button**.
The VLink Client Configuration Add/Edit window closes. The added client is now visible in the Client Configuration window.

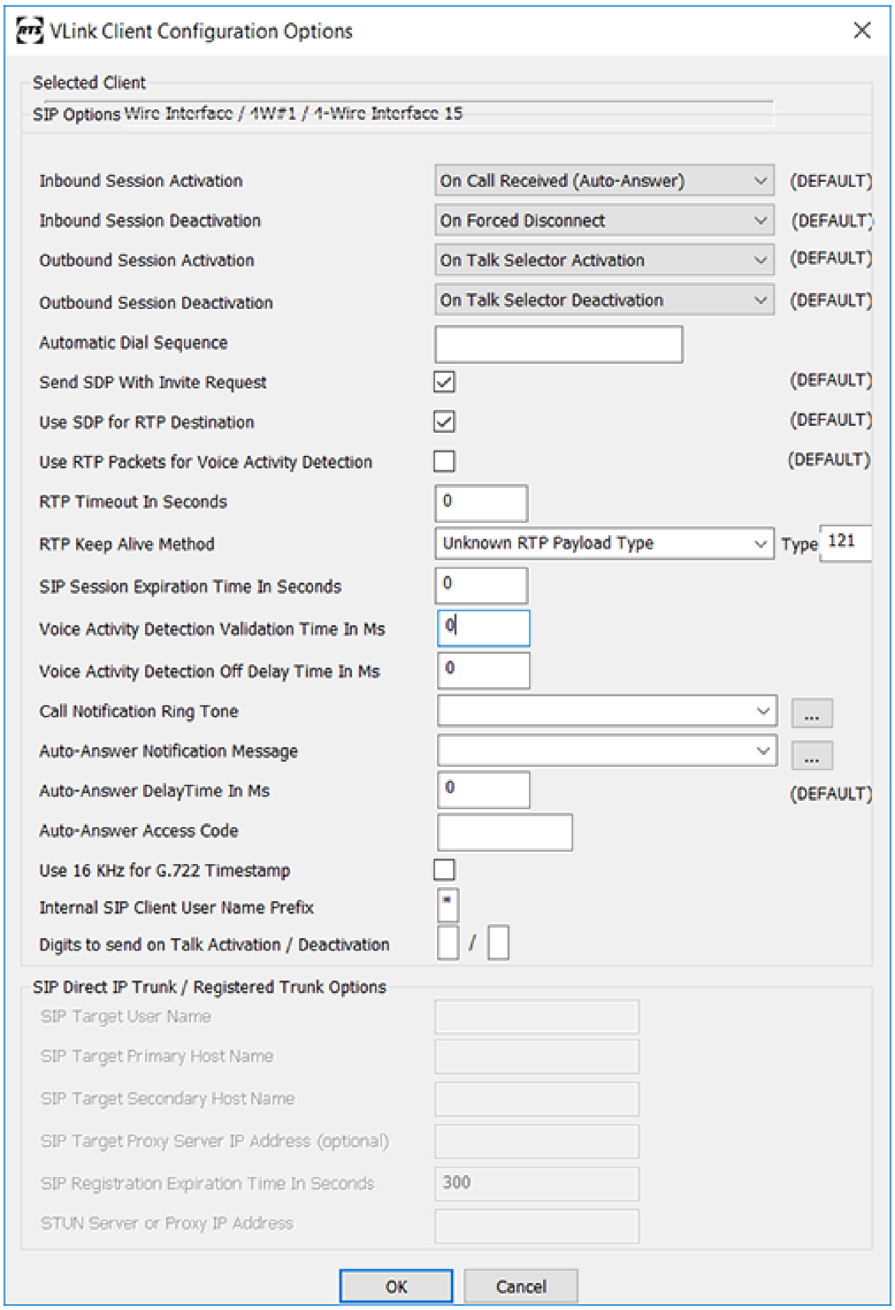


Configuring Client Options

1. From the Client Configuration window, select the **client**.
   The selected row is highlighted in blue and the buttons at the bottom of the window are now active.
2. Click the **Options button**.
   The Client Configuration Options window appears. The Selected Client field is populated.

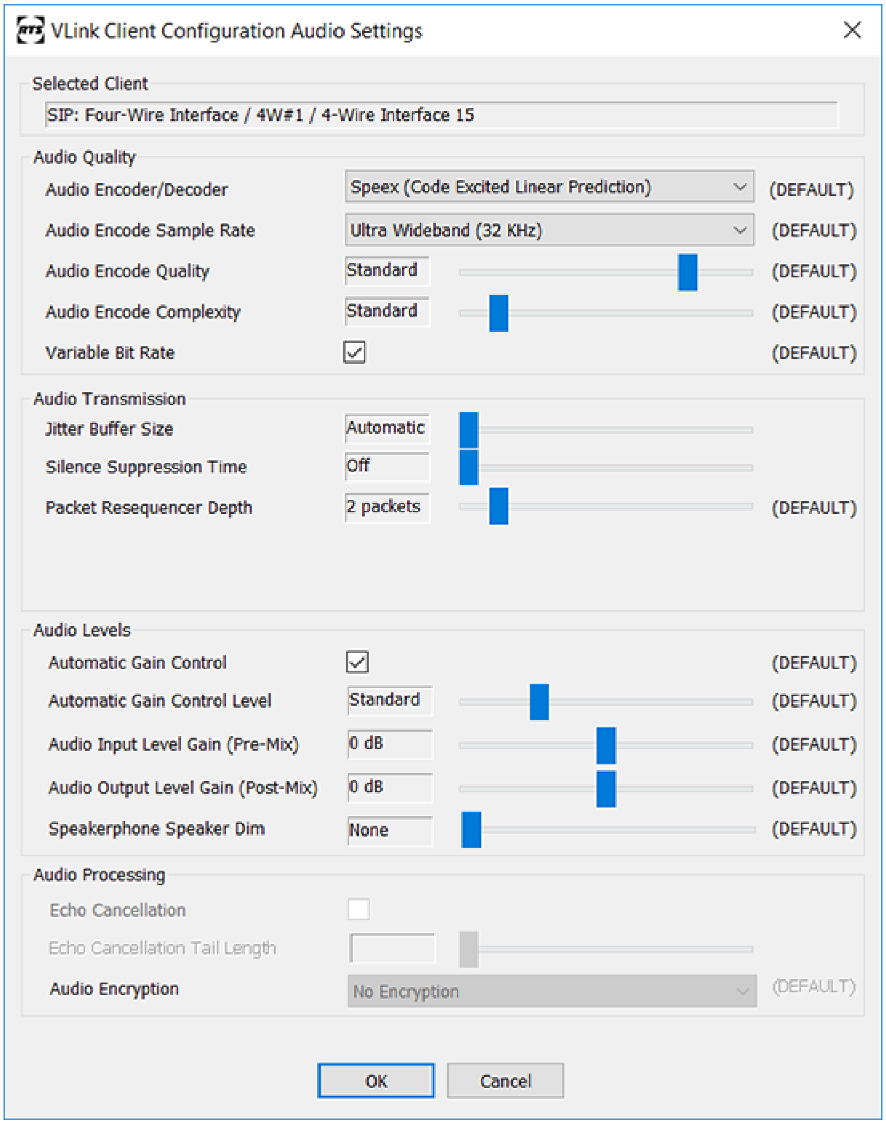
Control Panel/Device Interface Options Group Box

3. In the Voice Activity Detection Time in Ms field, enter **0**.
4. Click the **OK button**.
   The Client Configuration Options window closes.



Configuring Client Audio

1. In the Client Configuration window, verify the **same entry** is still highlighted.
2. Click the **Audio Settings button**.
   The Client Configuration Audio Settings window appears.



Audio Quality Group Box

3. From the Audio Encoder/Decoder drop down menu, select **No Compression / Very High Bitrate**.
4. From the Audio Encode Sample Rate drop down menu, select **Ultra Wideband (32 KHz)**.

Audio Transmission Group Box

5. Using the Jitter Buffer Size slider, set the jitter buffer size to **Automatic** (slide to the leftmost position).
6. Using the Silence Suppression Time slider, set the silence suppression time to **Off** (slide to the leftmost position).
7. Verify the **Packet Resequencer Depth is set to 2 packets**.

Audio Levels Group Box

8. Deselect the **Automatic Gain Control check box**.
9. Verify the **Automatic Gain Control Level is set to Standard**.

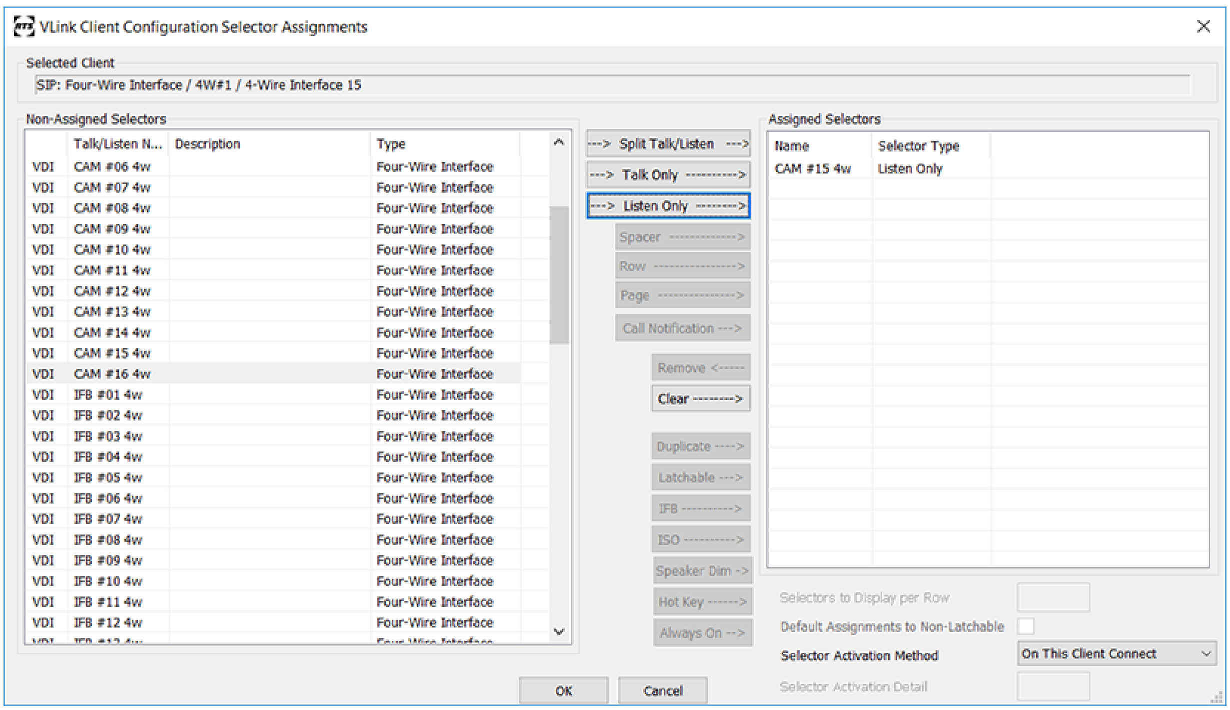
10. Verify the **Audio Input Level Gain (Pre-Mix) is set to 0 dB**.
11. Verify the **Audio Output Level Gain (Post-Mix) is set to 0 dB**.
12. Verify the **Speakerphone Speaker Dim is set to None (slide to the leftmost position)**.

Audio Processing Group Box

13. Verify the Echo Cancellation check box is **not selected**.
14. Using the Echo Cancellation Tail Length slider, set the echo cancellation tail length to **100 ms**.
15. Click the **OK button**.
    The Client Configuration Audio Setting window closes.

Configuring Selector Assignment Button

1. In the Client Configuration window, verify the **same entry** is still highlighted.
2. Click the **Selector Assignments button**.
   The Client Configuration Selector Assignments window appears. The Selected Client field is populated with the newly created 4-wire client.



3. From the Non-Assigned Selectors panel on the left, select the desired **4-wire interface**.
4. Click the **Listen Only button**.
   The selected 4-wire interface is now displayed in the Assigned selectors panel.
5. From the Selector Activation Method drop down menu, select **On This Client Connect**.
6. Click the **OK button**.
   The Client Configuration Selector Assignments window closes.

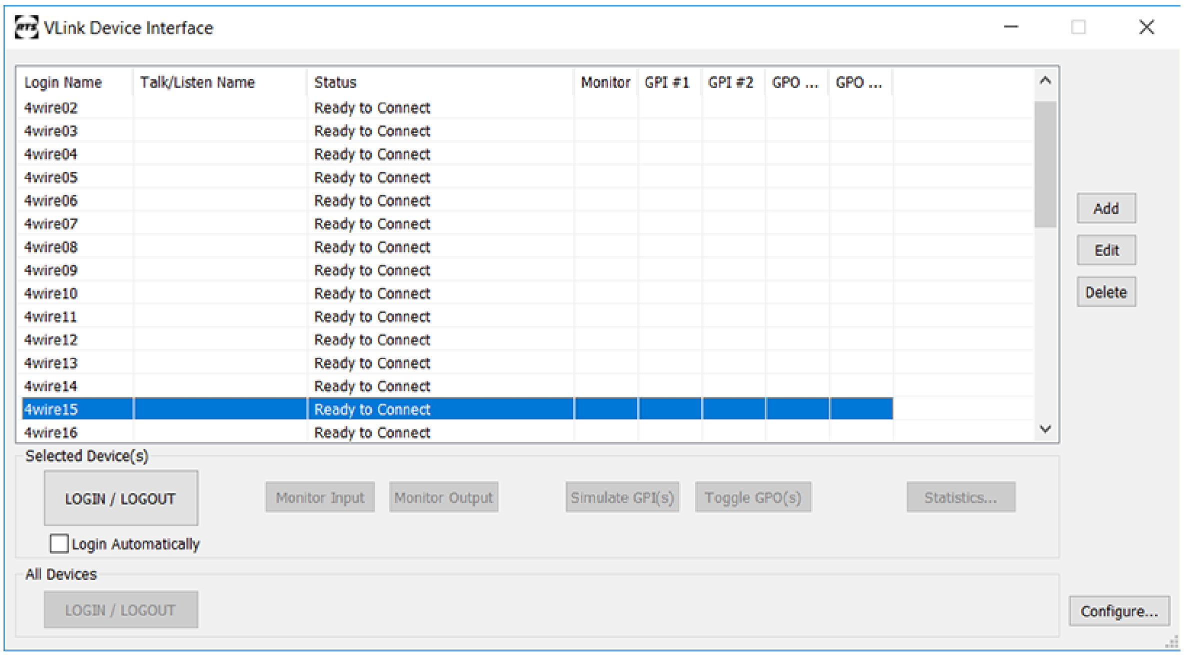
NOTE: Repeat these steps to add additional lines, as necessary.

Internal Server Dante Virtual Soundcard Channel Connections

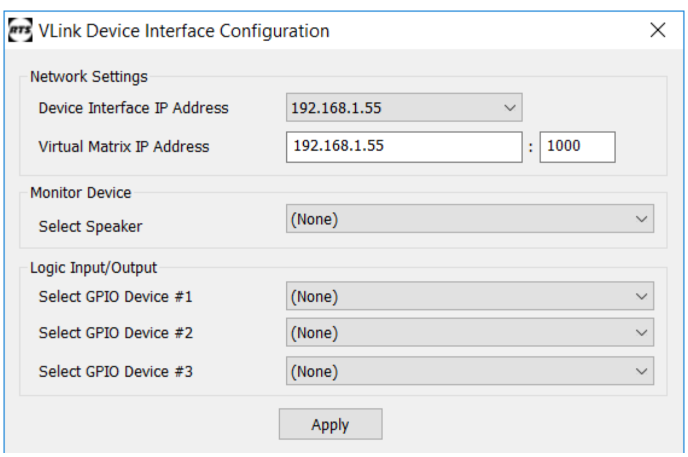
To **set up the internal audio connections between each phone line's 4-wire interface and the Dante Virtual Soundcard audio channels**, do the following:

1. After adding one 4-wire for each phone line in the system, open the **VLINK Device Interface Application**.
The Device Interface window appears

   NOTE: This application would have been installed as part of the original VLINK Server install, and would likely be found as an icon on your desktop. 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 at www.rtsintercoms.com

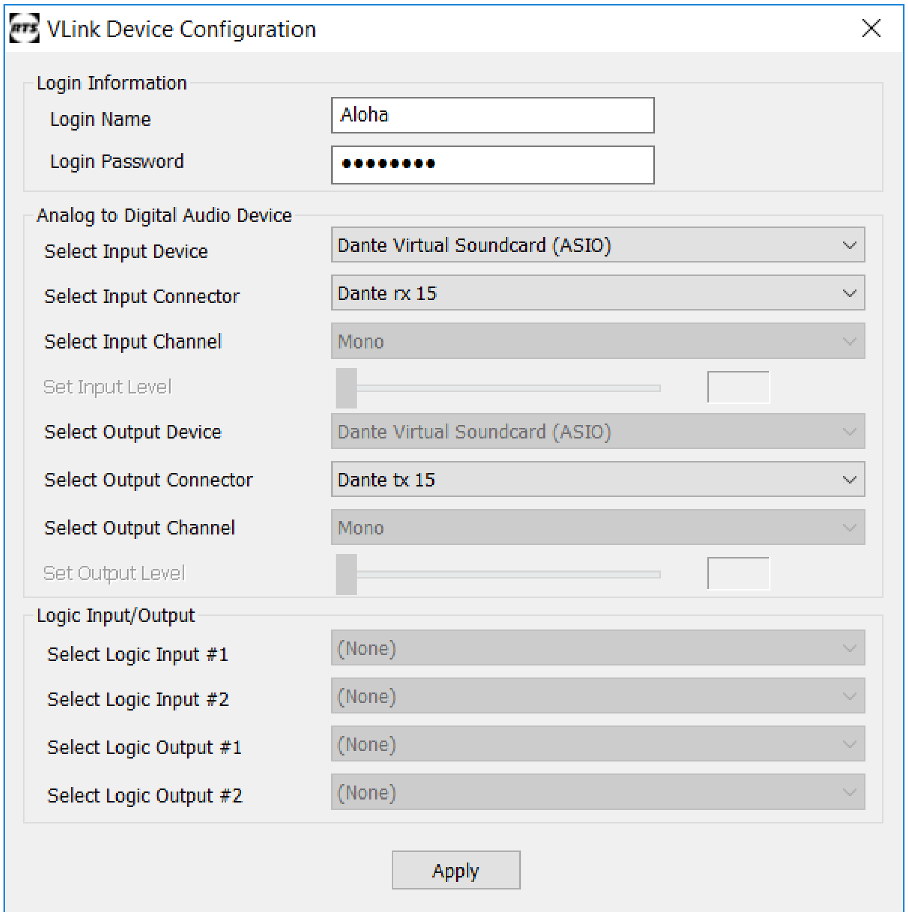


2. Click the **Configure button**.
The Device Interface Configuration window appears.



3. Enter your **Audio Device Interface PC IP address** in the Virtual Matrix IP Address field.
4. Click the **Apply button**.
The Device Interface Configuration window disappears.

5. From the Device Interface window, click the **Add button**.
The VLink Device Configuration window appears.



6. Enter the **Login Name** and **Login Password** that were just used when setting up each 4-wire interface connection.

IMPORTANT: It is critical that the same login names and passwords are used (these are case sensitive) as were previously configured for the 4-wire interface connections.

7. From the Select Input Device drop down menu, select **Dante Virtual Soundcard (ASIO)**.
8. From the Select Input Connector drop down menu, select **Dante rx xx, where xx is the sequential phone line number & 4-wire interface number**.

   EXAMPLE: If the first phone line associated with 4-wire 4W#1 is being added, select **Dante rx 01**.
9. From the Select Output Connector drop down menu, select **Dante rx xx, where xx is the sequential phone line number & 4-wire interface number**.
10. Click the **Apply button**.
The Device Configuration window closes and the added device is visible in the Device Interface window.
11. Repeat these steps to add **one Device Interface entry for each phone line / 4-wire interface**.

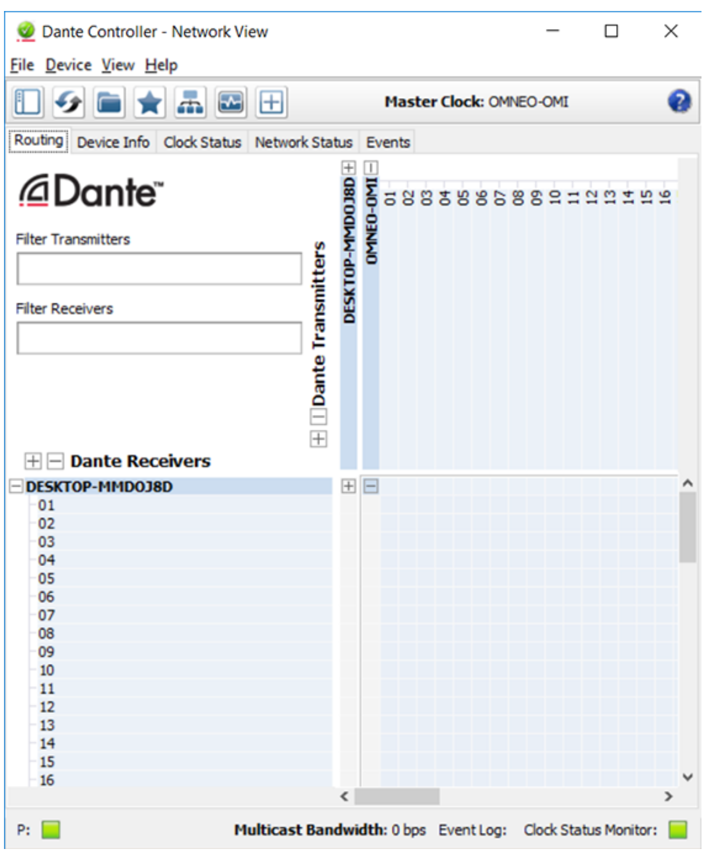
CHAPTER 5

Setup of Audio Channels Between VLINK Server and Omneo Card(s)

Once the SIP phone line has been configured to connect to the VLINK SIP Server PC, and internally the Audio Device Interface PC has been configured to send audio from an installed Dante Virtual Soundcard, the next step in configuration is to map audio channels from the Audio Device Interface PC's virtual Soundcard to the associated OMI channels.

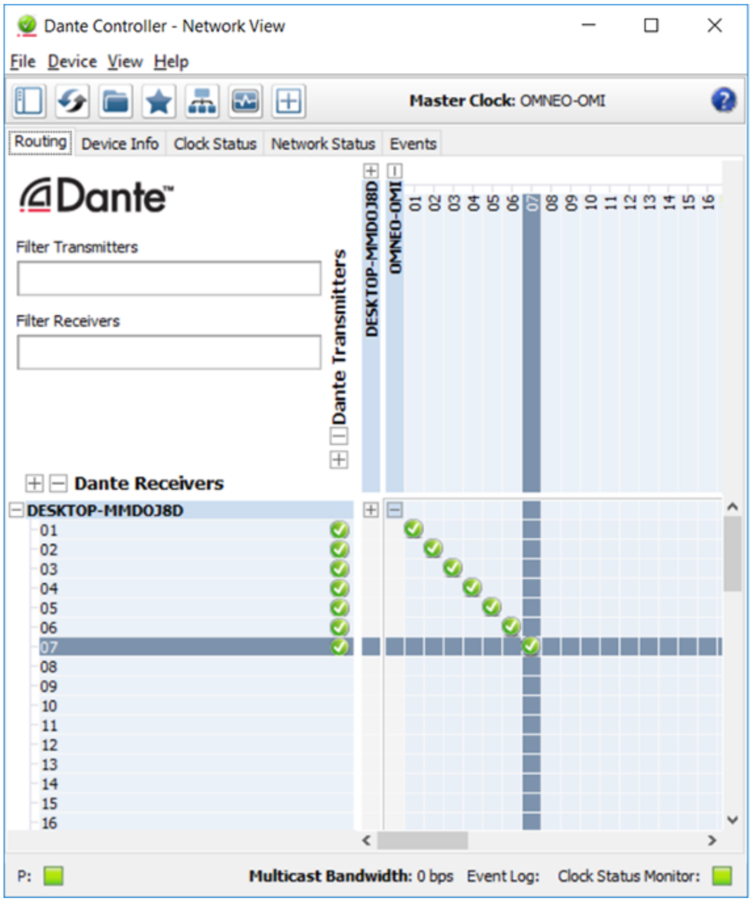
IMPORTANT: You must pre-design which phone lines will connect to which intercom port, as this mapping relationship will also need to be used later for phone line control.

To **configure a one-for-one mapping of phone lines in the Server to intercom ports via Omneo in the intercom** using the Dante Controller software, do the following:

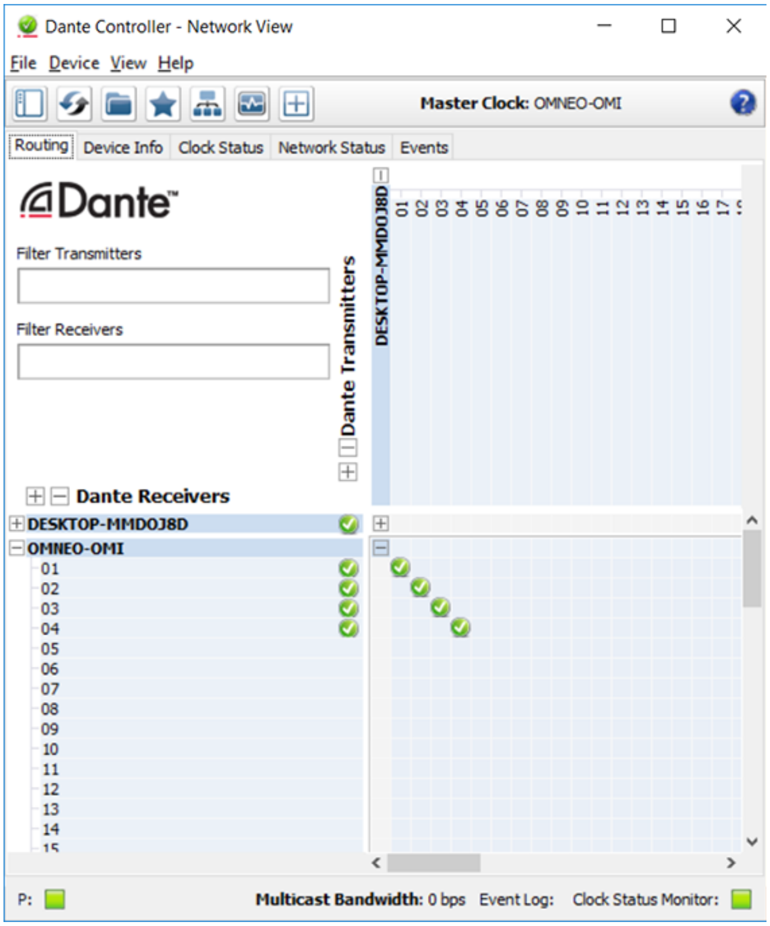


1. Click the **+ button next to the Audio Device Interface PC name on the left**.

2. Click the **+ button next to the OMI Card name on the top**.
3. Click on the **boxes within the grid at the intersection of the desired Audio Device Interface PC line numbers and the desired OMI line numbers**.
   This creates an audio connection (indicated by a green circle) in a diagonal so each Audio Device Interface PC channel is mapped in one direction to the associated OMI card channel.



4. Click the **- button next to the Audio Device Interface PC name on the left**.
5. Click the **- button next to the OMI Card name at the top**.



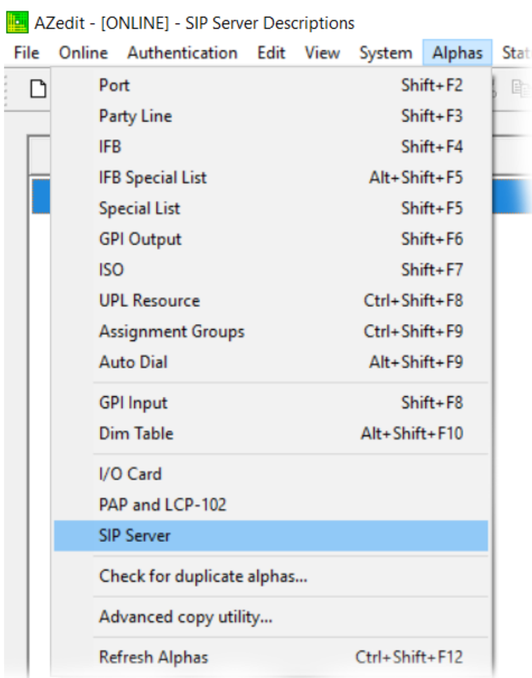
6. Click the **+ next to the OMI Card name on the left**.
7. Click the **+ next to the Audio Device Interface PC name on the top**.
8. Click on the **boxes within the grid at the intersection of the desired OMI line numbers and the desired Audio Device Interface PC line numbers**.
   This creates an audio connection (indicated by a green circle) in a diagonal so each OMI card channel is mapped in the other direction to the associated Audio Device Interface PC channel.

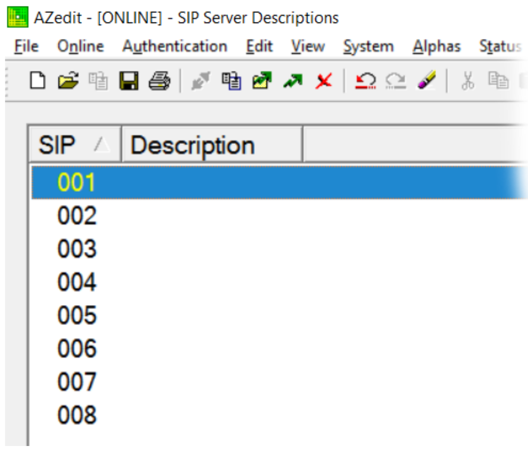
ADAM/AZEDIT VLINK Server Setup

To **connect a VLINK SIP Server PC to be used for SIP phone line connection to an ADAM intercom using AZedit software**, do the following:

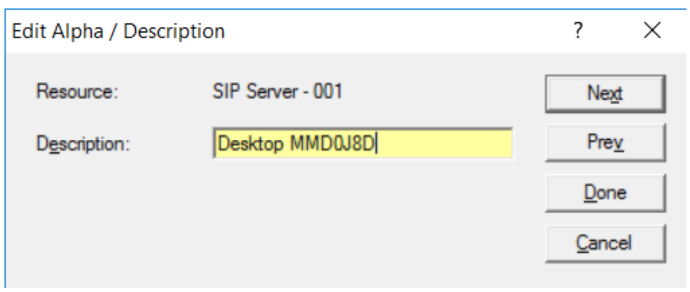
NOTE: The RTS ADAM intercoms support up to eight redundant and separate SIP Servers.

1. Click **Alphas | SIP Server**.
   The list of SIP servers is displayed.

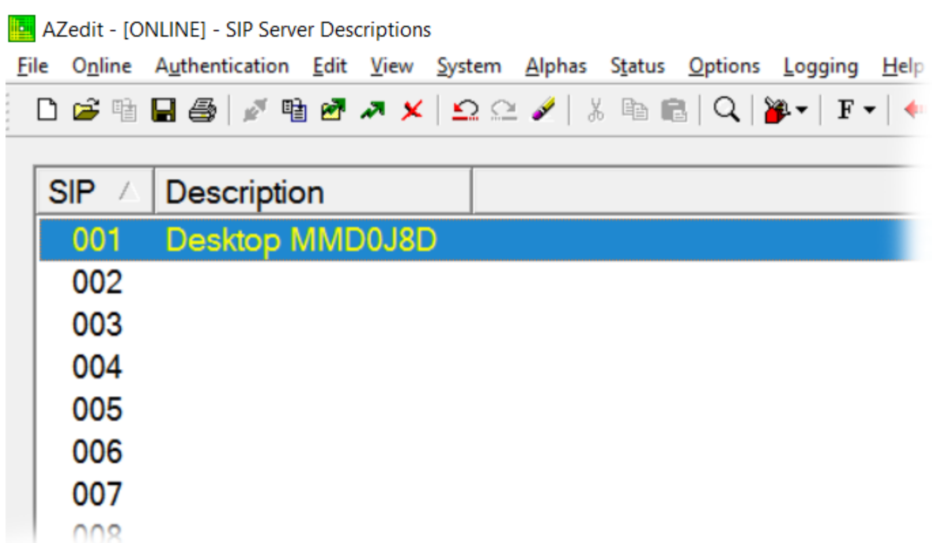




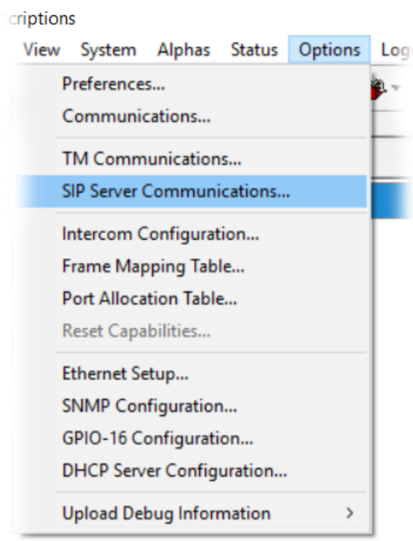
2. Double click the **SIP Server being configured**.
   The Edit Alpha / Description window appears.

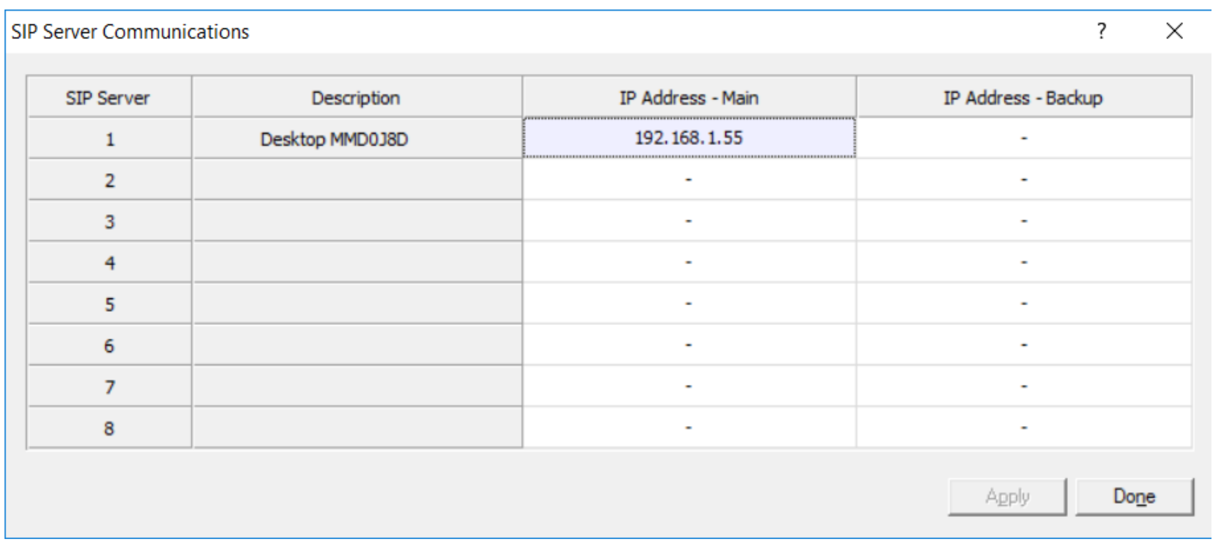


3. In the Description field, enter the **SIP Server Description**.
4. Click the **Done button**.
   The Edit Alpha / Description window disappears. The newly added description is visible next to the relevant SIP.



5. Click **Options | SIP Server Communications**.
   The SIP Server Communications window appears..





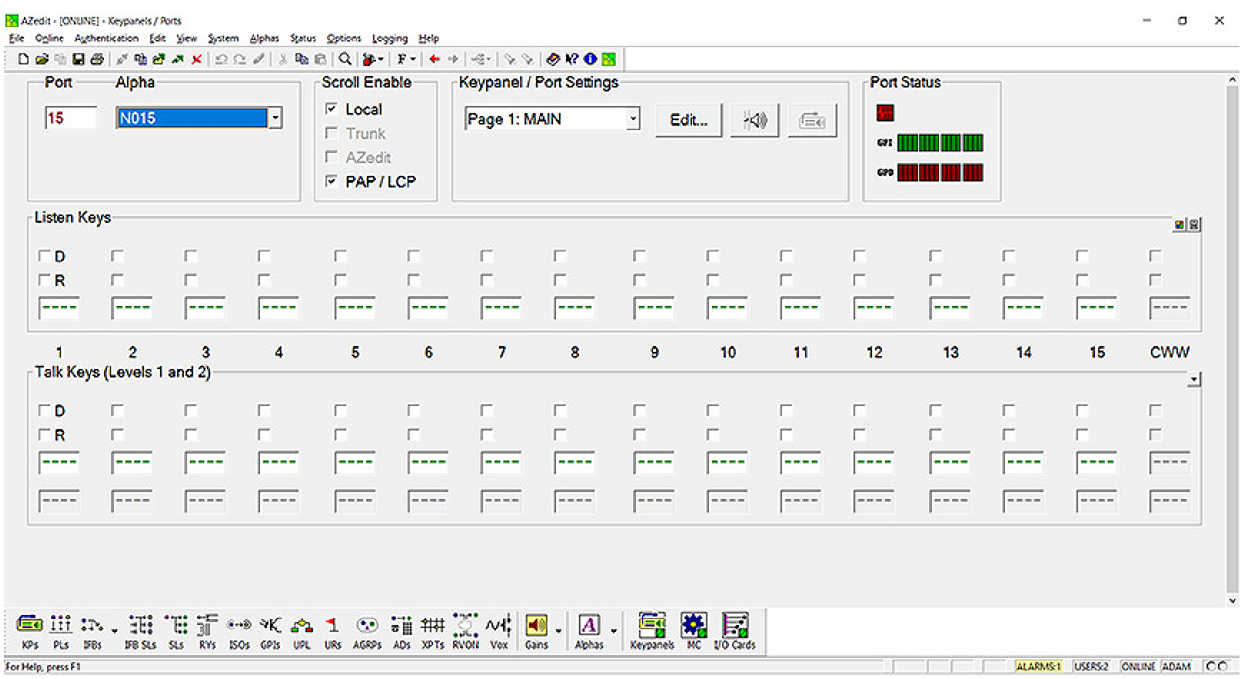
6. In the IP Address - Main field for the desired SIP Server, verify the correct **main IP Address for the VLINK SIP Server PC** is listed.
7. In the IP Address - Backup field for the desired SIP Server, enter the **backup IP Address for the VLINK SIP Server PC**, if applicable.
8. Click the **Done button**.

Intercom Port to SIP Line Association

To **assign each SIP phone line to an intercom port using AZedit software**, do the following:

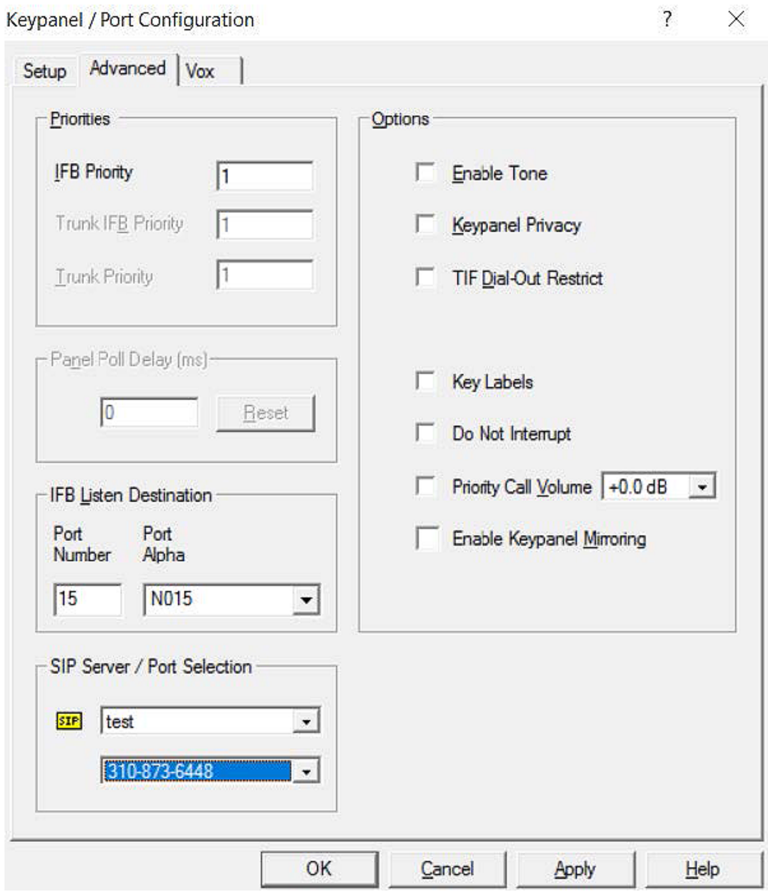
IMPORTANT: This must be consistent with the setup of the phone line connection to the Server 4-wire, and the internal Dante Virtual Soundcard channel to the Omneo intercom port channel.

1. On the Keypanels/Ports screen, in the Port field, enter the **number of the first port which you would like to assign to a SIP phone line**.



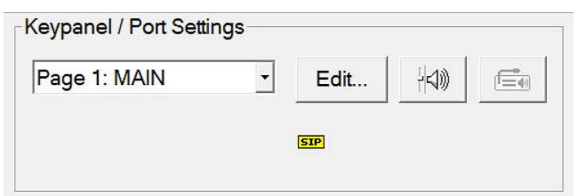
2. In the Keypanel / Port Settings group box, click the **Edit button**.
The Keypanel / Port Configuration window appears.

3. Click the **Advanced tab**.



4. In the SIP / Server / Port Selection group box, from the first pulldown menu, select the **SIP Server for the phone lines being configured**.
5. From the second pulldown menu, select the **number of the phone line to be associated with this intercom port**.
6. Click the **OK button**.
   The Keypanel / Port Configuration window disappears.

NOTE: On the Keypanels / Ports page, in the Keypanel / Port Settings group box, there is now a yellow "SIP" indicator box visible under the Edit button, if the process was completed successfully.



NOTE: Do this for each intercom port to be connected to each of the VLINK Server SIP phone lines.

CHAPTER 8

Testing the Completed System After Setup

To **prepare for testing the completed system**, verify the following:

- All systems are turned on and connected to the IP network.
- The VLINK Device Interface Application is running on the VLINK SIP Server PC.
- The Audinate Dante Virtual Soundcard (DVS) Application is running on the Audio Device Interface PC (NOT the Dante controller Program, which does not need to be running).

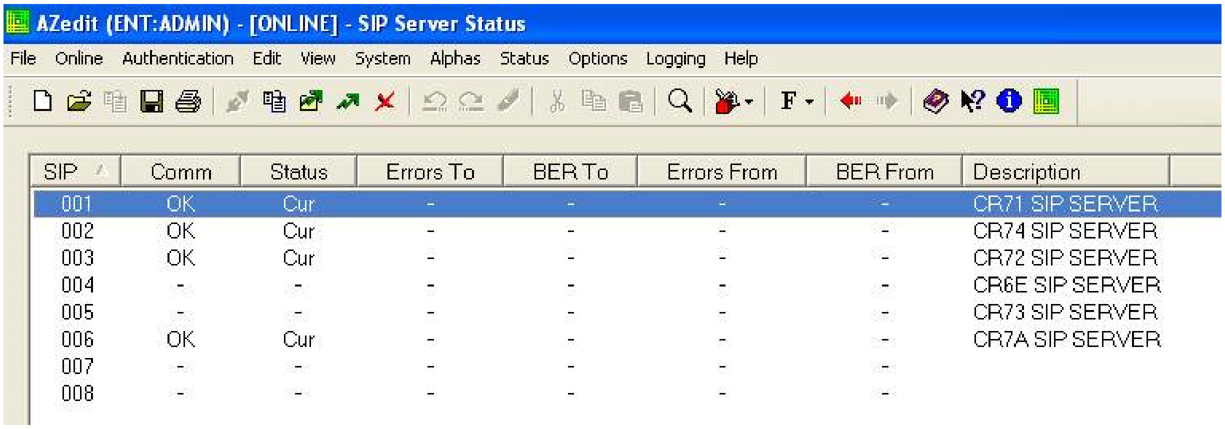
To **test the completed system**, do the following:

1. Open a **command (DOS) prompt on the VLINK SIP Server PC**.
2. One-by-one, ping the **IP addresses for the following** and verify a reply is received:
   - IP address of phone switch or SIP service provider network
   - Optional TIF Tally Display PC
   - ADAM primary master controller card
   - ADAM back-up master controller card
   - Dante Omneo card

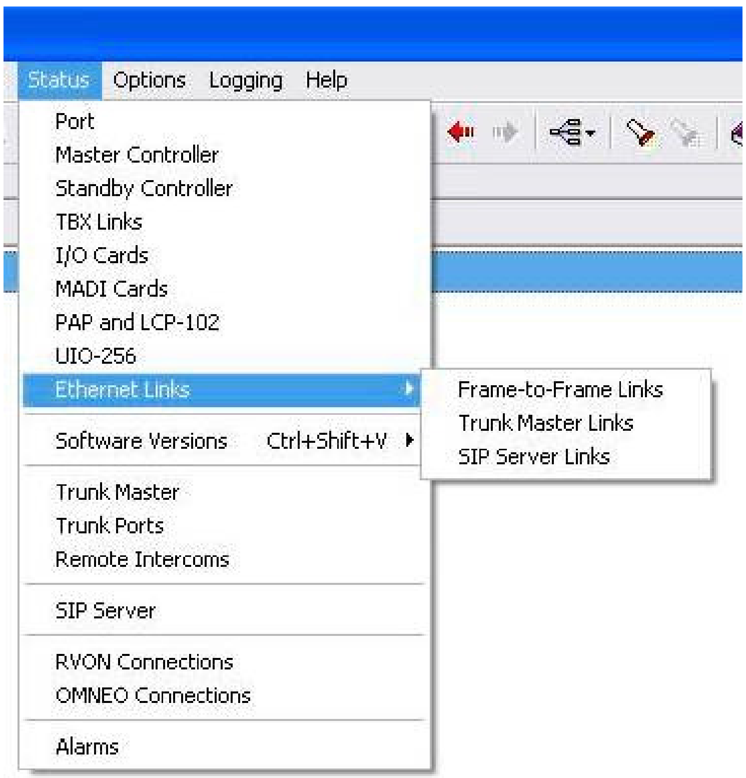
   NOTE: If any of the devices are not pingable, check the **IP network configurations and connections**.
3. On the VLINK SIP Server PC, using the System Administration program, click **client statistics**.
4. Verify the SIP line clients are shown as online. If not, check the configured SIP Provider / Phone switch settings and SIP login information for each line.
5. Also on the VLINK SIP Server PC, maximize the **Device Interface program**, which is running.
6. Verify 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 SIP Server PC. If necessary, click EDIT for each of the device interfaces and check the Dante Virtual Soundcard channel mapping.

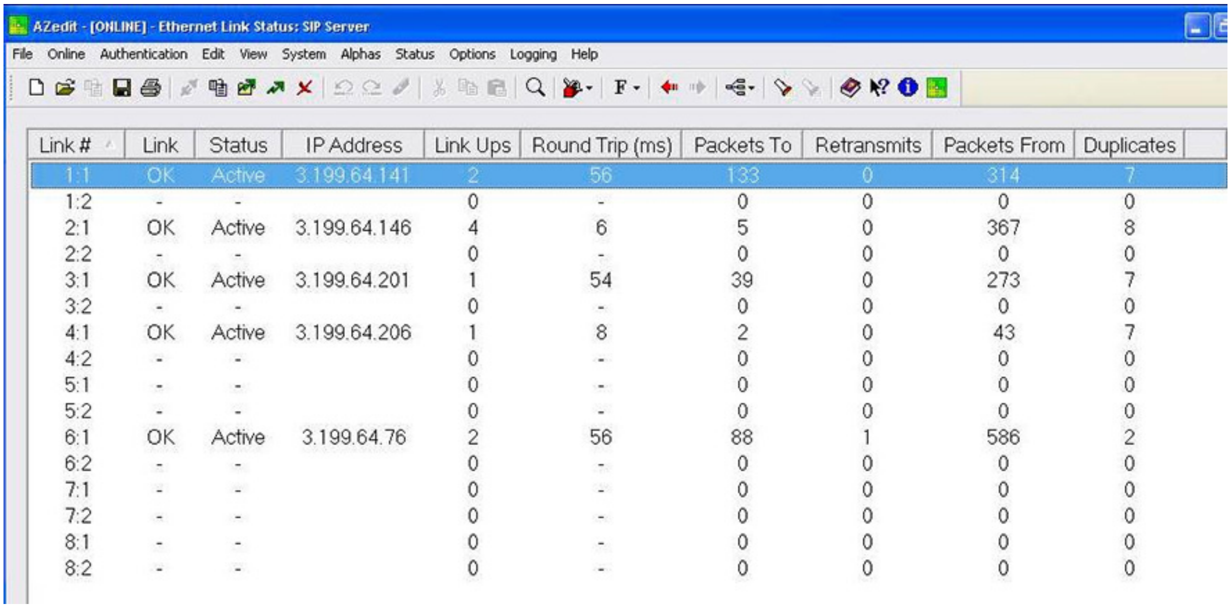
WARNING: Never exit or close the VLINK Device Interface Program. Always just minimize it when you are done looking at status.

7. Check the **SIP Server status in AZEDIT**.
When the VLINK Server is correctly connected for SIP Line Control, you can see the status under Status/SIP Server.



8. Check the more **detailed status information** 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

9. If status is not shown correctly, check the **IP address listed for the VLINK SIP Server PC** and check the **network connection between the VLINK SIP Server PC and the ADAM Master Controller** again.
10. Place **key assignments** on any RTS panel for the ports used for the SIP phone lines.
11. Dial into each line from an external phone and check the **functionality of auto-answer**, **manual answer by keypanel**, **and audio** in both directions.

RTS ADAM control of telephone line interfaces

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 FIF 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, do the following:

1. 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.
2. 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 KP-96 series keypanel**, 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 an RP1000 or KP series keypanel**, 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 widow. 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.

To **test the hanging up functionality**, do the following:

1. On the keypanel, press the **TIF key** down to turn it on.
   Auto Dial, Hang Up, Manual Dial, and Redial appear in the panel display.
2. Using the arrow buttons, select **Hang Up**.
3. Press the **SEL button**.
   The call is disconnected.

Notes:

Bosch Security Systems, Inc.
12000 Portland Avenue South
Burnsville, MN 55337 U.S.A.
www.boschcommunications.com