

RTS

Innovating the Future of Global Communications

VLink TIF Setup Guide

Bosch Security Systems
www.rtsintercoms.com



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INTRODUCTION

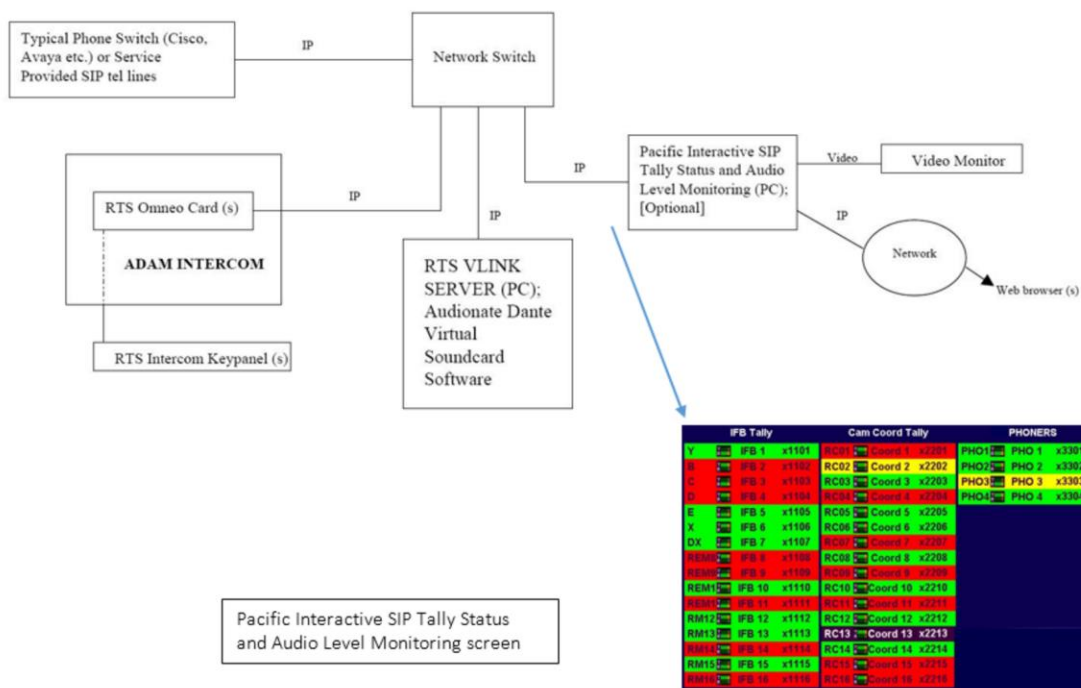
VLINK provides seamless integration between RTS ADAM intercoms and many compatible external communications methods. This document provides the basic outline of configurations required to use VLINK to connect ADAM to SIP phone systems and provide intelligent control and status of phone lines.

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.

A PC running a software package designed for line status descriptions, tally, and audio level metering 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



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.

Phone Switch / PBX Connect Method

There are two possibilities for the VLink TIF server to connect to a Phone Switch or PBX. The first is via a "Registered" connection where we use a username and password to register a SIP Client to the PBX and then receive and initiate calls via SIP Invites from/to the PBX in the same manner as a SIP client. The second is via a "Direct IP" or non-registered connection in which we only use the user name to receive and initiate calls via the PBX. In this case there is no registration or authentication but rather simply SIP Invites from/to the PBX. At some customer installations, this second method is used to install a "SIP Trunk" between the VLink TIF server and an offsite 3rd Party provider for the phone line access.

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 SIP signaling server. Also, all UDP ports must be open between VLink server and the RTP media servers.

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 SIP Server port licensing

Also before proceeding please ensure that minimally your VLINK Server is licensed for the necessary ports required for acting as a SIP Server. Additional VLINK port licenses are preferred to allow for local Server control panels used for testing, control, and diagnostics. Steps for licensing are detailed in the General VLink System Configuration section of this guide.

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 Audinate'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/products/software/dante-controller> and download and install the Dante Controller, 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.

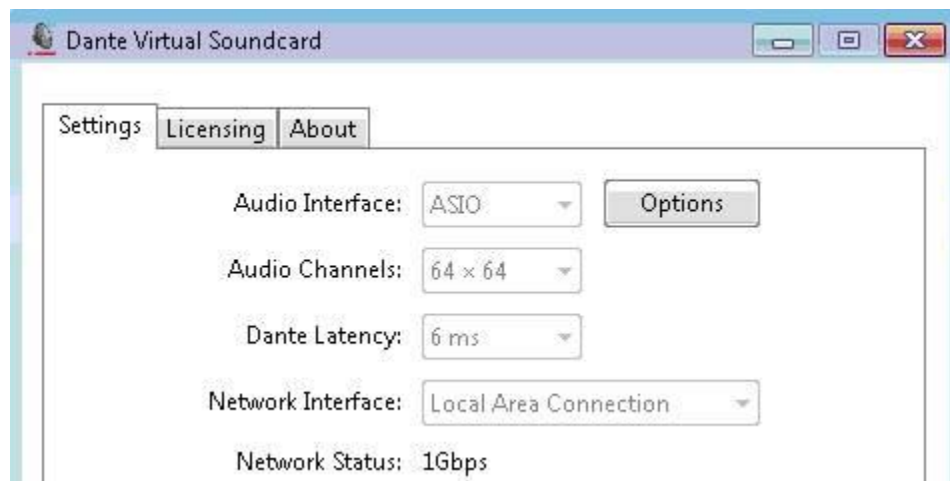
Once activated, select setup parameters as shown below:

Audio Interface = ASIO

Audio Channel = 64 x 64

Dante Latency = 6ms

Network Interface = (choose the appropriate Ethernet connection the PC uses)

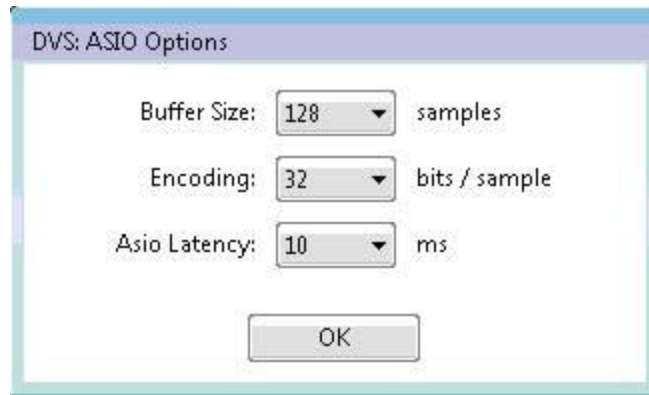


Next, click the options button and configure the fields as shown:

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

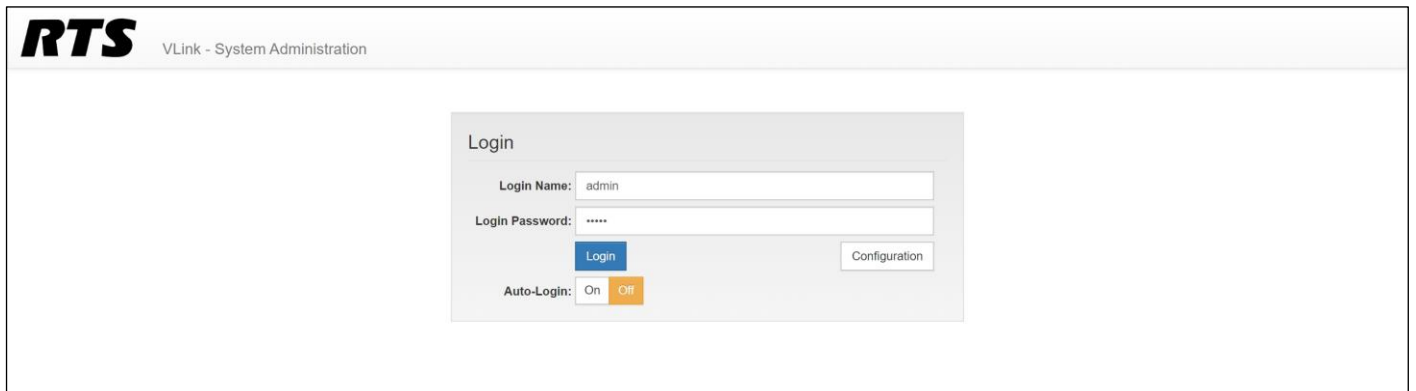
Based on the number of lines being interfaced, and the number of channels on the RTS OMNEO card, it is important at an early stage to clearly identify which phone line will connect to which intercom port. For example, if a 16 channel Omneo card is used, and it is installed as intercom ports 97-112, and 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 3.1.0 or later (3.4.1 or later preferred), and AZEDIT version 4.7.0 or later (5.1.0 or later preferred). Contact your RTS support representative should you require this software or assistance with upgrades.

VLINK SERVER CONFIG: VLINK SERVER TO SIP / PHONE SWITCH

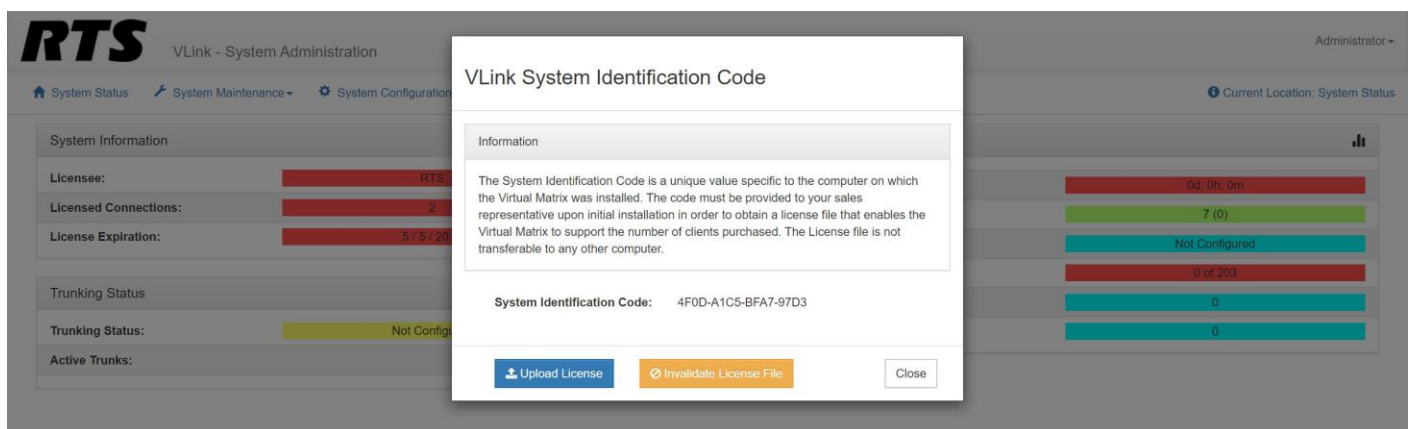
General VLINK System Configuration

Launch the VLink System Administration shortcut. At the Login screen enter your “Login Name” (default: “admin”) and “Login Password” (default: “admin”)



For detailed instructions on using the VLink System Administration application, and configuring your VLink system in general, refer to the VLink System Administration Guide:

<http://www.rtsintercoms.com/us/rts/file?i=98673&lg=eng>



If the system has not yet been licensed you will be greeted by this screen upon login (this menu can also be accessed by clicking “System Maintenance” then “License”). Send your System Identification Code to your RTS sales representative to obtain a license key file. Once you have the license file, click “Upload License” then select the file to license the system.

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.

Lines Show Inactive Lines +

Line #01 SIP
xxxx-xxxx-xx01
ON-HOOK

Status  Dial Plan Collapse

Connection to MC: Not Configured

Status  Dial Plan Collapse

Login Name:

Login Password:

Server IP Address (Local Network Interface)  

RTS Master Controller Main IP/Port

:

RTS Master Controller Backup IP/Port

:

PBX Connect Method (100/102 Registered) 

PBX Hostnames

Primary

Secondary

Disable Domain Authentication ON OFF

RTS Telephone Interface Network Settings

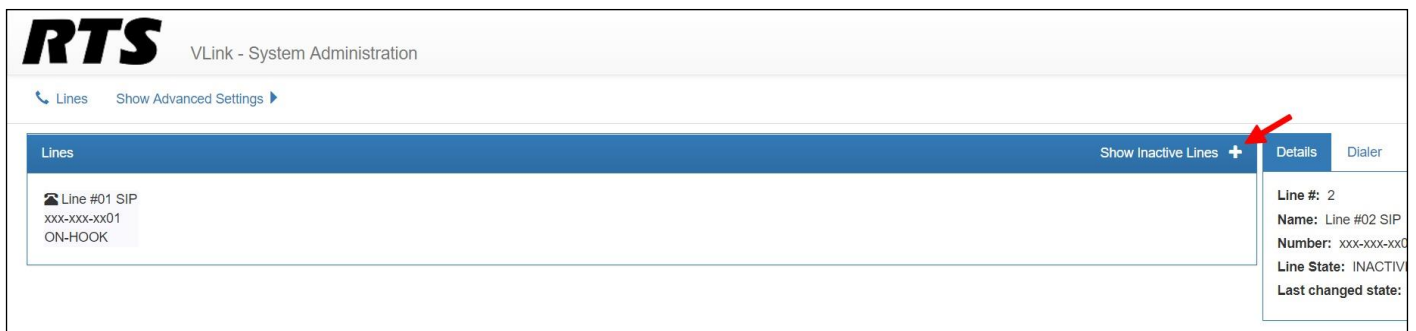
“Server IP address (Local Network Interface)” should be the IP address of this VLINK Server under most circumstances.

“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 “Show Inactive Lines” from the main System Administration Window. From the lines list, select a line and click the settings gear button.



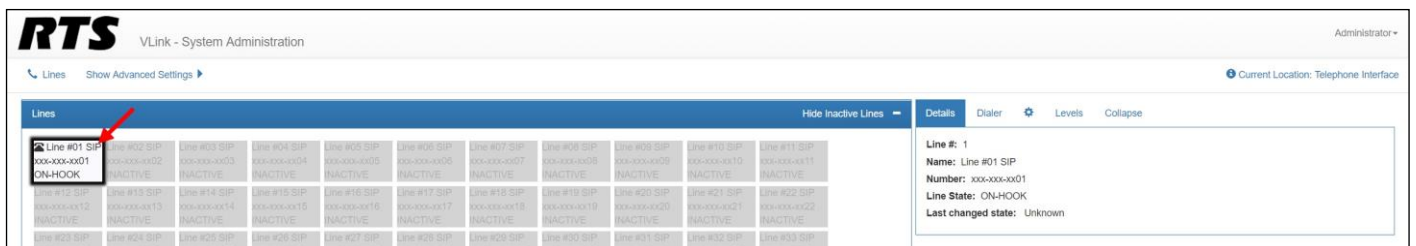
RTS VLink - System Administration

Lines Show Advanced Settings ▾

Lines Show Inactive Lines + Details Dialer

Line #01 SIP
xxx-xxx-xx01
ON-HOOK

Line #: 2
Name: Line #02 SIP
Number: xxx-xxx-xx02
Line State: INACTIVE
Last changed state:

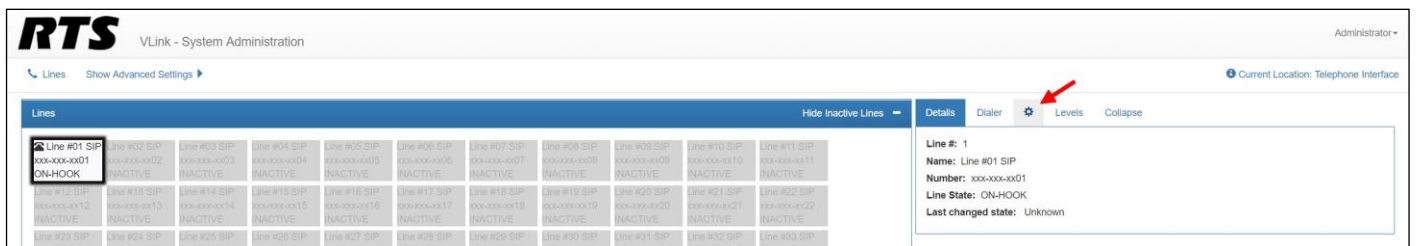


RTS VLink - System Administration Administrator

Lines Show Advanced Settings ▾ Hide Inactive Lines Details Dialer Levels Collapse

Line #01 SIP xxx-xxx-xx01 ON-HOOK	Line #02 SIP xxx-xxx-xx02 INACTIVE	Line #03 SIP xxx-xxx-xx03 INACTIVE	Line #04 SIP xxx-xxx-xx04 INACTIVE	Line #05 SIP xxx-xxx-xx05 INACTIVE	Line #06 SIP xxx-xxx-xx06 INACTIVE	Line #07 SIP xxx-xxx-xx07 INACTIVE	Line #08 SIP xxx-xxx-xx08 INACTIVE	Line #09 SIP xxx-xxx-xx09 INACTIVE	Line #10 SIP xxx-xxx-xx10 INACTIVE	Line #11 SIP xxx-xxx-xx11 INACTIVE
Line #12 SIP xxx-xxx-xx12 INACTIVE	Line #13 SIP xxx-xxx-xx13 INACTIVE	Line #14 SIP xxx-xxx-xx14 INACTIVE	Line #15 SIP xxx-xxx-xx15 INACTIVE	Line #16 SIP xxx-xxx-xx16 INACTIVE	Line #17 SIP xxx-xxx-xx17 INACTIVE	Line #18 SIP xxx-xxx-xx18 INACTIVE	Line #19 SIP xxx-xxx-xx19 INACTIVE	Line #20 SIP xxx-xxx-xx20 INACTIVE	Line #21 SIP xxx-xxx-xx21 INACTIVE	Line #22 SIP xxx-xxx-xx22 INACTIVE
Line #23 SIP xxx-xxx-xx23 INACTIVE	Line #24 SIP xxx-xxx-xx24 INACTIVE	Line #25 SIP xxx-xxx-xx25 INACTIVE	Line #26 SIP xxx-xxx-xx26 INACTIVE	Line #27 SIP xxx-xxx-xx27 INACTIVE	Line #28 SIP xxx-xxx-xx28 INACTIVE	Line #29 SIP xxx-xxx-xx29 INACTIVE	Line #30 SIP xxx-xxx-xx30 INACTIVE	Line #31 SIP xxx-xxx-xx31 INACTIVE	Line #32 SIP xxx-xxx-xx32 INACTIVE	Line #33 SIP xxx-xxx-xx33 INACTIVE

Line #: 1
Name: Line #01 SIP
Number: xxx-xxx-xx01
Line State: ON-HOOK
Last changed state: Unknown



RTS VLink - System Administration Administrator

Lines Show Advanced Settings ▾ Hide Inactive Lines Details Dialer Levels Collapse

Line #01 SIP xxx-xxx-xx01 ON-HOOK	Line #02 SIP xxx-xxx-xx02 INACTIVE	Line #03 SIP xxx-xxx-xx03 INACTIVE	Line #04 SIP xxx-xxx-xx04 INACTIVE	Line #05 SIP xxx-xxx-xx05 INACTIVE	Line #06 SIP xxx-xxx-xx06 INACTIVE	Line #07 SIP xxx-xxx-xx07 INACTIVE	Line #08 SIP xxx-xxx-xx08 INACTIVE	Line #09 SIP xxx-xxx-xx09 INACTIVE	Line #10 SIP xxx-xxx-xx10 INACTIVE	Line #11 SIP xxx-xxx-xx11 INACTIVE
Line #12 SIP xxx-xxx-xx12 INACTIVE	Line #13 SIP xxx-xxx-xx13 INACTIVE	Line #14 SIP xxx-xxx-xx14 INACTIVE	Line #15 SIP xxx-xxx-xx15 INACTIVE	Line #16 SIP xxx-xxx-xx16 INACTIVE	Line #17 SIP xxx-xxx-xx17 INACTIVE	Line #18 SIP xxx-xxx-xx18 INACTIVE	Line #19 SIP xxx-xxx-xx19 INACTIVE	Line #20 SIP xxx-xxx-xx20 INACTIVE	Line #21 SIP xxx-xxx-xx21 INACTIVE	Line #22 SIP xxx-xxx-xx22 INACTIVE
Line #23 SIP xxx-xxx-xx23 INACTIVE	Line #24 SIP xxx-xxx-xx24 INACTIVE	Line #25 SIP xxx-xxx-xx25 INACTIVE	Line #26 SIP xxx-xxx-xx26 INACTIVE	Line #27 SIP xxx-xxx-xx27 INACTIVE	Line #28 SIP xxx-xxx-xx28 INACTIVE	Line #29 SIP xxx-xxx-xx29 INACTIVE	Line #30 SIP xxx-xxx-xx30 INACTIVE	Line #31 SIP xxx-xxx-xx31 INACTIVE	Line #32 SIP xxx-xxx-xx32 INACTIVE	Line #33 SIP xxx-xxx-xx33 INACTIVE

Line #: 1
Name: Line #01 SIP
Number: xxx-xxx-xx01
Line State: ON-HOOK
Last changed state: Unknown

In the “Line Name” section enter the phone number. The “Phone Number Section” will be the name that will appear on the VLink Control Panel selector, and also become the default shown on the optional Pacific Interactive TIF Tally Display. Enter the “Sip User Name” and “SIP User Password”.

Important Note: The SIP User Name and Password should be that provided by your SIP phone line provider for that particular line. Click "OK".

Important note: The "Line Name" 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 Phone Number will be the default that is displayed on the optional Pacific Interactive Status Display System, however you will also be able to explicitly assign the specific ADAM type and function number (ie. IFB, # 1) as the alpha that will be displayed on a line-by-line basis.

Current Location: Telephone Interface

DetailsDialer⚙LevelsCollapse

Line State: ON OFF

Line Name:

Phone Number:

SIP User Name:

SIP User Password:

Auto Answer: ON OFF

Auto Answer Sound File:

Upload new Answer Sound File: No file chosen

Ringtone Sound File:

Upload new Ringtone: No file chosen

RTS Function Type

RTS Function Number

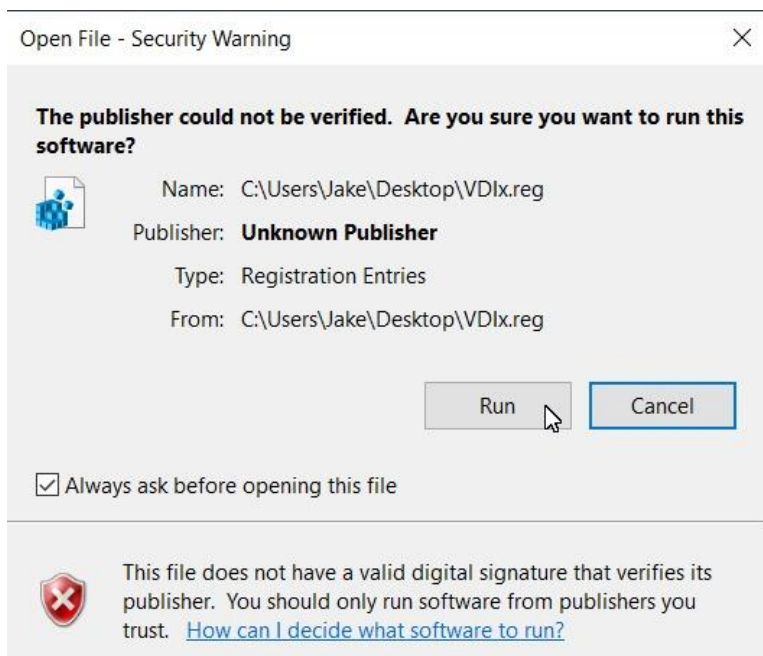
RTS Alpha

Internal Server Dante Virtual Soundcard Channel Connections

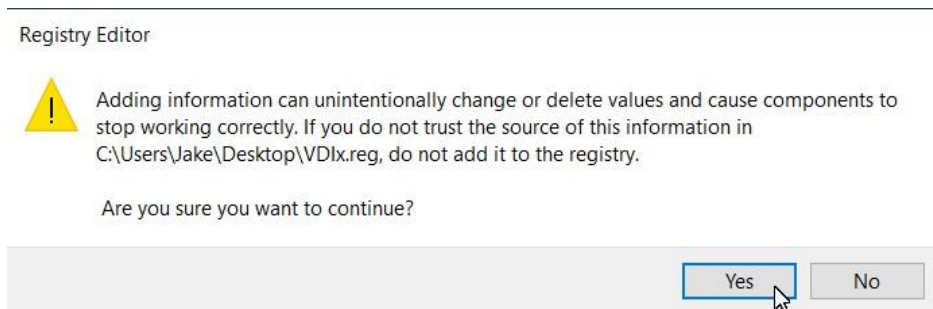
Open the VLINK Device Interface Application. This application would have been installed as part of the original VLINK Server install. If needed for detailed instructions on how to install and use the VLINK Device Interface application refer to the VLINK Device Interface application User Guide which can be found here: https://s3.amazonaws.com/rtsc/downloads/VLink_Releases_Historical/171214_v500_VLink_Device_Interface.zip

Download the VLINK Device Interface configuration registry file: https://drive.google.com/open?id=1laxl28HbeVvNGFAIA8AcwzvwHzDI_AhY. Once the registry edit is made VLINK Device Interface will be automatically populated with the necessary 4-Wire connections.

Double-click the registry file. When prompted, click "Run".



Click "Yes".



Launch VLINK Device Interface. Enter in your VLINK Server IP address in the Virtual Matrix IP Address field and click "Apply".

VLink Device Interface Configuration

Network Settings

Device Interface IP Address: 192.168.2.107

Virtual Matrix IP Address: 192.168.2.107 : 1000

Monitor Device

Select Speaker

Logic Input/Output

Select GPIO Device #1: (None)

Select GPIO Device #2: (None)

Select GPIO Device #3: (None)

Apply

You will see each client log in and turn green as shown below.

Login Name	Talk/Listen Name	Status	Monitor	GPI #1	GPI #2	GPO #1	GPO #2
4wire17	CAM #01 4w	Connected					
4wire18	CAM #02 4w	Connected					
4wire19	CAM #03 4w	Connected					
4wire20	CAM #04 4w	Connected					
4wire21	CAM #05 4w	Connected					
4wire22	CAM #06 4w	Connected					
4wire23	CAM #07 4w	Connected					
4wire24	CAM #08 4w	Connected					
4wire25	CAM #09 4w	Connected					
4wire26	CAM #10 4w	Connected					
4wire27	CAM #11 4w	Connected					
4wire28	CAM #12 4w	Connected					
4wire29	CAM #13 4w	Connected					
4wire30	CAM #14 4w	Connected					
4wire31	CAM #15 4w	Connected					
4wire32	CAM #16 4w	Connected					
4wire33	Phoner #01 4w	Connected					
4wire34	Phoner #02 4w	Connected					
4wire35	Phoner #03 4w	Connected					
4wire36	Phoner #04 4w	Connected					

Selected Device(s)

LOGIN / LOGOUT Monitor Input Monitor Output Simulate GPI(s) Toggle GPO(s) Statistics...

Login Automatically

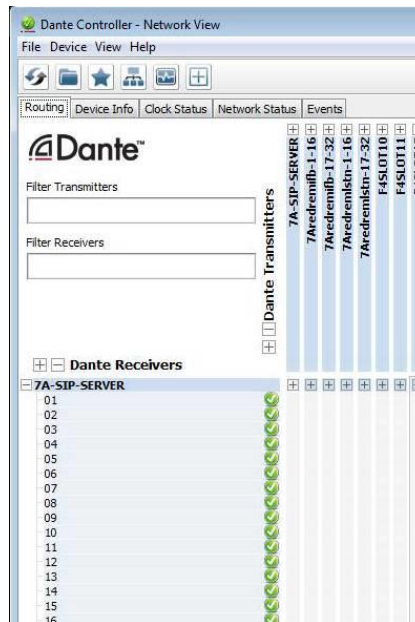
All Devices

LOGIN / LOGOUT Configure...

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.



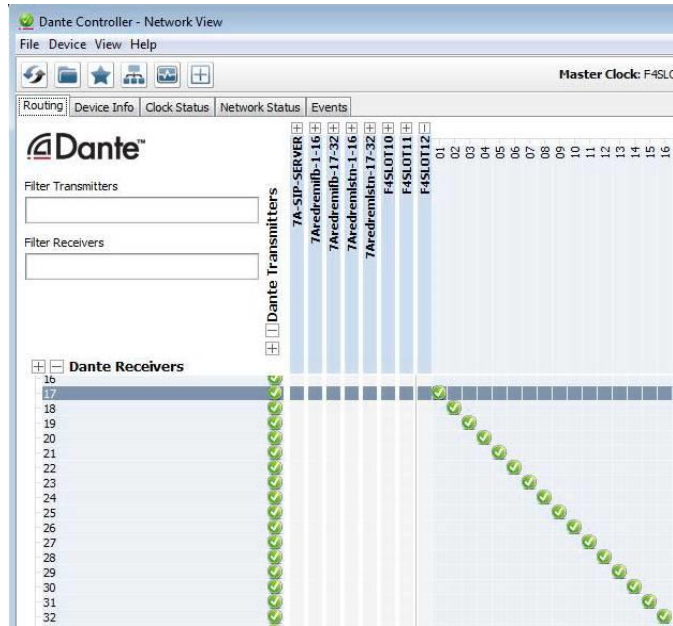
Click the + next to the VLINK Server PC name on the left.

Click the + next to the RTS Omneo Card name on the top.

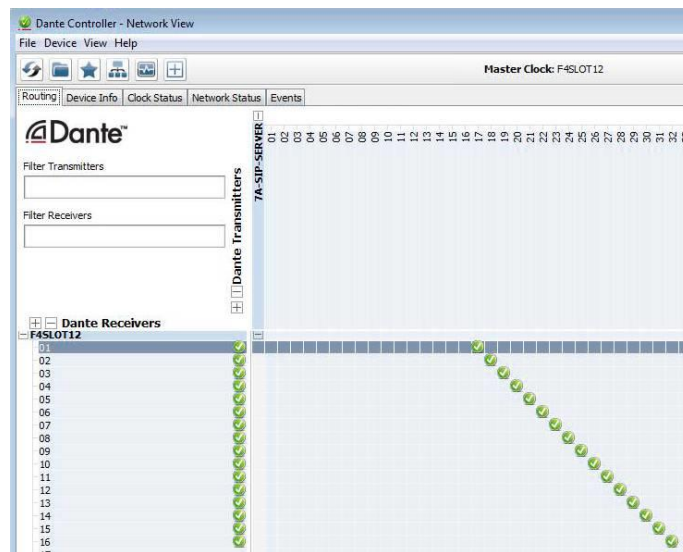
Click at the crosshairs to create an audio connection (indicated by a green circle) in a diagonal so each VLINK Server channel is mapped in one direction to the associated Omneo card channel.

Click the – next to the VLINK Sever PC name on the left.

Click the – next to the RTS Omneo 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 Omneo Card name at the top.

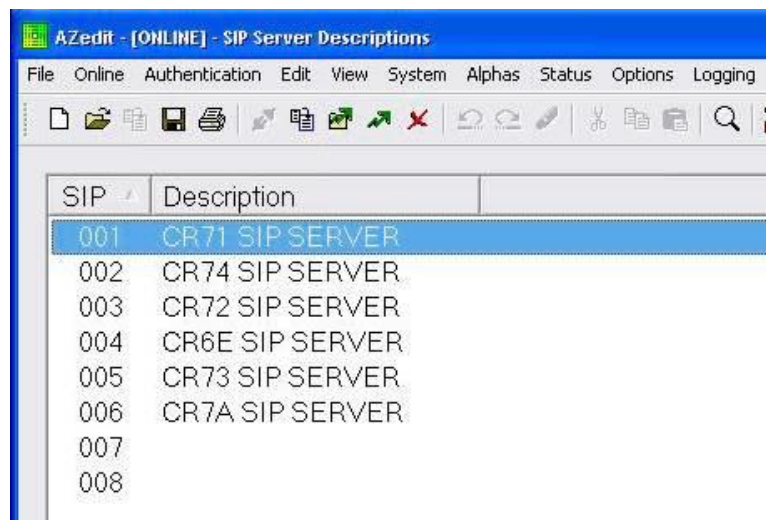
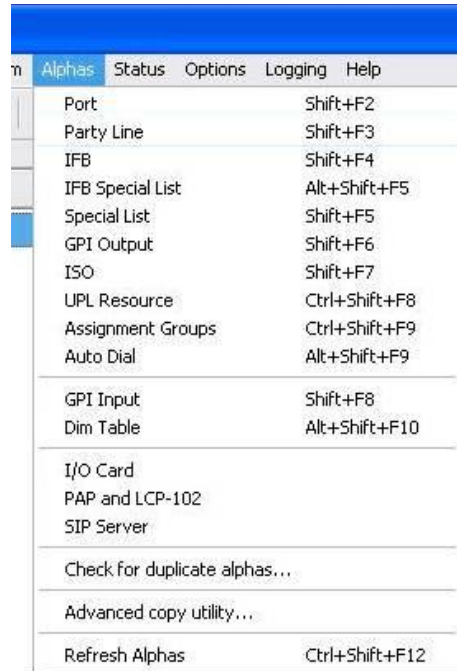


Exit the Dante Controller software program.

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.



The screenshot shows a dialog box titled "SIP Server Communications" with a table containing 8 rows of data. The columns are "SIP Server", "Description", "IP Address - Main", and "IP Address - Backup".

SIP Server	Description	IP Address - Main	IP Address - Backup
1	CR71 SIP SERVER	3.199.64.141	-
2	CR74 SIP SERVER	3.199.64.146	-
3	CR72 SIP SERVER	3.199.64.201	-
4	CR6E SIP SERVER	3.199.64.206	-
5	CR73 SIP SERVER	-	-
6	CR7A SIP SERVER	3.199.64.76	-
7		-	-
8		-	-

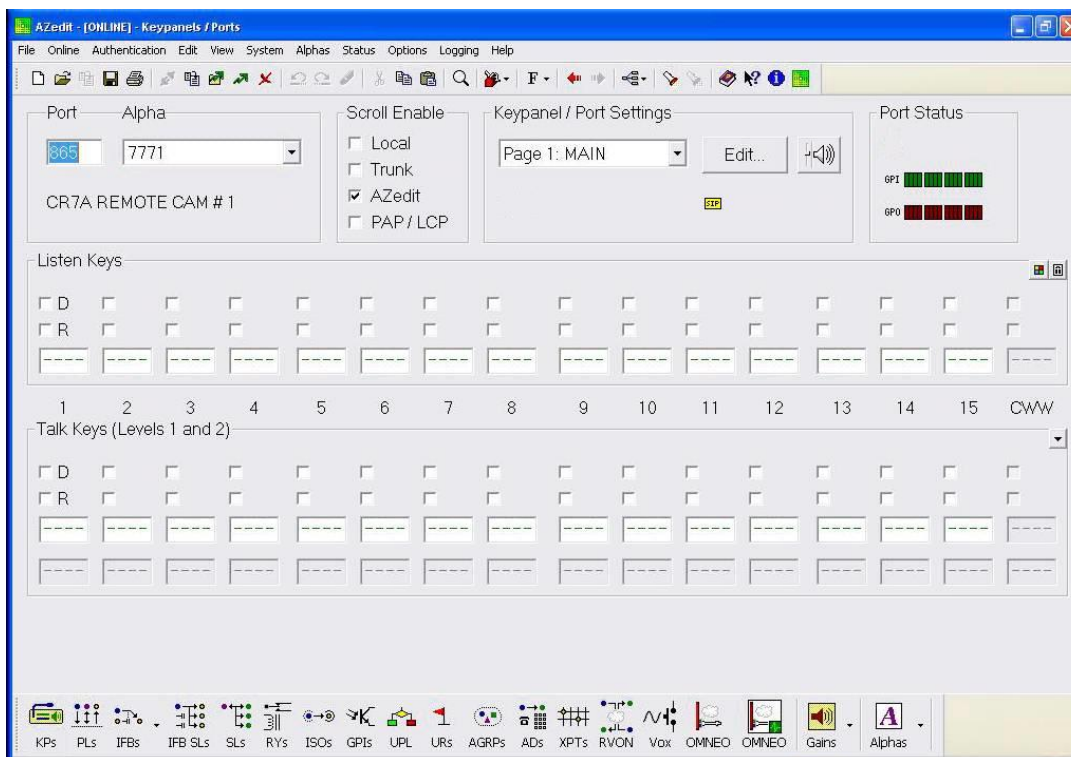
Buttons for "Apply" and "Done" are visible at the bottom right of the dialog box.

INTERCOM PORT TO SIP LINE ASSOCIATION

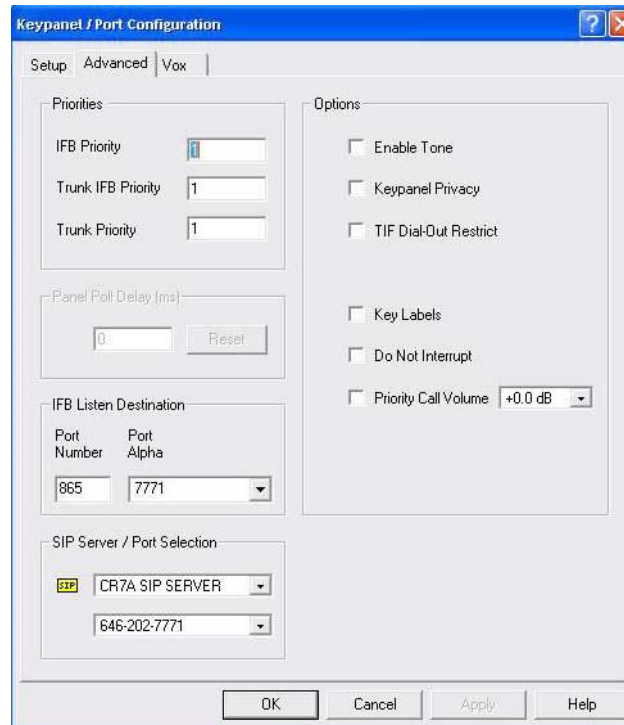
Next you must tell the intercom which intercom port is connected via audio to which VLINK device interface channel / SIP Phone Line. This must be consistent with how you have designed the phone line connection to the Server 4-wire, and the internal Dante Virtual Soundcard channel to the Omneo intercom 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.

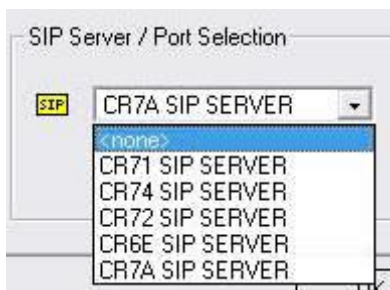
From the keypanel setup screen for the first SIP phone line port, Click the EDIT button.



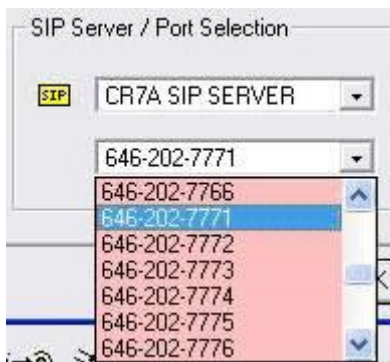
Click the Advanced Tab:



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.

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.

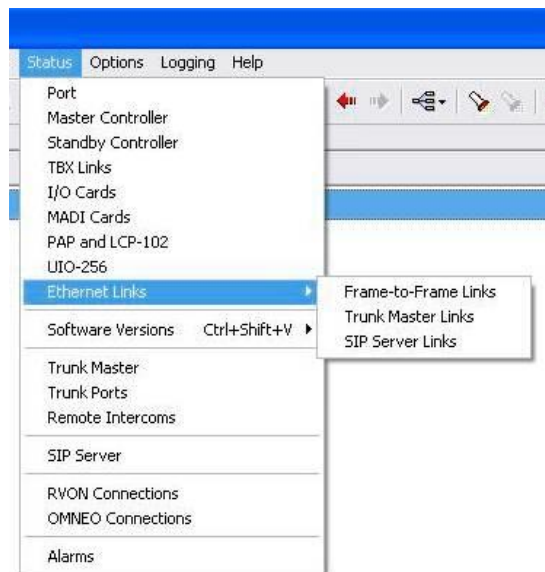
When the VLINK SIP SERVER is correctly connected for SIP Line Control, you can see the status under Status/SIP Server as shown below:

AZedit (ENT:ADMIN) - [ONLINE] - SIP Server Status

File Online Authentication Edit View System Alphas Status Options Logging Help

SIP	Comm	Status	Errors To	BER To	Errors From	BER From	Description
001	OK	Cur	-	-	-	-	CR71 SIP SERVER
002	OK	Cur	-	-	-	-	CR74 SIP SERVER
003	OK	Cur	-	-	-	-	CR72 SIP SERVER
004	-	-	-	-	-	-	CR6E SIP SERVER
005	-	-	-	-	-	-	CR73 SIP SERVER
006	OK	Cur	-	-	-	-	CR7A SIP SERVER
007	-	-	-	-	-	-	
008	-	-	-	-	-	-	

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



AZedit - [ONLINE] - Ethernet Link Status: SIP Server

File Online Authentication Edit View System Alphas Status Options Logging Help

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 IP address 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 keypad, 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 keypad 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 keypad 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 keypad 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.