

Innovating the Future of Global Communications

VLINK SIP Setup Guide

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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

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DO NOT OPE CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE SERVICEABLE PARTS



THE EXCLAMATION POINT WITHIN THE TRIANGLE IS A WARNING SIGN ALERTING YOU OF IMPORTANT INSTRUCTIONS ACCOMPANYING THE 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.

\sim	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.



RTS SIP INTERFACE SOLUTION (Dante) BLOCK DIAGRAM

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

6 Introduction

chapter 2 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 = 64×64
 - Dante Latency = 6ms
 - Network Interface = Choose the appropriate Ethernet connection the PC uses

ettings	Licensing Abou	t				
	Audio Inter	rface:	OIZA	-	Options	
	Audio Char	nnels:	64 × 64	*		
	Dante Lat	ency:	6 ms	*		
	Network Inter	rface:	Local Are	a Conn	ection 👻	
	Network St	tatus:	1Gbps			

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

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).

WLink System Administration Login	×
Login Name: admin Login Password:	LOGIN
Configure	Login Automatically

- 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

VLink System Administration Configuration					
Network Settings System Configuration IP Address Virtual Matrix IP Address	10.211.55.3 ··································				
Use Domain Authentication					
	Apply				

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.*

System Information		System Configuration	
Licensee:	IntraCom	System Settings	
Licensed Connections:	1000	Client Configuration	
License Expiration:		Group Configuration	
System Status		Remote Configuration	
System Up Time:	2d, 1h, 41m	User Interface Settings	
Processor Utilization:	1 (0)		
Failover Status:	PRIMARY ACTIVE Secondary Off-line	System Maintenance Restart System	
Active Connections:	3 of 303	Force Failover	
Active Audio Inputs:	2	System Information	
Active Audio Outputs:	1	Client Statistics	
Trunking Status		SIP Registrations	
Trunking Status:	Not Licensed	Activity Log	
Active Trunks:		Reports	

 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.

Master System Administrator Login Login Name:	admin	_							
Login Password:	1ntr@c0r	ток							
Primary Server Network Settings Server IP Address (Local Network Interface)	74.208.20)1.229						Secondary (Failover) Server Network Settings Server IP Address	52.8.255
Server NAT IP Address	74.208.20	74.208.201.229						Server NAT IP Address	52.8.255.16
Server IP Ports for Client Data / Audio	1000 / 5060 /		1001					Server IP Port for VCOM Client Data / Audio	1000
Server IP Ports for SIP Data / RTP Audio Base			0					Server IP Port for Failover Data	1001
Server SIP Domain Name	intracom	systen	ns.net						
Disable Domain Authentication									
Trunking Network Settings								Telephone Interface Network Settings	
Server IP Address (Local Network Interface)	1.1.1.1							Server IP Address (Local Network Interface)	
RTS Trunk Master Main / Backup IP Address	1.1.1.2			1	1	.3		RTS Master Controller Main / Backup IP	
RTS Trunk Master Main / Backup IP Port	27415			1	2	5		RTS Master Controller Main / Backup IP Port	
Ignore 8 Character Unicode Alphas									
Audio Settings									
Audio Mix Sample Rate	Full Band	(48 K	Hz)				~		
Audio Output Level Gain (Post-Mix)	0 dB								

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.

Master System Administrator Login admin Login Name: admin Login Password:	VLink System Settings	×
Login Name: admin Login Password:	Master System Administrator Login	
Login Password: Primary Server Network Settings Server IP Address (Local Network Interface) Server IP Ports for Client Data / Audio 1000 Server IP Ports for Client Data / Audio Server IP Ports for SIP Data / RTP Audio Base Server SIP Domain Name Disable Domain Authentication Secondary (Fallover) Server Network Settings Server IP Address Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Address Server IP Address Server IP Address (Local Network Interface) Master Controller Main / Backup IP Port USE only 4 character alphas Audio Settings Audio Settings Audio Settings Failover Settings Failover Settings Failover Activation Delay 0 (seconds) Automatic Failback Never. Failback done manually (hth:r Geo Mapping Server Network Settings <td>Login Name:</td> <td>admin</td>	Login Name:	admin
Primary Server Network Settings Server IP Address (Local Network Interface) Server NAT IP Address Server IP Ports for Client Data / Audio Server IP Ports for SIP Data / RTP Audio Base Server SIP Domain Name Disable Domain Authentication Secondary (Failover) Server Network Settings Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for VCOM Client Data / Audio Server IP Port for Failover Data 1001 RTS Telephone Interface Network Settings Server IP Address (Local Network Interface) Master Controller Main / Backup IP Port USS entings Audio Settings Audio Settings Audio Output Level Gain (Post-Mix) Voice Activity Indication Color Voice Activity Indication Color Failover Activation Delay 0 (seconds) Automatic Failback Never. Failback done manually (hth:r Geo Mapping Server Network Settings Geo Mapping Server Network Settings	Login Password:	
Server IP Address (Local Network Interface) 192.168.1.55 Server NAT IP Address 1000 Server IP Ports for Client Data / Audio 1000 Server IP Ports for SIP Data / RTP Audio Base 5060 Server SIP Domain Name 0 Disable Domain Authentication	Primary Server Network Settings	
Server NAT IP Address Server IP Ports for Client Data / Audio 1000 Server IP Ports for SIP Data / RTP Audio Base Sofo Server SIP Domain Name Disable Domain Authentication Secondary (Failover) Server Network Settings Server IP Address Server IP Address Server IP Address Server IP Port for VCOM Client Data / Audio Server IP Port for Failover Data 1001 RTS Telephone Interface Network Settings Server IP Address Server IP Address (Local Network Interface) Master Controller Main / Backup IP Address Master Controller Main / Backup IP Port 27415 / Use only 4 character alphas Audio Settings Audio Output Level Gain (Post-Mix) 0 dB Voice Activity Indication Color Text Background Failover Activation Delay 0 (seconds) Automatic Failback Never. Failback done manualty (thir:	Server IP Address (Local Network Interface)	192.168.1.55
Server IP Ports for Client Data / Audio 1000 / 1000 Server IP Ports for SIP Data / RTP Audio Base 5060 / 0 Server SIP Domain Name	Server NAT IP Address	
Server IP Ports for SIP Data / RTP Audio Base 5060 / 0 Server SIP Domain Name	Server IP Ports for Client Data / Audio	1000 / 1000
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Use only 4 character alphas Use only 4 character alphas Image: Construction of the second	Master Controller Main / Backup IP Port	27415 / 27415
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Geo Mapping Server IP Address	Geo Mapping Server Network Settings	
	Geo Mapping Server IP Address	
Geo Mapping Server IP Port	Geo Mapping Server IP Port	
OK Cancel	ОК	Cancel

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*.

	Tune	Talk Listen Name	Logio Namo	Logia Degeword	Description		Auda Cattings	Ontione Templai			1
-	Type	Taik/Listen Name	Login ivame	Login Password	Description		Audio Settings	Options Templa	te		
SIP	Direct IP Trunk	CAM #01	17771		645-202-77	71	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #02	17772		646-202-77	72	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #03	17773		646-202-77	73	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #04	17774		646-202-77	74	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #05	17775		646-202-77	75	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #06	17776		646-202-77	76	CAM #01	CAM #01			Add
SIP	Direct IP Trunk	CAM #07	17777		646-202-77	77	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #08	17778		646-202-77	78	CAM #01	CAM #01		_	Edit
SIP	Direct IP Trunk	CAM #09	17779		646-202-77	79	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #10	17780		646-202-77	80	CAM #01	CAM #01		11	Delet
SIP	Direct IP Trunk	CAM #11	17781		646-202-77	81	CAM #01	CAM #01			(auto
SIP	Direct IP Trunk	CAM #12	17782		646-202-77	82	CAM #01	CAM #01			Dupic
SIP	Direct IP Trunk	CAM #13	17783		646-202-77	83	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #14	17784		646-202-77	84	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #15	17785		646-202-77	85	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #16	17786	-	646-202-77	86	CAM #01	CAM #01		٣	
Select	ed Client					Templates					
S	elector Assignments	Audio Settin	gs	Options		Link Selector Assignments	Link Audio Se	ettings	Link Options		
						Unlink Selector Assignments	Unlink Audio S	ettinos	Unlink Options		

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

VLink Client Configuration Add/Edit				×
Client Identification Client Type	SIP Devic	e: Intersystem,	/PBX Direct IP Trunk	~
Client Description	555-360-4	4456		
Login Name:	29992		Allow Anonymous Login	
Login Password:			Use Domain Authentication	
Selector Talk/Listen Name:	LINE #01	l		
Selector Listen Only Name:				
External Alpha (8U characters)				
External Alpha (4 characters)				
Options Disable Client Login / Connection Always Show Selector when Off-line Latch Disable Talk Selector Party Line Operation IFB Destination ISO Destination Allow Assigment To Multiple Party Line: Standard System Administration Privilege	s	Selector Assi No Local As No Local As No Remote No Remote	gnment Restrictions signment By Administrator signment By User Assignment By Administrator Assignment By User	
	OK	Cancel		

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.

- 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.

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.

- 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.

Talk/Listen N ct IP Trunk SSS5 -Wire Interface IFB #01.4w -Wire Interface IFB #03.4w -Wire Interface IFB #04.4w -Wire Interface IFB #04.4w -Wire Interface IFB #05.4w -Wire Interface IFB #05.4w -Wire Interface IFB #05.4w -Wire Interface IFB #05.4w	Listen Only N Lo 12 44 45 45 45 45 45 45 45 45 45 45 45 45	ogin Name 23456 wire01 wire02 wire03 wire04 wire05	Login Passw password	Description SSS-SSS-SSSS	RTS Alpha	Port 39 101 102 103	Audio Settin IFB #01 4w IFB #01 4w	Options Tem	^
ct IP Trunk 5555 -Wire Interface IFB #01.4w -Wire Interface IFB #02.4w -Wire Interface IFB #04.4w -Wire Interface IFB #04.4w -Wire Interface IFB #05.4w -Wire Interface IFB #05.4w -Wire Interface IFB #05.4w -Wire Interface IFB #05.4w -Wire Interface IFB #07.4w	12 44 44 44 44 44 44 44	23456 wire01 wire02 wire03 wire04 wire05	password	555-555-5555		39 101 102 103	IFB #01 4w IFB #01 4w		
-Wire Interface IF8 #01 4w -Wire Interface IF8 #02 4w -Wire Interface IF8 #03 4w -Wire Interface IF8 #04 4w -Wire Interface IF8 #05 4w -Wire Interface IF8 #06 4w -Wire Interface IF8 #07 4w	4) 4) 4) 4) 4) 4) 4) 4) 4) 4)	wire01 wire02 wire03 wire04 wire05				101 102 103	IFB #01 4w IFB #01 4w		
-Wire Interface IF8 #02 4w -Wire Interface IF8 #03 4w -Wire Interface IF8 #04 4w -Wire Interface IF8 #05 4w -Wire Interface IF8 #06 4w -Wire Interface IF8 #07 4w	4) 4) 4) 4) 4) 4) 4)	wire02 wire03 wire04 wire05				102 103	IFB #01 4w		
-Wire Interface IF8 #03 4w -Wire Interface IF8 #04 4w -Wire Interface IF8 #05 4w -Wire Interface IF8 #06 4w -Wire Interface IF8 #07 4w	41 41 41 41 41	wire03 wire04 wire05				103	100 #01 day		
-Wire Interface IF8 #04 4w -Wire Interface IF8 #05 4w -Wire Interface IF8 #06 4w -Wire Interface IF8 #07 4w	41	wire04 wire05					1-0 ±01 4M		
-Wire Interface IFB #05 4w -Wire Interface IFB #06 4w -Wire Interface IFB #07 4w	41	wire05				104	IFB #01 4w		
-Wire Interface IF8 #06 4w -Wire Interface IF8 #07 4w	4					105	IFB #01 4w		
-Wire Interface IFB #07 4w	4.	wire06				106	IF8 #01 4w		
		wire07				107	IFB #01 4w		
-Wire Interface IFB #08 4w	41	wire08				108	IFB #01 4w		
-Wire Interface IFB #09 4w	41	wire09				109	IFB #01 4w		
-Wire Interface IFB #10 4w	41	wire10				110	IFB #01 4w		
-Wire Interface IFB #11 4w	41	wire11				111	IFB #01 4w		
-Wire Interface IFB #12 4w	41	wire12				112	IFB #01 4w		
-Wire Interface IFB #13 4w	41	wire13				113	IFB #01 4w		
-Wire Interface IFB #14 4w	41	wire14				114	IFB #01 4w		
-Wire Interface IFB #15 4w	41	wire15				115	IFB #01 4w		~
								>	
ent				Templates					
	o Settinos		Options	Link Selector Assi	ionments Link Au	udio Sett	inas	Link Options	
-Wir -Wir	e Interface IFB #14 4w e Interface IFB #15 4w	e Interface IFB #14 4w 4 e Interface IFB #15 4w 4 signments Audio Settings	e Interface IFB #14 4w 4wire14 e Interface IFB #15 4w 4wire15 signments Audio Settings	e Interface IFB #14 4w 4wire14 e Interface IFB #15 4w 4wire15 signments Audio Settings Options	e Interface IF8 #14 4w 4wire14 e Interface IF8 #15 4w 4wire15 signments Audio Settings Options Link Selector Ass	e Interface IFB #14 4w 4wire14 e Interface IFB #15 4w 4wire15 signments Audio Settings Options Unk Au	e Interface IFB #14 4w 4wire14 114 e Interface IFB #15 4w 4wire15 115 signments Audio Settings Options Unk Selector Assignments Unk Audio Sett	e Interface IFB #14 4w 4wire14 114 IFB #01 4w 115 I	e Interface IFB #14 4w 4wire14 114 IFB #01 4w e Interface IFB #15 4w 4wire15 115 IFB #01 4w >

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.

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.

The Client Configuration Selector Assignments window appears. The Selected Client field is populated.

IP: Direct IP Trunk / SIP Trunk 8 / GSM Pre -Assigned Selectors Talk/Li Description Type	rred						
Assigned Selectors Talk/Li Description Type							
Talk/Li Description Type					Assigned	Selectors	
		^	> Spl	it Talk/Listen>	Name	Selector	
. 19 Cha 19 char user nam Apple iOS			The Tak	Ik Oak			
. 20 Cha Test for 20 Char Apple iOS			10	ik only manning			
. Adam G. Adam Godlewski Windows D			> Lis	ten Only>			
. Alan x Windows D							
. alan3 alan3 Windows D			Spi	acer>			
alan4 Windows D			Ro	w			
. Analyst Analyst Apple iOS							
. Annou Windows D			Pa	ge>			
. Annou Windows D			0	II Notification			
. Antonio Antonio Esposito Google And			Co	in Nourication>			
. Arm St Arm Star 1 Google And	·			Demana			
. Arm St Arm Star 2 Google And				Kemove <			
. Arm St Arm Star 3 Google And				Clear>			
. Arm St Arm Star 4 Google And							
. Arm St Arm Star 5 Google And				D. F. L			
. Arm St Arm Star 6 Google And				Dupricate>			
. Arm St Arm Star 7 Google And				Latchable>			
. Arm St Arm Star 8 Apple iOS							
. Arnie Arnie Apple iOS				IFB>			
. Barry Barry Driver Google And				150			
Belt Pa Apple iOS				150			
. Beltpa Apple iOS				Speaker Dim ->			
. Beltpa Apple iOS					Colorite	en te Diseleu eus Deux	
Ben Ben Gullette Apple iOS				Hot Key>	Selecto	rs to Display per Kow	
. benopus benopus Windows D		~		Always On>	Default	Assignments to Non-Latchable	
hatart hatart Gasala Any				Andys on as	Calanta	· Anti-ation Mathed	On This Client Connect
					Selecto	r Activation Method	on the cherry connect

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.

	Туре	Talk/Listen Name	Login Name	Login Password	Description		Audio Settings	Options Templat	e	^	
SIP	Direct IP Trunk	CAM #01	17771		645-202-77	71	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #02	17772		646-202-77	72	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #03	17773		646-202-77	73	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #04	17774		646-202-77	74	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #05	17775		646-202-77	75	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #06	17776		646-202-77	76	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #07	17777		646-202-77	77	CAM #01	CAM #01			-
SIP	Direct IP Trunk	CAM #08	17778		646-202-77	78	CAM #01	CAM #01		-	
IP	Direct IP Trunk	CAM #09	17779		646-202-77	79	CAM #01	CAM #01			6
SIP	Direct IP Trunk	CAM #10	17780		646-202-77	80	CAM #01	CAM #01		-	1
IP	Direct IP Trunk	CAM #11	17781		646-202-77	81	CAM #01	CAM #01			6
IP	Direct IP Trunk	CAM #12	17782		646-202-77	82	CAM #01	CAM #01			E
IP	Direct IP Trunk	CAM #13	17783		646-202-77	83	CAM #01	CAM #01			
IP	Direct IP Trunk	CAM #14	17784		646-202-77	84	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #15	17785		646-202-77	85	CAM #01	CAM #01			
SIP	Direct IP Trunk	CAM #16	17786		646-202-77	86	CAM #01	CAM #01		٣	
elect	ed Client					Templates					
5	Selector Assignments	Audio Settin	gs	Options		Link Selector Assignments	Link Audio Se	ttings	Link Options		
						Inlink Selector Assignments	Unlink Audio S	ettings	Linlink Ontions		

15. Click the **OPTIONS button**.

The Client Configuration Options window appears. The Selected Client field is populated.

Answer)	
Answer)	
~ (0	DEFAULT
	DEFAULT
on v (C	DEFAULT
ation V (D	DEFAULT
(0	DEFAULT
(0	DEFAULT
(D	EFAULT
Гуре 🗸 Тур	pe 121
~	
~	
_	
_	
_	

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 Requires an RTS intercom panel with this phone line's key to manually answer the call. *Activation*

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.*

lient	List										
	Туре	Talk/Listen N	Listen Only N	Login Name	Login Passw	Description	RTS Alpha	Port	Audio Settin	Options Tem	^
SIP	Registered Trunk	310-8736448		101235_231	Intra73915c	310-873-6448		1	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #02		sip02		xxx-xxx-xxx02		2	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #03		sip03		x00x-x00x-x0x03		3	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #04		sip04		x00x-x00x-x0x04		4	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #05		sip05		xxx-xxx-xxx05		5	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #06		sip06		x00x-x00x-x0x06		6	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #07		sip07		x00r-x00r-x0t07		7	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #08		sip08		3000-3000-30008		8	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #09		sip09		x000-x000-x0009		9	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #10		sip10		x00-x00-x010		10	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #11		sip11		x00r-x00r-x011		11	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #12		sip12		x00r-x00r-x0r12		12	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #13		sip13		>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>		13	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #14		sip14		>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>		14	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #15		sip15		x00x-x00x-x0x15		15	310-8736448	310-8736448	
SIP	Direct IP Trunk	IFB #16		sip16		x00x-x00x-x0x16		16	310-8736448	310-8736448	~
<										>	
elec	ted Client					Templates					
				_		, emplotes					_

- 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.

VLink Client Configuration Add/Ed	lit			×
Client Identification Client Type Client Description Login Name: Login Password: Selector Talk/Listen Selector Listen Only Name:	SIP Device 4-Wire Int Aloha password 4W#1	: Four-Wire)	interface Allow Anonymous Login Use Domain Authentication	
Options Disable Client Login / Connection Always Show Selector when Off-line Latch Disable Talk Selector Party Line Operation IFB Destination ISO Destination Allow Assigment To Multiple Party Line Standard System Administration Privile		Selector As No Local As No Local As	signment Restrictions signment By Administrator signment By User	
(ОК	Cancel		

9. Click the OK button.

The VLink Client Configuration Add/Edit window closes. The added client is now visible in the Client Configuration window.

ent	List										
	Туре	Talk/Listen N	Listen Only N	Login Name	Login Passw	Description	RTS Alpha	Port	Audio Settin	Options Tem	^
VDI	Four-Wire Interface	CAM #10 4w		4wire26				126	IFB #01 4w		
/DI	Four-Wire Interface	CAM #11 4w		4wire27				127	IFB #01 4w		
/DI	Four-Wire Interface	CAM #12 4w		4wire28				128	IFB #01 4w		
/DI	Four-Wire Interface	CAM #13 4w		4wire29				129	IFB #01 4w		
VDI	Four-Wire Interface	CAM #14 4w		4wire30				130	IFB #01 4w		
VDI	Four-Wire Interface	CAM #15 4w		4wire31				131	IFB #01 4w		
VDI	Four-Wire Interface	CAM #16 4w		4wire32				132	IFB #01 4w		
/DI	Four-Wire Interface	Phoner #01		4wire33				133	IFB #01 4w		
VDI	Four-Wire Interface	Phoner #02		4wire34				134	IFB #01 4w		
VDI	Four-Wire Interface	Phoner #03		4wire35				135	IFB #01 4w		
/DI	Four-Wire Interface	Phoner #04		4wire36				136	IFB #01 4w		
VCP	Windows Desktop	Monitor #1		monitor1	MONITOR1	System Monitor #1		201			
VCP	Windows Desktop	Monitor #2		monitor2	MONITOR2	System Monitor #2		202			
VCP	Windows Desktop	Monitor #3		monitor3	MONITOR3	System Monitor #3		203			
SIP	Four-Wire Interface	4W#1		Aloha	password	4-Wire Interface 15		41			
											~
6										>	
electe	ed Client					Templates					
S	elector Assignments	Audi	o Settings		Options	Link Selector Assignme	ents Link A	udio Sett	ings	Link Options	

Configuring Client Options

- 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.
- 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.*

Selected Client		
SIP Options Wire Interface / 4W#1 / 4-Wire Interf	ace 15	
Inbound Session Activation	On Call Received (Auto-Answer)	V (DEFAULT
Inbound Session Deactivation	On Forced Disconnect	OEFAULT
Outbound Session Activation	On Talk Selector Activation	V (DEFAULT)
Outbound Session Deactivation	On Talk Selector Deactivation	OEFAULT
Automatic Dial Sequence		
Send SDP With Invite Request	\checkmark	(DEFAULT
Use SDP for RTP Destination	\checkmark	(DEFAULT
Use RTP Packets for Voice Activity Detection		(DEFAULT
RTP Timeout In Seconds	0	
RTP Keep Alive Method	Unknown RTP Payload Type	✓ Type 121
SIP Session Expiration Time In Seconds	0	
Voice Activity Detection Validation Time In Ms	0	
Voice Activity Detection Off Delay Time In Ms	0	
Call Notification Ring Tone		×
Auto-Answer Notification Message		×
Auto-Answer DelayTime In Ms	0	(DEFAULT
Auto-Answer Access Code		
Use 16 KHz for G.722 Timestamp		
Internal SIP Client User Name Prefix		
Digits to send on Talk Activation / Deactivation		
SIP Direct IP Trunk / Registered Trunk Options		
SIP Target User Name		
SIP Target Primary Host Name		
SIP Target Secondary Host Name		
SIP Target Proxy Server IP Address (optional)		
SIP Registration Expiration Time In Seconds	300	
STUN Server or Proxy IP Address		

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.

VLink Client Configuration Aud	dio Settings	×
Selected Client SIP: Four-Wire Interface / 4W#1 /	4-Wire Interface 15	
Audio Quality Audio Encoder/Decoder Audio Encode Sample Rate Audio Encode Quality Audio Encode Complexity Variable Bit Rate Audio Transmission Jitter Buffer Size Silence Suppression Time Packet Resequencer Depth	Speex (Code Excited Linear Prediction)	(DEFAULT) (DEFAULT) (DEFAULT) (DEFAULT) (DEFAULT)
Audio Levels Automatic Gain Control		(DEFAULT)
Automatic Gain Control Level	Standard	(DEFAULT)
Audio Input Level Gain (Pre-Mix)	0 dB	(DEFAULT)
Audio Output Level Gain (Post-Mix)	0 dB	(DEFAULT)
Speakerphone Speaker Dim	None	(DEFAULT)
Audio Processing		
Echo Cancellation		
Echo Cancellation Tail Length		
Audio Encryption	No Encryption	(DEFAULT)
	OK Cancel	

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.

electi SIP:	Four-Wire Interfa	ace / 4W#1 / 4-Wire Inte	rface 15								
on-As	signed Selectors							Assigned Select	ors		
	Talk/Listen N	Description	Туре	^	> S	plit Talk/Liste	n>	Name	Selector Type		
VDI	CAM #06 4w		Four-Wire Interface		> T	alk Only	·····>	CAM #15 4w	Listen Only		
/DI	CAM #07 4w		Four-Wire Interface		-	dik only					
/DI	CAM #08 4w		Four-Wire Interface		> L	isten Only	>				
/DI	CAM #09 4w		Four-Wire Interface		9	nacor					
/DI	CAM #10 4w		Four-Wire Interface		5	pocer					
/DI	CAM #11 4w		Four-Wire Interface		R	ow	>				
'DI	CAM #12 4w		Four-Wire Interface		P	age	>				
'DI	CAM #13 4w		Four-Wire Interface				_				
'DI	CAM #14 4w		Four-Wire Interface		0	all Notificatio	n>				
/DI	CAM #15 4w		Four-Wire Interface				_				
/DI	CAM #16 4w		Four-Wire Interface			Remove	<				
/DI	IFB #01 4w		Four-Wire Interface			Clear	>				
'DI	IFB #02 4w		Four-Wire Interface				_				
'DI	IFB #03 4w		Four-Wire Interface			Durchaste					
'DI	IFB #04 4w		Four-Wire Interface			Duplicate	>				
'DI	IFB #05 4w		Four-Wire Interface			Latchabl	e>				
/DI	IFB #06 4w		Four-Wire Interface			100					
/DI	IFB #07 4w		Four-Wire Interface			1-8	>				
/DI	IFB #08 4w		Four-Wire Interface			ISO	>				
/DI	IFB #09 4w		Four-Wire Interface			Combus					
/DI	IFB #10 4w		Four-Wire Interface			Speaker	Dim ->				
/DI	IFB #11 4w		Four-Wire Interface			Hot Key	>	Selectors to D	isplay per Row		
/DI	IFB #12 4w		Four-Wire Interface			Abumuna	2.2. 2	Default Assign	ments to Non-Latchabl	e	
10.7	100 e10 A		Cour Mire Interface	~		Always	Ju>				

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

wire02 wire03	Ready to Connect	0141	GPI #Z	GPO	GPO	~	
wire03	Ready to Connect						
	Ready to Connect						
wire04	Ready to Connect						
wire05	Ready to Connect						
wire06	Ready to Connect						Add
wire07	Ready to Connect						
wire08	Ready to Connect						Edit
wire09	Ready to Connect						
wire10	Ready to Connect						Delete
wire11	Ready to Connect						
wire12	Ready to Connect						
wire13	Ready to Connect						
wire14	Ready to Connect						
wire15	Ready to Connect						
wire16	Ready to Connect					~	

2. Click the **Configure button**.

The Device Interface Configuration window appears.

VLink Device Interface Conf	guration	×
Network Settings	102 160 1 55	
Virtual Matrix IP Address	192.168.1.55	1000
Monitor Device		
Select Speaker	(None)	~
Logic Input/Output Select GPIO Device #1	(None)	~
Select GPIO Device #2	(None)	~
Select GPIO Device #3	(None)	\sim
	Apply	

- 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*.

Login Namo	Aloha	
Login Name		
Login Password	•••••	
Analog to Digital Audio Device —		
Select Input Device	Dante Virtual Soundcard (ASIO)	\sim
Select Input Connector	Dante rx 15	\sim
Select Input Channel	Mono	
Set Input Level		
Select Output Device	Dante Virtual Soundcard (ASIO)	\sim
Select Output Connector	Dante tx 15	\sim
Select Output Channel	Mono	\sim
Set Output Level		
Logic Input/Output		
Select Logic Input #1	(None)	\sim
Select Logic Input #2	(None)	\sim
Select Logic Output #1	(None)	\sim
Select Logic Output #2	(None)	\sim

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:

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Routing Device Info Clock Status Networ	rk Sta	tus	E١	ent	s												
@Dante [®]		+ 080	- IM0-0	10	38	8 4	8	90	07	8	6 9	2 =	12	13	4	2 4	2
Filter Transmitters	ters	IMM-90	OMNEC														
Filter Receivers	ransmit	DESKT															
	ante																
	a =																
+ - Dante Receivers																	
Dante Receivers DESKTOP-MMD038D		÷	-														
Dante Receivers		÷	-														Ì
Dante Receivers DESKTOP-MMD038D 01 02 03 03 01 02 03 03 03 04 05		÷	=														Ì
Dante Receivers DESKTOP-MMD038D 01 02 03 04		÷	=														ĺ
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05		Ŧ	=														ĺ
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05 06 06 0 0 0 0 0 0 0 0 0 0		÷	=														ĺ
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05 06 07 07 0 0 0 0 0 0 0 0 0 0		÷															ľ
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05 06 07 08		÷	8														
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05 06 07 08 09		÷															
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05 06 07 08 09 10		Ŧ	8														
Dante Receivers DESKTOP-MMDD38D 01 02 03 04 05 06 07 08 09 10 11		÷	8														
Dante Receivers DESKTOP-MMD038D 01 02 03 04 05 06 07 08 09 10 11 12		÷															
Dante Receivers DESKTOP-MMDOJ8D 01 02 03 04 05 06 07 08 09 10 11 12 13		Ŧ															
Dante Receivers DESKTOP-MMDOJ8D 01 02 03 04 05 06 07 08 09 10 11 12 13 14 4		Ŧ															
Dante Receivers DeskTOP-MMD038D 01 02 03 04 05 06 07 08 09 10 11 12 13 14 15		Ŧ															

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.

🥺 Dante Controller - Network View													_]		>	<
<u>F</u> ile <u>D</u> evice <u>V</u> iew <u>H</u> elp																				
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Routing Device Info Clock Status Netw	ork Sta	tus	E	ven	ts															
Dante [®]		- 0800	01	02	83	4	ß	8	07	80	8	10	-11	12	13	14	15	16	17-	5
Filter Transmitters	ters	IMM-90																		
Filter Receivers	ransmit	DESKT																		
- Dante Pereivers	+ Dante																			
PDESKTOP-MMD018D		Ŧ																		^
BOMNEO-OMI -01 -02 -03 -04 -05 -06 -07 -08 -09 -10 -11 -12	0000		0	0	0	2														
- 13 - 14 - 15		<																	>	~
P: 📃 Multica	st Ban	dwi	dtl	h: 0) bp	s	Ev	en	t Lo	og:		Clo	ock	Sta	atu	s M	loni	itor	:	

- 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.

CHAPTER 6

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.



• Eil	AZ	edit	- [0	NLIN	VE] -	SIP S	erve	r Des	crip	tions	Surte	 Alp	har	Status
10	n N	<u>6</u> 2	m2a		anen ا	l .at	m 3	<u>con</u>		~	<u>J</u> yste	Ξ·P		Ba
		-	4E		₩	18.	비크	~	~	^	1 = 2	 •	1 01	
	S	IP	\triangle	D)es	crip	tio	n				 		
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		00	2											
		00	3											
		00	4											
		00	5											
		00	6											
		00	7											
		00	8											

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

Edit Alpha / Desc	ription	? ×	<				
Resource:	SIP Server - 001	Ne <u>x</u> t]				
Description:	Description: Desktop MMD0J8D						
		Done					
		Cancel					

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.

National Alexandre Server Descriptions	
<u>File Online Authentication Edit View System Alphas Status Options Logging E</u>	<u>l</u> elp
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SIP 🛆 Description	
001 Desktop MMD0J8D	
002	
003	
004	
005	
006	
007	
008	

5. Click Options | SIP Server Communications.

The SIP Server Communications window appears..

ription	IS							
View	System	Alphas	Status	Options	Logo			
I	Preference	5			e			
(Communic	ations						
1	TM Comm	unication	s					
9	SIP Server (Communi	cations					
1	ntercom C	onfigurat	ion					
I	Frame Mapping Table							
ł	Port Alloca	tion Table	ż		I			
ł	Reset Capa	bilities						
I	Ethernet Se	tup						
5	SNMP Con	figuration	·		I			
(GPIO-16 Co	onfigurati	on					
I	DHCP Serv	er Config	uration					
	Upload Del	oug Inforr	mation	>				

SIP Server	Description	IP Address - Main	IP Address - Backup
1	Desktop MMD0J8D	192.168.1.55	-
2		-	-
3		•	-
4		•	-
5		•	-
6		•	-
7		•	-
8		•	-

- 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**.

CHAPTER 7

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.

Port	Alpha		1	Scroll Enab	ole	Keypanel /	Port Setting:	s			Port S	tatus			
15	N015			✓ Local ✓ Trunk ✓ AZedit ✓ PAP/L	СР	Page 1: M	IAIN	• Ec	iit ∦⊲		671 1 671 1 679 1				
sten Ke	ays			200											a s
D	Г	Г	Г	Г	Г	Г	Г	Г	Г	Г	П	Г	Г	Г	Г
R	E	П	E	Г	Г	E	Г	Г	Г	Г	Г	Г	Г	Г	Г
1 ilk Key	2 s (Levels 1	3 and 2)	4	5	6	7	8	9	10	11	12	13	14	15	cww
D	Г	Г		Г	Г	Г	Г	Г	П	Г	П	Г	Г	Г	Г
R	П	П	Г	Г	Г	Г	П	Г	Г	Г	Г	Г	Г	Г	Г
				[
												- CO			

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

2	Cliak	tha	Advanged	tah
ა.	CHCK	une	Auvanceu	l lad.

riorities	Options
FB Priority	Enable Tone
Trunk IF <u>B</u> Priority 1	Keypanel Privacy
Irunk Priority	TIF <u>D</u> ial-Out Restrict
^y a <u>n</u> el Poll Delay (ms)	Key Labels
0 <u>R</u> eset	Do Not Interrupt
FB Listen Destination	Priority Call <u>V</u> olume +0.0 dB
Port Port Number Alpha	Enable Keypanel Mirroring
15 N015 •	
IP Server / Port Selection	1
siP test 💽	
(

- 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.

	-	Edit	-<)))	
•			11 49	

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.

AZedit (El	NT:ADMIN)	[ONLINE] -	SIP Server Stat	us			
e Online A	Authentication	Edit View	System Alphas :	Status Options	Logging Help		
0 😅 🖻		" 🖻 🛃 🔎	• × ∽ ⊂ ,	/ % Pa 6	Q 🎉 - F -	- 🔶 🧼	k? 🛈 🖪
SIP /	Comm	Status	Errors To	BER To	Errors From	BER From	Description
001	OK	Cur	÷ 👋	20 E	2	12	CR71 SIP SERVER
002	OK	Cur	2	2	2	-	CR74 SIP SERVER
003	ОК	Cur	<u>a</u>	<u>a</u>	<u> 8</u> 2	3328	CR72 SIP SERVER
004	120		-	-	5	1.70	CR6E SIP SERVER
005	1763	1070	5	5	2	1.7	CR73 SIP SERVER
006	OK	Cur	-	-	()	89 7 8	CR7A SIP SERVER
007	-	-	-	-	=		
008	140	100	<u></u>	-	<u>~</u>	2240	

8. Check the more detailed status information by going to Status/Ethernet Links/SIP Server Links:

Port		01000						10	-	1.53
Master (optroller				4 11 1	P.	-9-	1	1	e
Standby	Controller			I						
TBX Link				İ						
I/O Card	s									
MADI Ca	rds									
PAP and	LCP-102									
UIO-256							1.5.1.00		- 2	
Ethernet	Links				Fran	ne-t	o-Fram	e Links		Ľ.
Software	Versions	Ctrl-	+Shift+\	•	Trun SIP (ik M Serv	laster L ver Link	inks s		
Trunk Ma	ster			1	_	-				
Trunk Po	rts			- 1						
Remote	ntercoms			_						
SIP Serv	er									
RVON CO	nnections									
OMNEO	Connections									
Alarms				-						

Zedit - [ONLIN	IE) - Ether	net Link Stati	us: SIP Server	us Onlines La	aning Halp					
ink# 🔺	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	<u>12</u> 11	0	0	0	0	
2:1	OK	Active	3.199.64.146	4	6	5	0	367	8	
2:2	1.0	-		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		120		0	-	0	0	0	0	
5:1	-	-		0		0	0	0	0	
5:2	-	-		0	1211	0	0	0	0	
6:1	OK	Active	3.199.64.76	2	56	88	1	586	2	
6:2	1.2	10.000 and 1		0	07.0	0	0	0	0	
7:1	12	-		0	020	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:

- 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