Gateway SIP Dialling

This quick reference user guide has been devised to act as a basic configuration instruction to enable Gateway SIP connectivity and panel SIP Dialling:

EHX/Matrix

With a frame containing an MVX card(Pico's can select Tel 14 as a port option on ports 17-32), set your chosen port to be allocated as a Tel-14. On the global options section of the tab set 'Hardware Detection' to off by unchecking the box.

Layout Hardware Hel	p Cards an	id Ports 🗙			
Card Slot					
1: MVX-A16	Port Number F	Port Function	Label	Description	
	1.1 (1)	Telephone (TEL-14) ~	Gateway1	Telephone (TEL-14)	

Fig 1 – MVX port allocated as Tel-14

 Global Options 	
Auto Listen	
Auto Signal Tone	
Global ISO	
Hardware Detection	
Latch Disable	
Mute Listen With Talk	-
Partyline Turnaround	
Prevent Reply Signalization	
IFB Caller Priority	3
IFB	
IFB Attenuation	Full Cut

Fig 2 – Tel-14 Global Options, Hardware detection 'unchecked'

Once this is done place a key onto a panel with a keypad and apply changes(without reset). The method for dialling from the panel face will be explained later in the document. From the MVX rear card we now run a CAT5 cable from the Tel-14 port to the physical Gateway device.



Fig 3 – MVX CAT5 connection to Gateway; these connections must match with how the Gateway ports have been configured ie: MVX Tel-14 port 1 to Gateway port 1 etc..

Gateway Web Browser configuration

We need to first make sure that we can gain access to our Gateway unit via IP, this done by connecting via a switch or directly to port seven on the Gateway itself. The default Gateway IP address is 10.1.1.253. For this guide the IP address has been changed to 192.168.42.23.



Fig 4 – Gateway IP connection to either a switch or direct to a PC

In addition to the IP connection to the Gateway you also need to be able to connect to the Gateway via the USB connection on the rear of the unit. This is so we can connect directly to the unit utilising the Gateway software:



Fig – 5 Gateway rear USB PC connection for software access

We begin by connecting to the unit via the web browser. This is done by opening a web browser of your choice and entering the units IP address, it will then prompt you for a username and password; these will be set to default username: admin, password: admin if they have not been changed.

You will be greeted with the VOIP status screen:



VoIP/RoIP Gateway Configuration

		System Information
System Information		Network
• Status	Address	192.168.42.23
Network Setup	Gateway	0.0.0.0
 Network 	DNS Primary	0.0.0.0
SIP Setup	DNC Cocondamy	0.0.0
O SIP Accounts	DNS Secondary	0.0.0
O SIP Codecs	DNS Tertiary	0.0.0.0
O SIP Directory		System
 Advanced 	CPU Utilization	21%
 Interface 	Memory Available	1005532
RTP Sessions		1009
RTP Sessions	User	1009@192.168.42.27
Blade Link Setup	Registration Status	PROXY REGISTERED
 Gateway Link Setup 	Registration Status	HOM_REGISTERED
 Gateway Link Status 		
O Gateway Link Config		
O PSTN Setup		
System		
O Local Analog Port		
 Administration 		
 Date/Time 		
 Backup/Restore 		
 Upgrade Firmware 		
Logout		

Fig – 6 Gateway VOIP status screen

The network tab is where we can alter the IP address and other parameters of the Gateway unit.

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VoIP/RoIP Gateway Configuration

		Network Configuration
System Information		General
 Status 	Host	CCVoIP
Network Setup		
 Network 	Domain	
SIP Setup	Connection Type	🔍 Dynamic IP 🖲 Static IP 🔍 PPP0E
O SIP Accounts		Static IP Address
 SIP Codecs 		
 SIP Directory 	Address	192.168.42.23
O Advanced	Mask	255 255 255 0
 Interface 		
RTP Sessions	Default Router	0.0.0
RTP Sessions	DNS Primary	0.0.0.0
Blade Link Setup	DNS Secondary	0.0.0
 Gateway Link Setup 	Dito occondary	
 Gateway Link Status 	DNS Tertiary	0.0.0
 Gateway Link Config 		Additional Settings
O PSTN Setup	MTU Size (advanced)	1500
System		VIAN
 Local Analog Port 		
 Administration 	VLAN	Enabled
 Date/Time 	ID	4 (value: 0 to 4094)
 Backup/Restore 	User Priority	0 - Best Effort V (default: 0)
 Upgrade Firmware 		Save Changes
 Logout 		

Fig 7 – Gateway Network tab

SIP Accounts is where we configure each port on the Gateway and how we want it to Perform its functions. In this example we have assigned a phone number and set the IP address of the SIP server, in this guides example the SIP server IP address is set to 192.168.42.27.



VoIP/RoIP Gateway Configuration

SIP Account				
System Information				
 Status 	SI	P Account Management		
Network Setup	Select Existing Account	1000		
 Network 	Select Existing Account			
SIP Setup	SI	P Account Configuration		
 SIP Accounts 	Account Active	Enable		
o SIP Codecs	Description	1009		
 SIP Directory 	Username/Number	1009		
 Advanced 	Dischar News	1000		
 Interface 	Display Name	1009		
RTP Sessions	SIP Domain/IP Address	192.168.42.27		
 RTP Sessions 	Accor	isted Analog Port Fattings		
Blade Link Setup	ASSOC	lated Analog Port Settings		
 Gateway Link Setup 	Use VoIP Module Analog Port?	(last port on gateway)		
 Gateway Link Status 	Cateway Module Number	1 Module Number (1 - 4)		
 Gateway Link Config 	Gateway Housie Maniber			
 PSTN Setup 	Gateway Module Port	1 Module Port Number (1 - 2)		
System				
 Local Analog Port 	Auto Answer Calls	 Image: A start of the start of		
 Administration 	Translate DTMF	(OOB DTMF is converted to analog DTMF)		
0 Date/Time		· · · ·		

Fig 8 - SIP Account tab

Scrolling further down the SIP Account tab make sure that the Keying type is set to 'voice activated', and that the Key Radio DTMF address and Clear DTMF buffer as set as they are below. Also make sure to set the proxy settings as the SIP server address and also make sure that whatever username and password you set here will match what we will configure on the SIP server later in the guide. When these changes are complete click 'Save Changes' at the bottom of the page.

SIP Endpoint Radio	Keying Options (if not using Blade Link)
Enable SIP Endpoint Keying	
Keying Type	Voice Activated DTMF Activated
Key Radio DTMF Address	*
Clear DTMF Buffer	#
Use Keying Tones	■ F1 - 1950Hz ▼
Mute Input DTMF Code	1
UnMute Input DTMF Code	2
Play PTT/Mute Notification To Endpoint	8
	Proxy Settings
Proxy	192.168.42.27 (leave blank to register with domain)
Proxy Port	0 (advanced; set to 0 to auto detect)
TCP/IP Port	0 (advanced; default 5060)
Username	1009
Password	••••
Register with Server	Enabled
	Additional Settings
Silence Suppression	(suppress RTP packets when silent)
AGC	(automatic gain control)
Jitter Critical Depth	300 (ms) default = 300
Jitter Target Depth	90 (ms) default = 90
Session Timer	102 (Seconds)
	Save Changes

Fig 9 – SIP Account page bottom

Next we move onto the SIP Codec page, on this page please make sure that the correct codecs are allocated properly. The matrix can utilise codec G711, so select as shown below:



Fig 10 – Codec selection

Once the codec has been selected make sure you again click 'Save Changes' and proceed onto SIP directory.

Set the SIP Directory settings as below including the Enable Phone directory.

On the SIP Directory tab set the number you wish to dial and then the URI on the right. This will include the number and the SIP server address:



VoIP/RoIP Gateway Configuration							
	Ana	alog-to-SIP F	Phone Dire	ectory			
System Information							
 Status Network Setup 	This phone book allows analog dev on a given port use the "dtmf" telr	vices such as radios, tel net command (help loca	lephones, and PST ated in the telnet to	N ports to dial SIF erminal).	P endpoints.	To adjust DTMF d	etection parameters
Network SIP Setup	Enable Phone Directory						
 SIP Accounts 	Clear Down Digits		#				
SIP Codecs SIP Directory	Call Hangup		*				
9 Advanced	Provide Ringback						
• Interface	DTMF Code	e	URI				
RTP Sessions	1005		1005@192.168.42	.27			
 RTP Sessions 							
Blade Link Setup							
 Gateway Link Setup 							
 Gateway Link Status 							
 Gateway Link Config 							
PSTN Setup							

Fig 11 – SIP Directory

Once the changes made in the directory have been saved, the user now needs to configure the Gateway via the software on the PC through the USB connection.

Gateway Software Configuration

Once you have installed the Gateway software you can connect to the unit using the USB cable. Once this connection is in place you can open the software and click 'connect to device'. Once connected the user should see the following:



Fig 12 – Gateway software main screen

The user is primarily interested in the first analogue 2/4W port. By clicking on this first port a further port menu will open. On this menu the user is looking to configure the DTMF functionality of the port. A smaller box should appear, in this click on the 'settings' button:

Communication System



Fig 13 – Analogue Port menu

A larger settings menu box should appear and in this select 'enable DTMF Detector' and then enter the Advanced menu by clicking on Advanced -

Analog 2/4W 1 - {c8f2e4b5-01	25-45ef-b559-e5	53896f	- • •
Port Status	rt Information	Martula Da	
YMT COR COR	98 Board 36		A wire E /M
PCV O PTT O Voted Mo	dule Position: 1	Analog 2	2/4-wire E/M
	dule Type: 153	Status	
Transmit Collapse Mo	dule Port: 1	Fun	ctioning
CTCSS SNR LEDs Muting Audio D	elays Keying Utilitie	es Events	
Description Location Tones Interface S	ettings Levels ALC	DTMF	DSP
DTMF address for this port.			
321		1 2	3 A
Enable DTMF Detector		4 5	6 B
		7	
Tracerti DT	VE Dista	/ 8	9 C
Advanced 15	9 Send	* 0	# D
			المنصن
			Save
			.::

Fig 14 – Analogue port Settings menu

Once in the advanced menu, select 'DTMF Detection to Extended' and click 'Done'

Configure Advanced DTMF O	
Port Name Analog 2/4W 1	
Digit Length and Pause Digit Length: How long in milliseconds that a DTMI digit is generated from this port (default is 64): Digit Pause: How long in milliseconds between DTMF digits generated from this port (default is 64)	F 64
Advanced DTMF Detection Settings	77
DTMF Detection Normal Extended	
Set Defaults	Done

Fig 15 – Advanced Analogue settings tab

Gateway configuration is complete, we move onto the SIP server setup followed by a walkthrough of how to enable Panel dialling.

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SIP Server

SIP servers of course vary from manufacturer to manufacturer but the core principles remain the same throughout.

The user will need to create accounts for each port of the Gateway device on the SIP server. So each port will be assigned a phone number and each port will require the Gateway units IP address.

This is of course where the users phones will also be connected, and for the purposes of this guide we have placed 2 x SIP phones on the server with numbers 1004 and 1005.

Dialling from the V-Series Panel

- 1. First press the number 1 key on the keypad, this starts the DTMF process at the panel face.
- 2. Press # to clear down any remaining digits from a previous call
- 3. Dial the number required, in this example we dial 1005
- 4. Press the Gateway Key on the panel to open the port
- 5. Wait for the dialling process to complete



Fig 16 – Successful dialling image