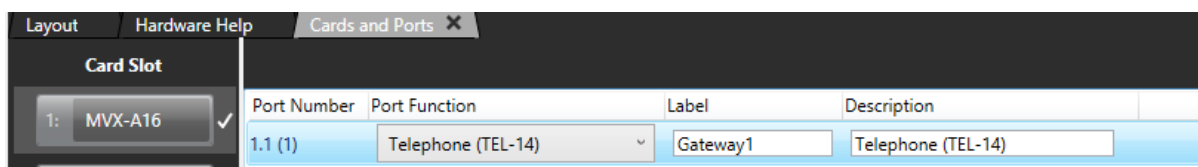


# Gateway SIP Dialling

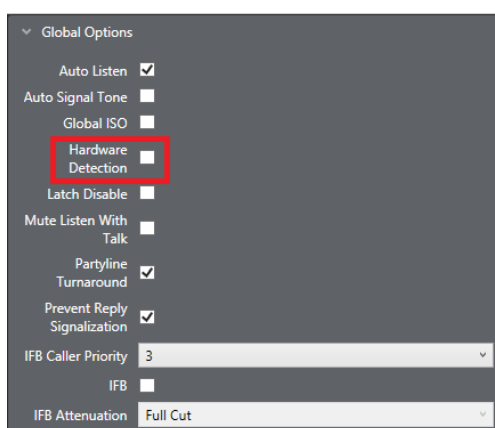
This quick reference user guide has been devised to act as a basic configuration instruction to enable Gateway SIP connectivity and panel SIP Dialling:

## 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.



**Fig 1 – MVX port allocated as Tel-14**



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

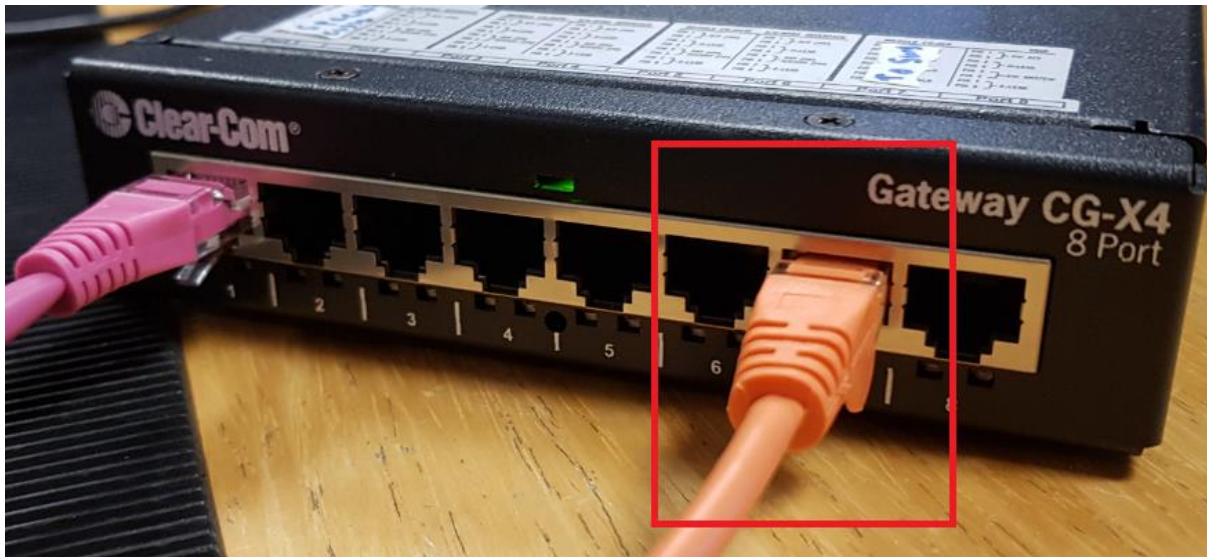
Once this is done place a key onto a panel with a keypad and apply changes(without reset). The method for dialling from the panel face will be explained later in the document. From the MVX rear card we now run a CAT5 cable from the Tel-14 port to the physical Gateway device.



**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:

Fig – 6 Gateway VOIP status screen

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

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	
<b>System Information</b>	
o Status	
<b>Network Setup</b>	<b>SIP Account Management</b>
o Network	Select Existing Account: 1009
<b>SIP Setup</b>	<b>SIP Account Configuration</b>
o SIP Accounts	Account Active: <input checked="" type="checkbox"/> Enable
o SIP Codecs	Description: 1009
o SIP Directory	Username/Number: 1009
o Advanced	Display Name: 1009
o Interface	SIP Domain/IP Address: 192.168.42.27
<b>RTP Sessions</b>	
o RTP Sessions	
<b>Blade Link Setup</b>	<b>Associated Analog Port Settings</b>
o Gateway Link Setup	Use VoIP Module Analog Port?: <input type="checkbox"/> (last port on gateway)
o Gateway Link Status	Gateway Module Number: 1 (Module Number 1 - 4)
o Gateway Link Config	Gateway Module Port: 1 (Module Port Number 1 - 2)
o PSTN Setup	Auto Answer Calls: <input checked="" type="checkbox"/>
<b>System</b>	Translate DTMF: <input checked="" type="checkbox"/> (OOB DTMF is converted to analog DTMF)
o Local Analog Port	
o Administration	
o 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	<input type="checkbox"/>
Keying Type	Voice Activated <input checked="" type="radio"/> DTMF Activated <input type="radio"/>
Key Radio DTMF Address	* <input type="text"/>
Clear DTMF Buffer	# <input type="text"/>
Use Keying Tones	<input type="checkbox"/> F1 - 1950Hz
Mute Input DTMF Code	1 <input type="text"/>
UnMute Input DTMF Code	2 <input type="text"/>
Play PTT/Mute Notification To Endpoint	<input checked="" type="checkbox"/>
<b>Proxy Settings</b>	
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	<input checked="" type="checkbox"/> Enabled
<b>Additional Settings</b>	
Silence Suppression	<input checked="" type="checkbox"/> (suppress RTP packets when silent)
AGC	<input checked="" type="checkbox"/> (automatic gain control)
Jitter Critical Depth	300 (ms) default = 300
Jitter Target Depth	90 (ms) default = 90
Session Timer	102 (Seconds)

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:

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## VoIP/RoIP Gateway Configuration

### Status

**System Information**

- o Status
- Network Setup**
- o Network
- SIP Setup**
- o SIP Accounts
- SIP Codecs**
- o SIP Directory
- o Advanced
- o Interface
- RTP Sessions**
- o RTP Sessions
- Blade Link Setup**
- o Gateway Link Setup
- o Gateway Link Status
- o Gateway Link Config
- o PSTN Setup
- System**
- o Local Analog Port
- o Administration
- o Date/Time

**RTP Configuration**

Port Range:  to

**Codec Selection**

The G.729 codec may require a license arrangement between you (the user) and an intellectual property rights holder (IPR). It is your responsibility to determine whether any licenses or fees are due to an IPR holder.

Available		Preferred
<ul style="list-style-type: none"> <li>G.711 uLaw</li> <li>G.711 aLaw</li> <li>G.726 (16kbps)</li> <li>G.726 (24kbps)</li> <li>G.726 fixed payload</li> <li>G.726 (40kbps)</li> <li>DVI4 Narrowband</li> <li>Linear PCM</li> <li>Linear PCM (little endian)</li> <li>G.729</li> <li>ILBC-30</li> <li>ILBC-20</li> <li>SPEEX Narrowband</li> </ul>	<input type="button" value="Add &gt;&gt;"/> <input type="button" value="&lt;&lt; Remove"/>	<ul style="list-style-type: none"> <li>DVI4 Narrowband</li> <li>G.711 uLaw</li> </ul>
		<input type="button" value="Move Up"/> <input type="button" value="Move Down"/>

**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:

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## VoIP/RoIP Gateway Configuration

### Analog-to-SIP Phone Directory

**System Information**

- o Status
- Network Setup**
- o Network
- SIP Setup**
- o SIP Accounts
- o SIP Codecs
- SIP Directory**
- o Advanced
- o Interface
- RTP Sessions**
- o RTP Sessions
- Blade Link Setup**
- o Gateway Link Setup
- o Gateway Link Status
- o Gateway Link Config
- o PSTN Setup

This phone book allows analog devices such as radios, telephones, and PSTN ports to dial SIP endpoints. To adjust DTMF detection parameters on a given port use the "dtmf" telnet command (help located in the telnet terminal).

**Enable Phone Directory**

**Clear Down Digits**

**Call Hangup**

**Provide Ringback**

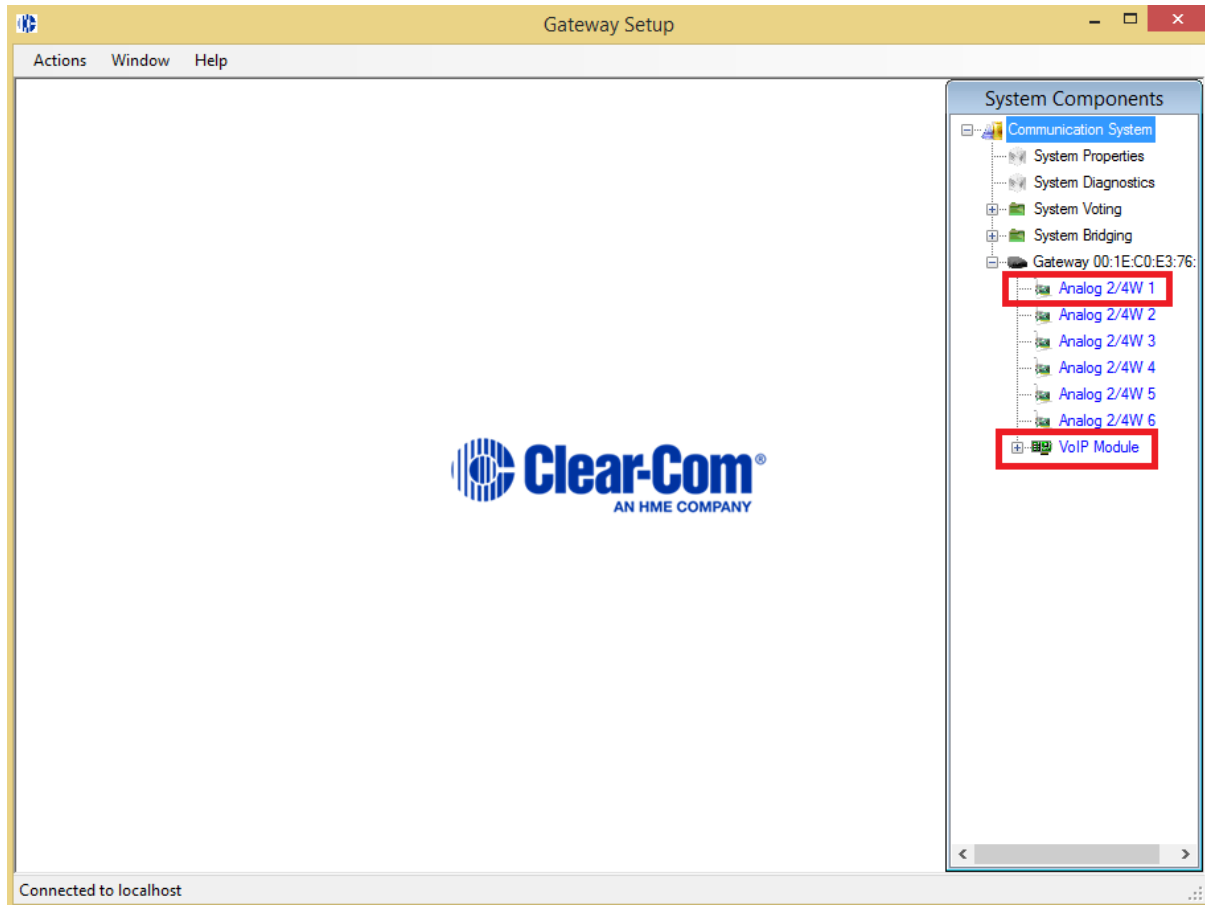
DTMF Code	URI
<input type="text" value="1005"/>	<input type="text" value="1005@192.168.42.27"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

**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**

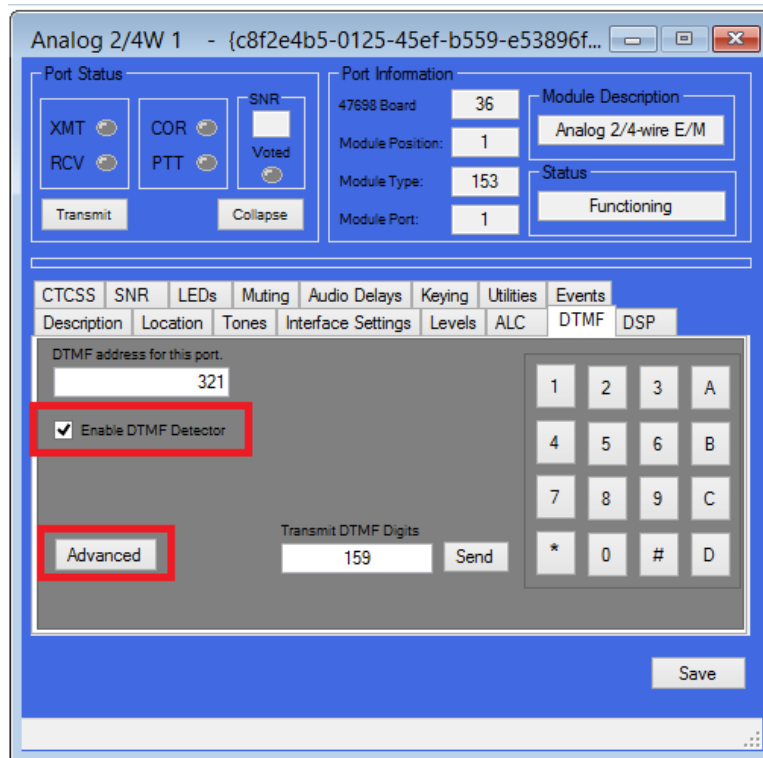
The user is primarily interested in the first analogue 2/4W port. By clicking on this first port a further port menu will open. On this menu the user is looking to configure the DTMF functionality of the port. A smaller box should appear, in this click on the 'settings' button:



**Fig 13 – Analogue Port menu**

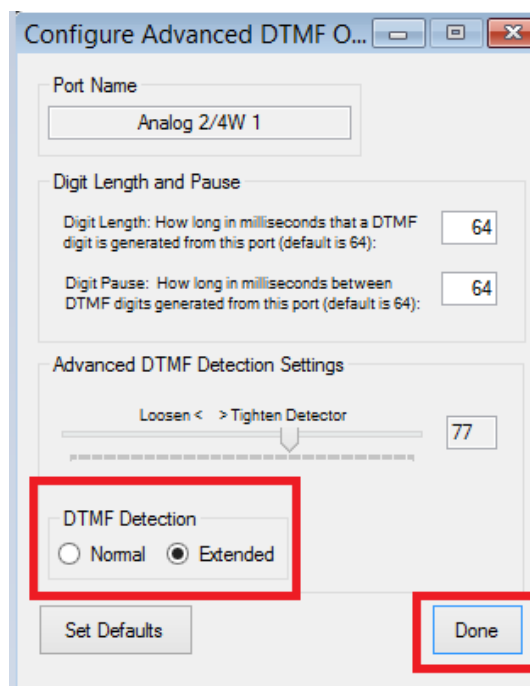


A larger settings menu box should appear and in this select 'enable DTMF Detector' and then enter the Advanced menu by clicking on Advanced -



**Fig 14 – Analogue port Settings menu**

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

## 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*